Issue No. 14 Summer through Winter 1989-90

The Audio Critic®

Retail price: $7

In this issue:

We review in depth seven different loudspeaker systems, between $798 and $3995 the pair, and try to figure out why no two of them sound remotely alike, not even those with the most accurate response.

The "Seminar 1989" transcript, now split into three parts, continues with Part II, again bringing you hours and hours of strong opinions on the major issues of audio, as argued by Bob Carver, Dave Clark, John Eargle, Stanley Lipshitz, Peter McGrath, and your Ed.

Once again, the terrifying subject of duplicating "the tube sound" with transistors raises its Gorgon head.

Plus other reviews, columns, and features, including another foray into the realm of high-tech video.

The Audio Critic

Issue No. 14 Summer through Winter 1989-90

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

For subscription information and rates, see inside back cover.

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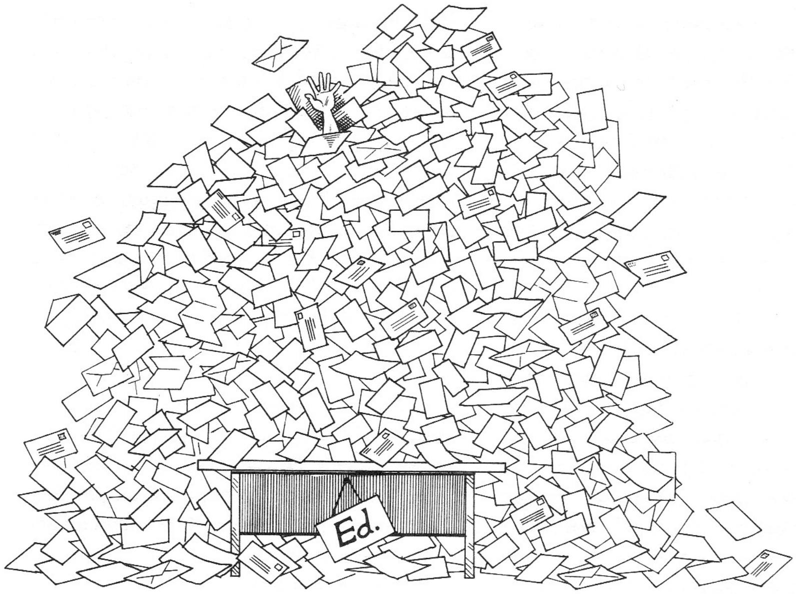
Note: All unsigned articles and reviews in this issue were written by Peter Aczel, Editor and Publisher.

Just a Few Words from the Editor/Publisher:

The subject of our continuing lateness is fully explored on the opposite page, in the first letter to the Editor and our reply to it. Here we just want to state for the record that we have published 5 issues within the past 26 months, proving at least the reality of our "resurrection," if nothing else. We feel we have also demonstrated that the verifiable truth about audio makes at least as interesting reading as myths, cultism, and politics, and that the finest minds in the business are on our side, not on the side of the myth-makers, cultists, and politicians. A lot more than that will have to be accomplished before we are a totally effective and businesslike operation; some of it, perhaps even most of it, we expect to take place in 1990. We thank all of you who have given us your greatly needed support.



Box 978

Letters to the Editor

We are fully aware that some people read this column as a kind of "Theater of Cruelty"—even if they have never heard of Antonin Artaud—but that is not its purpose. We greatly prefer correspondence from which we and our readers can learn something about audio; however, once we have brought up the name of a person or a company in our pages, we feel obligated to publish their letters, friendly or not, to which we then respond editorially. In the process, a few jackasses may acquire a higher profile than the serious, sincere majority of our correspondents, but c'est la vie. *The letters we print here may or may not be excerpted at the discretion of the Editor. Ellipsis (...) indicates omission. Address all editorial correspondence to The Editor, The Audio Critic, P.O. Box 978, Quakertown, PA 18951.*

The Audio Critic:

...I just want you to know how delighted I am that *The Audio Critic* is back and that it is in top form, too. Bravo! Yours is the only publication that wraps me in "can't put it down reading," like a good novel, as soon as each issue arrives.

The Audio Critic contains consistently the most brilliant writing about audio that I have seen in over 30 years as an audiophile. This has to do, I believe, both with what you say and how you say it. Special favorites of mine are your reviews of recordings. These are clearly from the heart and show you to be as much music lover as you are lover of the science of reproducing music. On the other hand, I am not so much in awe of your opinions that I must run out and buy your recommended components. I think I retain a healthy skepticism.

Whether I agree with you or not, I always enjoy the reading experience itself. Every word, every sentence seems polished to a high luster and fits logically, rather like the notes in a score by a great composer. The overall effect is the perfected effort of a single writer who cares deeply about the excellence of his work. Your publication is obviously not the result of a committee process, which is what the other audio magazines have become.

This brings me to...your statements regarding regularity of publication of *The Audio Critic.* You seem to be putting pressure on yourself to achieve your own notion of a regular publishing schedule. But, please, please do not feel any guilt or pressure to do so.

I realize that pressure can stimulate creativity. It can also cause quality to deteriorate. I have witnessed the transformation of other "underground" audio journals into fat, glossy, regularly published periodicals. In the process, I feel they have become watered down, verbose, and too driven by the need to make a "statement" with each issue.

I, for one, sorely miss the old irregular *Stereophile* in the days when J. G. Holt was "editor, chief tester, and drudge." Then, unlike now, it was written with wit and creativity, focus and consistency of point of view that today can only be found in *The Audio Critic.* JGH's views and yours differ, but the honest and readable quality of both the old *Stereophile* and *The Audio Critic* of today are of the highest caliber.

If pressure to publish regularly causes you to follow in *Stereophile*'s footsteps to what it has become today, then I say to you, don't do it. I am comfortable seeing you publish only when you have something to say that lives up to your standards of excellence. Besides, I like surprises...

Best regards,
Carl J. Weber
Philadelphia, PA

Thank you, kind sir. A letter like this relieves us of the rigors of false modesty; you praise us as expertly as we would praise ourselves if self-praise were permissible. Kind as you are, however, we must disagree with you on a number of points:

(1) We could name certain scores by Bach, Mozart and Beethoven that are more perfectly wrought than a typical issue of **The Audio Critic**. *(2) On the other hand,* **The Audio Critic** *is, and always was, better written than the old, not only the current,* Stereophile. *(3) The most significant difference between the old and the current* Stereophile *is the difference between the minds of J. Gordon Holt and Larry Archibald. (4) The pressure to publish regularly*

is almost purely an economic one; income from subscriptions and ads is derived on a per-issue basis—the more issues, the more income. (Our plan is to progress gradually from quarterly regularity in 1990—yes!—to a bimonthly and eventually monthly schedule in the years to follow, and we have no fear of a lapse in quality as long as we can expand our staff without bringing in mediocrities.)

Your comments also bring back memories of a particularly poignant letter we once received from a man who had quite obviously never read anything in his life except electronics and audio magazines, owner's manuals, and parts catalogs. He was astonished and apparently deeply moved by the quality of English writing he had discovered in our pages; we had somehow become his surrogate Dickens and Joyce. We found that responsibility a little more than we could handle.

Anyway, you definitely have the right attitude, Carl; what we need now is 50,000 more subscribers like you.

—Ed.

The Audio Critic:

It has come to my attention that considerable confusion and/or questions have arisen regarding bit stream, or 1-bit, D/A converters. I wish to share with you a simple explanation that may provide some answers.

The PCM digital audio signal, as it comes off the disc, is nothing like an analog audio signal. Both channels are mixed, the bits are out of sequence (because of the error-correction encoding), and miscellaneous bits are added for track number, time code and copy protection, to name a few.

Conventional D/A converters reconstruct analog signals by creating the "staircase approximations" we have all seen in many ads and articles. The "run" or distance between each "step" is based on the sampling frequency, while the "rise" or voltage amplitude of each step is based on currents running through resistances inside the D/A converter IC. The step voltages are subject to the accuracy of the currents and resistances, and in every sense are analog values.

There are two bit-stream systems now commercially available, and while they are different, they have enough in common for the purpose of this explanation. Bit stream D/A converters digitally, and at very high speed, convert the PCM signal into two digital signals that have much more similarity to analog signals. That is, a digital-to-digital conversion separates the two channels, decodes the error correction, removes the miscellaneous bits, and puts the left and right digital signals into a format that easily lends itself to further simple conversion. The digital-to-digital conversion is accurate, and removes the mechanism for errors and nonlinearity present in conventional D/A converters. The only further conversion required is low-pass filtering, which is inherently a simple linear process.

I firmly believe that bit stream converters will soon be in wide use. The cost of bit stream D/A conversion is reasonable; it requires no production alignments for linearity, and has excellent sound quality and measurement data.

With best regards,
Marty Zanfino
Vice President
Engineering and Technical Services
Harman Kardon, Inc.
Woodbury, NY

An in-depth overview of the state of CD playback technology, introducing an important new contributor to ***The Audio Critic****, will be published in our next issue (No. 15). The article will include, among many other things, a detailed critique of the new 1-bit DAC architectures. So far we have heard both endorsements and caveats regarding this new approach—and from equally authoritative sources. Meanwhile we must state that your explanation of it for the layman is by far the simplest and most lucid we have encountered.*

—Ed.

The Audio Critic:

I own an expensive audio system. I have never told anyone the price. Quite frankly I am embarrassed by its price. For years I laughed at Harry Pearson et al. for recommending expensive equipment. I even purchased the budget system you suggested (Hafler amp, PS Audio preamp). The fact is that I never liked the sound. It gave me a splitting headache. I replaced my electronics with a Moscode 300 and a C-J PV5. Flawed? Yes! But at least I liked the sound. (Quite frankly it was gorgeous.)

This brings me to the ABX comparator. I don't believe that all amps sound the same. I believe that it (ABX) is a signal processor. But if I'm wrong, I'm willing to put my money where my mouth is. I will buy for you the Radio Shack amp and equalizer of your choice and accept your Boulder 500 as an even swap. If that is not acceptable to you, I will donate to *The Audio Critic* the Radio Shack amp and equalizer of your choice. The only condition is that you conduct all your personal listening and professional testing with this amp and equalizer.

While I am sure you will find a way to slip out of this offer, I think your credibility on this issue would be firmly established if you accepted this offer.

In fact, while I am not rich, I extend this offer to Mr. Lipshitz, Mr. Carver and Mr. Clark.

Sincerely,
Reginald G. Addison
Attorney at Law
Washington, DC

CC: *The Absolute Sound*
Stereophile

To begin with, counselor, "splitting headache" as against "frankly gorgeous" does not quite make the grade as expert witness testimony, does it now? But that was not your reason for writing, so we shall let it pass.

As for the ABX comparator, if it is a "signal processor," then so is the source selector switch on the costliest preamps; the basic operating principle is exactly the same. Besides, ABX switching without the box, by means of hand-plugged cables, has been repeatedly proven to yield the same results. Talk about credibility! Today there exists no halfway respectable opponent of ABX testing who attacks the box itself. For better, though ultimately invalid, anti-ABX arguments see our article on the subject in Issue No. 12.

Your prophecy regarding our refusal of your offer is of course self-fulfilling. As you probably know, or must at least suspect, no Radio Shack stereo amplifier can be bridged for mono operation to swing 70 volts into a variety of loads at virtually zero distortion, like the Boulder. Regardless of listening quality, we need that kind of measurable performance for reference in our laboratory. Equalizer? You have lost us there; we hardly ever use one.

Your most fundamental misconception, however, is that we believe—or ever said—that a Radio Shack amplifier will under all conditions sound indistinguishable from a Boulder (or Krell or Mark Levinson). What we believe and are willing to reiterate is this: Any two amplifiers of conventional architecture—very high input impedance, near-zero output impedance, typically low distortion, normal stability—

and operated well within their voltage and current capabilities (very important!) will sound startlingly similar when matched in output level within 0.1 dB. You may end up identifying the expensive amplifier vs. the Radio Shack blind, on the basis of some subtle clue (say, the noise floor or maybe a rise or dip of 0.2 dB somewhere), but your audiophile heart will sink in the process—this is not the night-and-day difference the tweako reviewers led you to expect! On the other hand, the Radio Shack amplifier is much more likely to be driven beyond its voltage/current capability under various field conditions than the Boulder (or Krell or Mark Levinson), and then all bets will be off. Have we made ourselves clear? It should be added that tube amplifiers with their generally higher output impedances drive the speakers a little differently and may therefore sound different to the extent of the resulting spectral shifts and Q changes.

Anent your pointedly listed CC's, we would like you to ponder this: If you wrote a law journal a letter disagreeing with their viewpoint on obscenity legislation, would you then send CC's to Screw *magazine and* Hustler*?*

—Ed.

The Audio Critic:

Thank you for your "Seminar 1989: Exploring the Current Best Thinking on Audio." It's already the most fascinating piece I have read in any of the audio magazines, and I haven't seen Part II yet. *(And now it will be Part III keeping you in suspense because of our last-minute editorial decision necessitated by the monstrous length of the transcript.—Ed.)* I know most of the participants from the AES, and Dave Clark is a good friend. You certainly had the right group assembled for this, and the reason it is so interesting is that it shows what a diversity of opinion there is on how we should "do" audio. In the presence of such an august gathering I hesitate (ever so briefly) to say that I believe I have something to contribute to the discussion.

...Your group...got into the real meat of the matter, which is the whole area of how to do sound reproduction in an ideal way—how many channels do we really need, is Ambisonics the answer, binaural vs. stereophonic, room acoustics, etc. What I have to contribute to this is a new theory for stereophonic sound, a sort of way of thinking about the process, a *model* against which we can test the various ideas for validity. I call it the *image model* theory. It apears at first to be quite a bit different from the usual thinking in audio, and it gets pretty fundamental, so I think the best way to approach this would be to lay it out first, and then go back to your discussion and show how it ties in.

We must first clearly distinguish between *stereophonic* and *binaural*. These are two totally separate systems, every bit as different from each other as, say, sculpture and painting. Binaural is the simpler of the two systems, being nothing more than an attempt to replicate the ear signals that you would have heard if you had been in the same location as the dummy head during recording. You get exactly the same direct-to-reflected ratios, acoustics, everything as if you were transported to the hall or studio. The *stereophonic* system is a completely different approach. Rather than attempt to replicate the *ear signals* that a listener at the original event would have experienced, it starts from the *other* end and reproduces the orchestra itself, and the early reflected sound near the orchestra, in all its glory, right in front of you in your listening room. It is a *field*-type system in which we are physically reconstructing all of the salient characteristics of the original sound fields in the playback space. We tend to prefer the stereophonic system because the sound is real—not an illusion. We can move around, turn our heads, feel the chest-thumping bass, even move closer to or farther from the orchestra. In other words, even though stereo is not an exact facsimile reproduction of the original acoustics and everything, it still wins out in realism because the sound *is* really right there in front of you, and you can use your natural hearing and move around. Binaural is like a 3-D photograph of Jack the Ripper on the streets of London. Stereo is like being eyeball-to-eyeball with a sculpture of him in the Wax Museum. Two completely separate systems, OK? The reason I dwell on this is that the confusion between the two systems arises when audio theorists believe, or start with the assumption that, the object of "accurate" stereophonic reproduction is to reproduce the two channels as signals at the ears. This approach manifests itself in attempts to kill all room reflections or to put all of the drivers of a loudspeaker on one surface and aim them at your face. This is just the opposite of what needs to be done to get good sound, which brings me to my image model theory.

What we are going to do is to study the pertinent audible characteristics of a live sound field in order to see how to make the reproduction more like it. There are three such characteristics which can be distinguished: spectral, temporal, and spatial. We can study these characteristics by dividing up the arriving sound into its three main temporal stages: the direct field, the early reflected, and the reverberant sound. The direct sound arrives straight from the instruments and has a flat spectral balance. Spatially, the direct sound establishes the lateral localization of the sources. Next to arrive at the listener is the early reflected sound, from the front and side walls of the concert hall. The easiest way to understand its spatial nature is to make a drawing of the *image model* of the instruments and their first and second reflections. The reflections appear as additional sources behind the front and side walls (see drawing). The early reflected sound has two jobs: it provides us with a complete "view" of the radiation pattern of the musical instrument (giving us more information about what the instrument sounds like in an enclosed space than outdoors, for example), and it gives us the *spatial impression* that is so important to the enjoyment of music. This early reflected sound is stereophonic in nature, and the pertinent audible characteristic is a spatial broadening of the orchestra, lending a richness and depth to the sound. The full reverberant field, caused by all the remaining reflections in the concert hall, is virtually monophonic or nondirectional. It is fed by a buildup of the more steady-state tones, which tends to roll off the high frequencies above about 1 kHz and contributes to the "musicality" or sweetening of the sound.

So the spatial "shape" of live sound is a dash of direct sound, a very important and very stereophonic "splash" of early reflected sound from front and side walls, and a virtually nondirectional decay of reverberant sound. Now, if we look at the reproduction, we can draw an image model of loudspeakers in a playback room just as easily as musical instruments in a concert hall. Image model theory says that our goal should be to make the spatial shape of the playback, or the playback image model, as close to that of the live sound as possible. The theory sees the reproduction as a model of the real thing, with speakers pulled out well away from the walls, first-arrival sound coming directly from them, followed by an early reflected field of similar intensity and shape to live sound, and with the room itself (aided by surround speakers) building a reverberant field for playback just as the concert hall did live. To make a long story short, what I have found is that if this is done right, it will actually *decode* the spatial information contained in the recording much better than, say, a pair of highly directional speakers in a dead room, which tends to take all of the recorded spatial information and compress it so that it can come from only a limited set of incident angles, as defined by the separation of the loudspeakers.

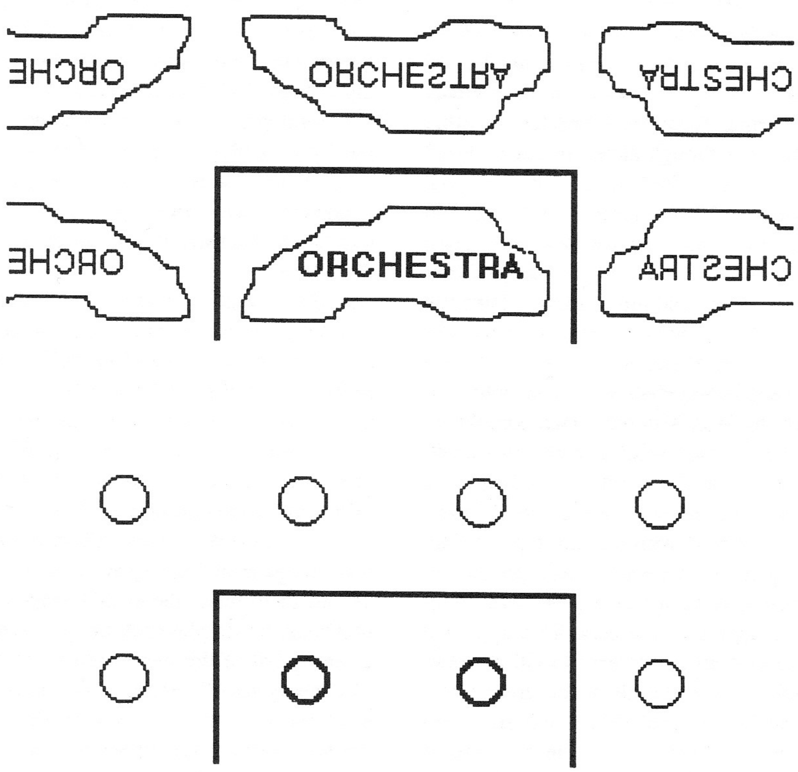
In any case, if you at least understand what the image model of live sound looks like, you can more easily see the advantages/disadvantages of various theories of reproduction or proposed systems…

On the question of whether two channels are sufficient or whether we need umpteen, I would point out that two are sufficient for the enjoyment of music. The stereophonic *system* as such has no limitations on the number of channels, and you could use one for each instrument if you thought it worthwhile, but such precise localization of the sources has nothing much to do with the appreciation of the musical message. Yet two channels contain all the information needed to generate lateral localization of the direct sound and all of the spatial patterns that are so important to the feeling of spaciousness and being immersed in a live sound field. Since the full reverberant field is monophonic, the only need in the reproduction is to fill out the back half of the sound field with something that is delayed with respect to the frontal sound. The difference (L – R) signal is just terrific for doing this without causing an echo of the center front soloist, and discrete surround systems are superfluous. I believe that a standard, passive Dolby surround decoder is all that is needed. You can use as many speakers as you need to even out the surround field, but stereophonic capability is not required in the surround channel—the stereo effects are completely taken care of by the front speakers and the early reflected sound.

On pp. 24-25, Stanley says that to begin to standardize some sort of ideal system, we should begin with how we hear—talk about the ear signals. Again, this would be a binaural system, and Stanley knows that, but then he goes on to imply, as have Duane Cooper, David Griesinger and others, that we could record binaurally and then process it over to stereo or anything we want. I definitely do not agree with this—the two systems are totally separate and incompatible. Binaural and stereophonic recordings contain completely different *kinds* of information, from different perspectives, and for different goals. A third possible system that he may be thinking of is one where you record binaurally and then play back on speakers, using a crosstalk cancellation circuit (Sonic Holography or Polk SDA) in a relatively dead room. That's a terrific idea, but it's still binaural (loudspeaker binaural), and you have to sit still in just the right spot, etc. But this is *not* a stereophonic system, nor will it translate to one. And while I'm on this subject, it is just as incorrect to play stereo recordings on loudspeaker binaural, which means that Bob's Sonic Holography is a fine thing but is only correct for recordings that were made binaurally. Interaural crosstalk is *not* a problem in the stereophonic system.

…[Regarding] Dave Clark's concept of standardizing the reflection signature of the playback situation…[let me] add that when it comes time to get serious about this stuff and sit down and completely specify the temporal, spectral, and spatial characteristics of the reverberation signature that Dave is talking about, we cannot do any better than to look at an image model of live sound. The image model completely specifies the intensities, direction of arrival, spectral content, time signatures, axial vs. power response, *everything* about all of the sound patterns that exist in the concert hall.

…If you agree with me that stereophonic sound is a real, physical model of live sound, then perhaps you would also agree that the four basic *parameters of fidelity* that we *can* manipulate are (1) physical size, (2) power, (3) accuracy of the storage and transmission in the electronic domain, and (4) spatial characteristics. As John Eargle said, we no longer have a problem with accuracy in the electronics. Paying more money for a CD player will yield no material improvement in your sound, and the only characteristic of amplifiers we should be interested in is how much *power* they have. Given a choice between a $6000 dilettante audiophile 100-watt amplifier and a Carver M-1.5, definitely opt for the Carver. It would take thousands and thousands of watts of power to come close to the power of live sound, so the more the better. The physical size of the playback room is very important, larger rooms sounding closer to the real thing because the scale of the model is closer to life size. I'm not sure yet what the upper and lower limits are, but most reviewers are using rooms that are too small for high-



Comparison of Reproduction Image Model to Live Sound
The spatial "shape" of the sound that we actually are hearing is a function of the positioning and intensities of all acoustic images with respect to each other. In the reproduction, speaker positions and direct-to-reflected ratios are adjusted to better approximate the live model.

fidelity use. Finally, the spatial characteristic refers to the ratios of direct to early reflected sound in the playback image model and the building of a realistic surround field in your room. It is manipulated by means of the radiation pattern of the speakers, the positioning of the speakers in the room, and the acoustics of the room. This is where the intense research is beginning to go, and it's about time.

Gary C. Eickmeier
Independent Industrial Designer
Lakeland, FL

We found it necessary to trim down your excessively long letter but trust that we have managed to retain the essence of your argument. We have also read the preprint of your October 1989 AES paper, "An Image Model Theory for Stereophonic Sound," which you were kind enough to enclose and which reveals that your theory is the basis for a new loudspeaker design you are in the process of developing. That puts things in a somewhat different perspective, making your argument a design rationale rather than a disinterested audio philosophy.

Even so, we find little if anything to contradict in your statements; everything you say is basically true, much of it widely recognized as such, but only your finished speaker is likely to convince us that yours is a higher truth in a discipline characterized mainly by trade-offs.

We have some minor quibbles; for one thing, your views on binaural vs. stereo seem to us a bit too black-and-white. There is an occasional overlap or in-between gray area. In 1959, when commercial stereo was very new, we attended a couple of the then-celebrated André Charlin's recording sessions for Erato in Paris. He used a single pair of Schoeps omni capsules on each side of a football-like (oblate spheroidal) baffle. Nothing could have been more like a dummy head, yet it did not occur to him call these recordings binaural; they were considered the very finest in stereo. In Part II of the seminar transcript, in this very issue, you will find some comments by Stanley Lipshitz and Peter McGrath about the use of such baffles between closely spaced omnis in stereo recording. Nor is Bob Carver's Sonic Holography strictly "loudspeaker binaural" anymore; he has over the years added a few little tricks and wrinkles that widen the sweet spot for listening and make the system do its thing with a larger variety of recordings. We agree with you, however, that the makers of a stereophonic recording presumably anticipated *the four different arrivals (L speaker to L ear, R speaker to R ear, L speaker to R ear, R speaker to L ear), of which Bob tries to wash out the last two, so that his processed wave launch is really an alteration of what we were* intended *to hear. We have been giving Bob a hard time about this for years, but like most highly creative individuals he has limited use for purists. On the other hand, you should never, never mention in the same breath the simple-minded copycat system inextricably embedded in the Polk SDA speakers.*

Overall, we feel you are headed in the right, or at least a *right, direction and are eagerly looking forward to the debut of your new loudspeaker.*

—Ed.

The Audio Critic:

...Your magazine articles are very good, but the "Seminar 1989" is the very best I ever read in any hi-fi magazine to this date. Thank you very much!

Very truly yours,
Fermín Avilés
Brooklyn, NY

We are flattered by your enthusiasm, but think about this: The mainstream hi-fi slicks have no room, and certainly no editorial niche, for anything as lengthy and leisurely as our seminar. The tweako/dilettante high-end journals, on the other hand, lack the intellectual credibility to attract panelists of the caliber of ours. That leaves the field to us, pretty much by default.

—Ed.

The Audio Critic:

Upon reading the article on speaker placement for best bass response in Issue No. 13 of *The Audio Critic*, I am struck by the fact that all too few audiophiles and industry people realize how to point a loudspeaker, and where to sit and position their transducers for optimum sound reproduction.

The key word we should pay attention to is reproduction. A loudspeaker is a very simple electromechanical device for pushing air. As such, there are different axes on which a listener may fire the drive elements into the room.

Home hi-fi is now mature enough to hope that audiophiles might understand that a speaker's smoothest delivery in typical rooms is on axis. In other words, the acoustic center of the speaker should fire at a centrally located "sweet seat."

In reality, however, most dynamic loudspeakers comprising multiple drive elements actually suffer a bit of crossover interference on axis. The best designs we are associated with sound smoothest 15° off axis or crossing directly behind the listener's head, for central imaging and a clean, open presentation of the soundstage. As my seven-year old daughter would say, "It sounds more better, Pop."

Oftentimes in travels to stores and fellow audiophile homes, I see speakers firing straight down the room. In other words, the listener is making value judgments based on 30° to 60° off-axis speaker response, which is any speaker's worst air movement. Off axis there can exist large peaks, broadband dips, and highly colored resonant ringing cycles. It is largely due to this off-axis listening that I believe many of the digital-haters have formed their opinions, in order to obtain a soundstage from their most presence-dominant vinyl discs. Perhaps making judgements using highly touted transducers with abysmally nonflat frequency responses has aided in colouring their viewpoint, too!

All speaker manufacturers should explicitly state in the owner's manuals the optimum listening axis and height for normal distances in ordinary living rooms (14' by 22' approx.). But then that would be agreeing to minimum standards and applying some measure of science. Yet isn't acoustics just that—an applied science?

Sincerely,
John Ötvös, President
Waveform Research
Ötvös Industries
Brighton, Ont., Canada

The Fourier 8e loudspeaker system, a 3-way design in which your Editor was heavily involved, used a mathematical compromise between optimum on-axis and 30° off-axis response to determine crossover network parameters and driver level matching. So, as you can see, some speaker designers do anticipate the problems you are talking about. (A fat lot of good it did in the case of Fourier.) The user should never assume automatically that the speakers must be toed in. It all depends on the design, and—we agree—the manufacturer should tell us about it.

—Ed.

The Audio Critic:

Mr. Rasnake's article on speaker placement in Issue No. 13 contains implications which I feel deserve some further comment.

I believe that Mr. Rasnake's article insufficiently emphasizes that the listening room, including the positioning of loudspeakers and listeners, is almost always the weakest link by far in the chain of recording and reproduction, and thus any improvements that can be achieved, however much a compromise they may be, are probably more significant than other possible improvements in equipment or recording quality. Unfortunately, the room is likely to be the most expensive "component" and also the costliest to change. This is not to criticize "mere positioning" as a "cheap fix."

In the opening paragraph he states that his analysis would be "especially useful if the room has not been built yet." However, that statement contains the assumption that people would deliberately build listening rooms with rectangular shapes and pairs of parallel surfaces. Apart from building a cubical room, that is the least desirable way to build for good acoustics using flat surfaces (assuming that curved surfaces are more costly to build and thus not considered). Even though perhaps fewer than one room in a million is delibrately built with no parallel flat surfaces, this analysis is really merely an attempt to make the best of a bad situation, caused by long-standing architectural and aesthetic prejudices, and the tyranny of T squares and triangles. It is not necessary to have walls that are out of plumb or sloping floors to achieve nonparallelism. According to *Acoustical Designing in Architecture* by Vern O. Knudsen and Cyril M. Harris (of the firm that did the final fix on Avery Fisher Hall at Lincoln Center), as little as 5% offset is sufficient to prevent flutter echo, a related problem (1950 edition, pp. 171, 187). This would mean, in a 20' long (or wide) room, as little as 1' narrower at one end than the other, or 6" wider or narrower at the middle compared to the ends. Similarly, a 10' ceiling would be dropped 6" at one end or side, or 3" at both ends (or sides) or across a center line. This may be enough to be noticeable, but not enough to make anyone dizzy or disoriented, or cause problems with the use of conventional furniture, or cause the loss of any significant amount of space.

Clearly then, creating a good listening room does not require the duplication of Berlin's Philharmonic Hall in miniature, or anything even remotely so exotic architecturally. The fact that fairly simple mathematical analyses can be applied to fairly simple room shapes is also no argument for using such shapes just to be sure that "at least we know what we've got." More complex shapes for the most part have less need of the necessarily more complex analysis until one reaches the size and complexity of large auditoriums, with the likelihood of some curved surfaces, overhanging balconies and, of course, high costs of construction or renovation.

Rasnake says very little about positioning the listener, except that listener distance to room length should be within a range related to speaker to room length distance. But just as speakers should not be placed near low-bass resonant areas, neither should the listeners, if the smoothest response is sought. This should be in terms of length, width, height, and diagonals, and for stereo equidistant to and with an angle of from about 40° to 60° between the speakers. But the mirror image positioning he suggests for the speakers would place the listener midway from left to right, rather than the more desirable splits of from 1/5, 4/5 to 4/9, 5/9 with 1/3, 2/3 the best. To achieve this the speakers cannot be positioned as mirror images, and either *Y* and *Z* dimensions must be interchanged or two different solutions used. The benefit achieved will hopefully be greater than the differences introduced in left and right channel response.

Thus the best arrangement for loudspeakers and listening position(s) in a symmetrical room will be asymmetrical. One must choose between visual aesthetics and aural smoothness or modify one's aesthetic standards.

Perhaps it should be noted that the proper placement of loudspeakers and listening position(s) is not going to eliminate room resonances, much less turn your room into an anechoic chamber. But it will allow you to avoid undue excitation of resonances and undue exposure to them, particularly in the low bass, which is where they cause the greatest response irregularities. Recordists should also use Rasnake's method to position microphones and musicians, and conductors should use it to place instruments with deep bass capabilities.

In the usual size of listening room, say 16' wide, the difference in distance between the listener's ears, say $5^3/_4$", is enough to place the two ears at different nodes, such as 1/3 and 4/11, with 5/16 and 3/8 close by on either side. Although Rasnake's Figure 3 only divides the room into eighths *[must be meaning Table 3, but those divisions are not eighths—Ed.]*, this represents only the fundamental and first two harmonics, whereas his Figures 1 and 2 deal with harmonics through the fifth *[actually sixth—Ed.]*. However, as 16' represents about 35 Hz, no more than eight times this frequency is likely to be critical, perhaps as little as four times it. If only resonant areas two feet apart need be considered, then probably 1/7, 1/9, and 4/11 should be eliminated from consideration *[Rasnake never mentions 4/11—Ed.]* in at least the shorter two dimensions (of the average listening room), as nodes and resonant areas are so close together as to create areas of steep pressure differentials and thus too critical of exact placement for practical purposes.

Node and decimal		Decimal and resonance		Inches difference in 16 feet
1/9	.111	.125	1/8	2.67
1/7	.143	.125	1/8	3.43
4/11	.364	.375	3/8	2.18
but the following are somewhat better				
1/11	.091	.125	1/8	6.55
2/9	.222	.250	1/4	5.33
2/7	.286	.250	1/4	6.86
2/5	.400	.375	3/8	4.80

Apparently the sideways placement of the head in an ordinary room is very critical, and one might also wonder about the use of loudspeakers with woofers larger than 8" in diameter (effective cone diameter 6"), or considering a $7^1/_2$ to 8-foot ceiling, larger than 4" (effective cone diameter 3"), as at least one part of the cone would be at an undesirable location in any case.

Anyone who considers how close in actual distance the above points are in an average room may well despair of any ideal or easy "fix" for the problems caused by parallel surfaces.

It is not that one avoids exciting or hearing resonances by locating at thirds or fifths, or avoids power-sapping nodes by avoding halves, fourths and eighths, but rather that by using other ratios these occur at higher frequencies where the more frequent occurrence of resonances tends to smooth out the overall effect. There are no "dead" spots, only many which are "artificially enlivened" and some which are fairly normal at low frequencies. Any position will be for some frequencies a resonant area and for others nodes of little or no pressure change. But some positions have far more resonant frequencies and/or nodes than do others. One might say that different points have widely varying frequency response, and none are flat.

Sincerely,
John F. Sprague

We passed your extremely interesting letter on to Bill Rasnake and asked him to write a reply. He said he would but he never did, even though we gave him plenty of time. It is true that he is a very busy guy, professionally as well as avocationally. Maybe we can publish some comments by him in the next issue. He did say something to us on the telephone to the effect that once the speaker loads into the room without exciting significant resonances, you will not then hear such resonances by sitting in a certain spot or holding your head in a certain position. He also agreed that the differences you analyze are indeed problematically small in small rooms.

—Ed.

The Audio Critic:

After 22 years in the audio industry, I have decided to place my first advertisement in your audio publication. I have never advertised prior to this point, in the belief that when one is involved in fundamental research on the level that I am, the product is so innovative and of such high caliber that it should sell itself.

While other audio manufacturers gloss over engineering deficiencies in their products by spending large amounts of funds on advertising, in the hope that such deficiencies will not become apparent until they have realized a short-term profit, I have put all that money into research and development. I cannot put all the blame on most of the audio companies either, because of the "no ad, no review" policies of some audio magazines. However, your publication from the very start has repeatedly given me excellent reviews on all my products because of creativity and their merits. I now feel that the maturation of my audio line is at a point where I can increase my product exposure through advertising in your respected audio journal, *The Audio Critic.*

Sincerely,
Sao Zaw Win
CEO
Win Research Group, Inc.

We agree that we are quicker to understand creativity such as yours than the dilettante cult journals. Also, at our full-page ad rate of $335, the temptation to be venal is pitifully small, even if we had no journalistic ethics whatsoever. But the best reason to advertise in our pages is that some of the finest and most influential minds in audio will be exposed to your ad.

—Ed.

The Audio Critic:

"Dear" Peter,

Attacking the person instead of his or her ideas is not a new technique. Its *[sic]* been used by intellectually and emotionally (in your case add financially) bankrupt individuals throughout history. Before you engage in such ugly tactics again in regards *[sic]* to me, please spare your reader(s) your petty delusions and send for a resume. You'll find I have a degree from Cornell University and an Academy Award nominated motion picture soundtrack to my credit, among other accomplishments.

But even if was *[sic]* a high school drop out, my credibility as a reporter and as a knowledgeable, keen-eared music and equipment reviewer in the high end audio industry is by now well established. I have no doubt that as your bratty, self indulgent, undisciplined, indefensible slur circulates (oozes up from the muck is more like it), you'll be hearing about it from other members of the high end community.

That out of the way, perhaps we can discuss digital recording, CDs and my *Music Connection/Goldmine* commentary, which by the way, was never submitted to *The Absolute Sound*. Another delusion of yours. The piece was meant to be provocative. How else do you fight four years of "perfect sound forever" out and out lies in the general press? I wanted to prevent *Goldmine* readers from making the mistake of their lives: trading in original pressings for CD reissues.

When I wrote the piece for *Music Connection*, "laser rot" was indeed a very controversial topic. *Billboard* was covering it weekly and an executive for Nimbus admitted that the inks they were using had indeed eaten through discs. How many other companies were using such inks? No one knew then.

There is now a glut of CD production facilities. Some plants have gone under. Competition is severe and wholesale prices are dropping. So is quality. Substandard production is yielding discs with thin and sometimes incomplete coatings. I've seen them and I question their longevity. Time will tell.

Stale canard? I think not. But if I was writing the piece today, I'd concentrate on the artificial sound. I've yet to hear a digitally remastered analogue recording on CD that sounds as good as a well mastered original analogue record. The problem is not with the analogue source. Its *[sic]* with the digital mastering equipment.

You delude yourself again if you think there's unanimity in the recording industry regarding digital recording and CDs. I've interviewed many engineers and CD masterers who admit both on and off the record that digital recording has many problems, particularly at the frequency extremes, and I've quote them in *TAS*.

Keith Johnson's CDs? I've heard them. Pretty good for CD. But I've also asked Keith Johnson about them. You obviously havn't *[sic]*. He'll tell you he thinks his LPs sound better. What's more, he'll detail for you—complete with oscilloscope tracings—why there are serious problems with today's CD technology that cause the sonic deficiencies I, and other discerning listeners hear, and find thoroughly objectionable—moreso *[sic]* than the problems we hear with analogue.

As for Tom Jung's CDs, to me they sound thick, dull, and airless—nothing at all like real music. The drum sound he goes for is sterile and gimmicky. His preferred mix—panning the drums across the soundstage, is to my ears, just plain silly. DMP recordings have as much to do with musical realism as did Enoch Light and The Light Brigade's Command recordings from the fifties and sixties. But unlike you, while I criticize Jung's taste, I don't question his credentials or his intelligence, or his "right" to record music as he sees fit—or his preference for CD.

I stand by every other characterization of the current state of digital recording and compact disc sound that you quoted and I really don't care whether I'm in the minority or the majority. I know what I hear, and I have too good a track record at this point to care about majority rule. Reagan was president for eight years.

Which brings me to the part of your bilious rant that does disturb me: your implication that any publication that chooses to run a piece that contradicts your opinion is somehow suspect. Tantrum? Credibility? You had better look in the mirror.

Sincerely,
Michael Fremer
Senior Music Editor: Popular
The Absolute Sound
Sea Cliff, NY

P.S.: You owe me a half a year subscription to your magazine from 1979.

The above is printed exactly as you wrote it, with all errors unedited; we have italicized the names of publications for the sake of clarity but made no other effort to make a hostile letter appear more literate

(continued on page 51)

Seven Loudspeaker Systems (If They Are All So Accurate, How Come They All Sound Different?)

We examine a varied assortment of up-to-date speaker designs, none of them less than good, none of them perfect, and try to sort out the advantages, disadvantages and trade-offs of each design approach.

The great paradox of loudspeaker design, as we have stated before, is that designers keep trying to come up with the perfect loudspeaker when there is really no accepted theoretical model of perfection. How can you achieve the best possible result without a clear definition of what the *ideal* result is? Amplifier designers have their straight-wire-with-gain ideal; tuner designers aim to make their units perform as if hard-wired to the broadcasting studio; but do speaker designers have anything nearly as definable and verifiable to aim at? Not to our knowledge.

Yes, we have been known to mutter on occasion that the output of an acceptable loudspeaker must resemble its input, but that is a somewhat flippant truism of limited usefulness, far short of a mathematical model. An amplifier channel has one input and one output, so you can easily ascertain the resemblance between the two. A loudspeaker has one input but *n* outputs, making the I/O comparison a bit more involved, alas. For example, the usual axial frequency-response test, although as good a starting point as any, is simplistic and insufficient; two different speakers with highly similar axial responses will in all probability sound totally different. On the other hand, no speaker with a ragged axial response can possibly sound first-rate, so the test is valid in a limited way and needs to be performed. Speaker evaluation is not a discipline for one-track minds.

It is far beyond the scope of this article, indeed of this publication, to bring order to the unsettled arguments for and against the point source, the line source, the figure eight, and various other Platonic forms promoted by their lobbyists as the model for the perfect loudspeaker. We must leave that to mathematicians like Stanley Lipshitz (who got very engrossed in a mathematical discussion of ribbon transducers with Bob Carver during lunch break at our Seminar 1989). Meanwhile we cannot ignore the fact that every speaker reviewed here sounds different from all the others, and not just in bass response and dynamics, which we expect to be dependent on size and usually price.

This Heinz-57-varieties syndrome—always a different flavor even in the frequency and dynamic range common to all good speakers—is one of the great remaining frustrations in audio, and we must emphasize that we are talking about the genuinely good designs, not the failures. What we have been saying about the convergence of the latest audio components toward total waveform fidelity to the recorded signal, and therefore very similar sound, is true only up to the speaker terminals. Past that point there is still a free-for-all, so much so that all generalizations about it are somewhat precarious, unless we restrict ourselves to narrowly defined categories such as forward-firing 3-way systems. We therefore prefer to discuss the possible reasons for each loudspeaker's sound separately, as related to its specific design.

Audio Concepts "Sapphire" with "Saturn"

Audio Concepts, Inc., 901 South 4th Street, La Crosse, WI 54601. "Sapphire" floor-standing 2-way loudspeaker system, $849.90 the pair as full kit (parts kit plus assembled cabinets). Matching pedestal, $99.00 the pair. "Saturn" subwoofer, $639.90 the pair as full kit (parts kit plus assembled cabinets). Passive crossover, $199.90 the pair. Tested samples on loan from manufacturer.

These very interesting speaker designs are available only in kit form, but the crossovers and cabinets come fully assembled, so that even the lightweight do-it-yourselfer can progress from shipping cartons to a listenable stereo pair in a matter of hours. We are basing that statement on our judgment rather than our experience, Audio Concepts having sent us finished units for testing—a special accommodation to the Fourth Estate and not without its own pitfalls, as we shall see.

Let us address the Sapphire first as a self-contained 2-way speaker system of almost, but not quite, full range, in the idiom defined by, say, the Spica TC-50 at one end and the overpriced Celestion SL-600 and SL-700 at the other. If that kind of speaker is your idea of happiness, the likelihood is that the Sapphire will make you very happy indeed. It is a very well-thought-out example of the genre. The two drivers around which it is designed are made in France by Focal, a company in the forefront of diaphragm and voice coil technolgy. The woofer is a 7" unit with die-cast frame, dual voice coil (remember the Watkins woofer?) and Kevlar sandwich cone—very high-tech (or at least very sexy). The tweeter has a 3/4" inverted dome, also of Kevlar. The crossover is first-order, with all the advantages (coherence, etc.) and disadvantages (higher distortion, etc.) of that piece-of-cake approach. Air-core L and polypropylene C, of course; Audio Concepts will not offend against the high-end code. The enclosure is an upright wedge of 0.45 cu. ft. internal volume, a bit like an elongated TC-50, but there is more to it than that.

What distinguishes the enclosure from conventional designs is that the front and two sides of the wedge—in other words, the entire surface of the enclosure except the back and the bottom it rests on—are covered with 2" thick acoustical foam; only the diaphragms of the drivers are left free, even the mounting plate of the tweeter being covered with a felt ring. The result is the most complete possible suppression of early reflections and edge diffractions—and what a difference that can make! Before we go into details, we must add that the naked enclosure with the exposed foam covering rates rather low on the wife-acceptance scale and that, on the other hand, the handsome Vandersteen-esque hood assembly makes the essentially satellite-sized speaker look quite large and also introduces a tiny resonant coloration. We left the hood off in our tests but had the speaker mounted on the optional pedestal, which also adds height and bulk.

As for the performance of the Sapphire, we would call it highly respectable on all counts and just about state-of-the-art in imaging. If you are an open-minded imaging freak (or is that an oxymoron?), you owe it to yourself to check out this speaker. We have other priorities; tonality happens to be more important to us than directional and spatial clues, although we love to have it all in one speaker; nevertheless, when the definition of the soundstage is as clear as it is with a pair of Sapphires, we are disarmed and delighted. Obviously, by comparison, conventional boxes radiate a lot of garbage off their fronts and sides almost simultaneously with the diaphragm-launched signals, hence the difference. It should be added that the acoustical foam covering of the Sapphire is deployed in such a way as to leave just enough clearance for the drivers' desired solid angles of radiation, so that the tonal quality of the first arrival is in no way compromised. In fact, the overall tonality of the Sapphire, once the obvious small-speaker limitations are discounted, is quite excellent; in a deliberately cruel side-by-side comparison with the Quad ESL-63, for example, we heard surprisingly small differences in instrumental timbre. The crossoverless Quad sounded more seamless in the low/mid/high transitions and marginally more neutral, the Sapphire erring perhaps on the bright side, not at all unpleasantly—but there was no wipeout! Enough said.

In our laboratory tests, the most outstanding characteristic of the Sapphire was its response to square pulses, which we swept from 1 ms width to about 125 μs. We have never seen more coherent waveshapes and fewer trailing squiggles coming out of any loudspeaker with a crossover. Only the Quad ESL-63 and the Carver ribbon (crossoverless above 100 Hz) are as good or better in this respect. We were impressed. The frequency response of the Sapphire is less impressive; we have seen flatter and smoother curves, although there are no egregious anomalies. The overall trend of the axial response is ever so slightly upward; there is a possibility that the tweeter level is set a smidgen too high. Furthermore, the tweeter by itself (Kevlar, schmevlar) is surprisingly ragged in response, with a pronounced peak at approximately 17 kHz; still, it works within the system, so why argue with success? The woofer is nice and flat; in this box it rolls off at 12 dB per octave below 80 Hz with an apparently perfect Butterworth contour. A little bit of positional juggling with respect to the corner distances (see Bill Rasnake's article in Issue No. 13) could boost that bass response just enough to make the Sapphire a stand-alone system for the bass-shy contingent—you know, those who think that deep, flat bass is somehow not "tight" enough.

And that brings us to the Saturn subwoofer, which is specifically offered by Audio Concepts as the ideal low-frequency extension of the Sapphire to build a super loudspeaker with down-to-22-Hz response for less than $1800 the pair. The Saturn is designed as a sealed column system of 2.5 cu. ft. internal volume, with two face-to-face mounted 12" polypropylene-cone drivers firing downward and upward at floor level in a push-pull-wired configuration. This is a genuinely valid space-saving and distortion-canceling idea, but it appears that we were sent the wrong samples for testing. Ours rolled off at 12 dB per octave below a –3 dB point of 42 or 43 Hz, much too high, and the passive crossover supplied was strangely mismatched to the system, creating a huge upper-bass hole between the subwoofer and the satellite, among other problems. Discussions with Jack Caldwell, designer of the Sapphire and Saturn systems, and Mike Dzurko, president of Audio Concepts, elicited two possible explanations: (1) the ad hoc kit builder provided to us as a favor had overstuffed the subwoofer enclosures with damping material, thereby choking the deep bass, and/or (2) a bad batch of 12" drivers with shifted electromechanical parameters had slipped through. We can believe the kit-building error because we caught one in the Sapphire, too: the wires to the woofer had been reversed in one unit, with disastrous effect on phase integrity and, until the error was corrected, on our opinion of the sound. We can also believe the incorrect-driver theory because it would explain the mismatch to the crossover. In any event, the subwoofer we tested was not what Audio Concepts would presumably sell you in the future; on the other hand, we can only report our actual experience with it. For the moment we are putting the Saturn on hold.

The Sapphire, however, is highly recommended, and before we forget we must also acknowledge the good looks and very solid construction of the various prefab blond-oak cabinets we were sent.

Carver "Amazing Loudspeaker" (Platinum II Edition)

Carver Corporation, P.O. Box 1237, Lynnwood, WA 98046. "The Amazing Loudspeaker" (Platinum II version), $2195.00 the pair. Tested preproduction samples on loan from manufacturer.

The Amazing Loudspeaker—we still wince a little as we use that embarrassing name—is emerging as a classic design, regardless of who agrees or disagrees with our high opinion of it. A classic design is one that totally defines a viable new format, and in that respect the Amazing joins the ranks of speaker systems like the Klipschorn, the AR-1, the Quad ESL and ESL-63, the Dahlquist DQ-10, the DCM Time Window, and just a very few others. Absolute performance is not the issue here here; the point is the originality, elegance, and long-term validity of the design concept. For example, the Duntech Sovereign 2001 may be a very good speaker but it is a trivial design and therefore not a classic, whereas both Quads, neither of which can do everything the Duntech can, are all-time classics—and so is the Amazing, which incidentally can do it all, especially in this new version.

Let us trace all the versions for the sake of clarity. What we reviewed in Issue No. 11 was more or less the first generation, although very slightly different earlier samples had already been demonstrated and sent out to just a few people. Then came the Platinum Edition (small wince), which represents the second generation and was never reviewed by us. It introduced a ribbon with corrugations, or crinkles, to kill small resonances and with twice the effective length of the previous ribbon through longitudinal splitting. The efficiency went up 6 dB as a result, and the woofers were modified to match that efficiency. That was a significant improvement, our main reservation about the original speaker having been its somewhat low efficiency compared with box-type speakers. An optional electronic control box completed the Platinum Edition, providing all kinds of signal-processing tweaks for those who wanted them, including the latest version of the Carver Sonic Hologram. To complicate the picture, a smaller version of the Amazing, called the Silver Edition (smaller wince), made its appearance at the same time, also with the newer ribbon technology and with greater dependence on the electronic control box than the bigger speaker (a crucial detail ignored in some dealer demos). We understand that a Silver II Edition is also in the works.

Now—let this be clearly understood—what we are reviewing here is none of the above. It is the Platinum II Edition, which is the third-generation version of the *big* Amazing, reviewed nowhere else so far and scheduled to replace its predecessor on the production line and in the stores after January 1, 1990. We tested the "beta version" (to borrow software terminology) of the Platinum II, i.e., prerelease samples intended to be totally identical to the production units. We had only a limited time for our tests, but we are satisfied that these were thorough enough to support valid conclusions. Our samples, incidentally, were in "natural oiled oak," which we happen to like a lot less than the original high-gloss black finish. That, too, will continue to be available, but at a higher price. Marketing, marketing.

There is a reason for the short life span of the Platinum as the top of the line and its rather sudden replacement by the Platinum II. Bob Carver came to the realization at *Stereophile* headquarters in Santa Fe, New Mexico, that the ribbon misbehaved at their altitude of 7000 feet, developing various peaks in its response as a result of the reduced air load. This affected only an infinitesimal fraction of the speaker's users, but a fix was definitely in order. (A particularly poignant footnote to the story is that all *Stereophile* tests of loudspeakers using thin, lightweight diaphragms—electrostatics, ribbons, leaf tweeters, small domes, etc.—are now necessarily under suspicion, as are their listening evaluations of other components through such speakers. The altitude effects may conceivably have skewed all of their findings.) The Platinum II is the first Amazing, according to Bob, to incorporate certain small design changes that make it altitude-insensitive. Otherwise it is basically the same speaker as the Platinum, with only minor refinements. The crossover network has been very slightly modified, mainly to accommodate two new "voicing" controls in the back, one for the upper midrange (2 to 5 kHz), the other for the high frequencies (6 kHz and up). Another small improvement is a new grille-cloth frame for the back of the woofer array, designed to eliminate residual interference with the rearward radiation of the ribbon. Overall, the big design changes took place from the first to the second generation; however, we are glad that we can report to our readers about the more polished third generation instead.

The basic design principles of the Amazing were explained in our original review and further discussed in our letters column; we therefore have no intention to go over the same ground again, although said principles have not yet penetrated some of the thicker skulls in high-end audio journalism. Let that be their problem, not ours. We have not quite recovered yet from our amazement (no wince) over the stunning simplicity of Bob's solution to the problem of obtaining deep, flat bass from an open-baffle system. The Platinum II produces even better bass than the original—and it ought to, with the redesigned woofers and 6 dB increase in efficiency. The bottom notes of the Zürich Tonhalle organ on the Dorian recordings of Jean Guillou come through with astonishing power and truly clean delineation, putting many a separate subwoofer to shame. The Waveform speaker system (see below) has comparable, but not necessarily better, bass performance; unfortunately the logistics of testing very large loudspeakers prevented us from making a side-by-side comparison. We have yet to test a third system even remotely in the same class basswise. The revised ribbon is equally impressive, very similar to an electrostatic in transparency but of course quite different in wave launch, and super smooth. The somewhat warm and mellow balance of the original Amazing has been changed to a slightly brighter, crisper, more neutral characteristic, which we definitely prefer; the two controls in the back can vary that within certain limits. The Q of the woofers can also be varied, simply by substituting resistors for a shorting wire at the input terminals in the back; one can choose a Q of 1.1 (as delivered, with the wire in place) or 0.7 (maximally flat, our preference) or 0.5 (critically damped, for the tight-bass contingent). With the optional electronic control box the wire is left in place, and the Q is then variable from 0.3 to 2, the straight-up position being 1.

Our measurements confirmed our general subjective impressions. The frequency response of the Platinum II is extremely flat and smooth, without any significant peaks or dips attributable to driver resonances or network anomalies. On the bottom end, flat response extends down to 22 Hz; on top, the roll-off begins at 17 kHz. (These are essentially 0 dB readings, not –3 dB points.) With the upper-midrange control in its neutral position, the "Gundree dip" we took mild exception to in the original version is just about gone; Bob sort of snuck it in so gradually that it is virtually undetectable, unless you know about his perversity on the subject and actually look for it. Well, at least there is no excess energy in that touchy "presence" region. The square-pulse response of the Platinum II is gorgeous, perhaps even better than that of the Quad ESL-63, but of course the ribbon can be expected to behave coherently above the single 100 Hz crossover. We are a little disappointed that the woofer array and the ribbon are still being driven with opposite polarities; however, when we reversed one pair of leads from the network to "correct" that, the wave launch in the crossover region became totally discombobulated, indicating that the physics of the drivers and the filters strongly resisted the "purer" approach. The roller coaster impedance curve of the Amazing is considerably improved in the Platinum II Edition; now the minimum is 4 ohms (at around 300 Hz) and the maximum 15 ohms (at 2.5 kHz and 7 kHz), with 8 ohms a reasonable average or nominal value. The standard sensitivity measurement being normalized to an assumed 8-ohm load, both sensitivity and efficiency measure approximately the same in this case: 88 dB. We are unaware of any other ribbon loudspeaker system that comes even close to that figure. At the same time, the Platinum II can easily handle the full power of an amplifier like the Carver Silver Seven-t. That kind of efficiency and power handling, combined with the low distortion inherent in the ribbon design, may be conducive to unnecessarily loud playback levels with CD's or DAT's, since there is never any sense of strain or impending breakup. Those who set the volume by cranking the music to the "ouch!" level and then backing off a hair will have to change their technique.

Since the "beta version" of the Platinum II made available to us was already broken in, we have no idea whether or not the 50-hour break-in requirement that we faulted in the original speaker has been reduced; according to Bob it has been, quite a bit. All we can say is that this particular pair of Platinum II's sounded sweet, smooth, open and uncolored from the start. To rate the listening qualities of the speaker against a widely known standard, we decided to stage a *mano a mano* confrontation with the Quad ESL-63 USA Monitor, which we consider to be the timbral-accuracy champion of the world and very bad news for most of the competition. The 63's can be moved around a bit more easily than the Waveforms, so this was not an unrealistic project. To our surprise, the Platinum II was not "blown away" (to use the favorite cliché of the 'philes), either in the reproduction of timbre or of spatial detail. For example, with the new Dorian sampler (Vol. II), the piano and strings of the Ames Quartet sounded perhaps a shade better on the Quads, but the lovely soprano of Julianne Baird came through even more palpably and ambiently on the Carvers. So let us say that the two speakers are more or less in the same league in terms of accuracy and musicality, but of course there is no contest when it comes to bass and dynamic range—or performance per dollar.

Where does that leave The Amazing Loudspeaker, Platinum II Edition—hey, we said it without wincing—in our hierarchy of speaker designs? Very close to the top, and several steps closer than the original. This particular application of the line source, with these particular refinements, seems to be giving the point source a run for its money (*pace* Stanley Lipshitz). We shall wait for our promised production samples of the "Plat Two" (Bob's shorthand) before we officially bless it as a reference; at this point we only wish to ask a rhetorical question: What other monolithic loudspeaker system goes down as low, goes up as high, can play as loudly, as cleanly, as transparently, render spatial detail as accurately—and at what price?

JBL L40t3

JBL Incorporated, a Harman International Company, 240 Crossways Park West, Woodbury, Long Island, NY 11797. L40t3 bookshelf 2-way loudspeaker system, $798.00 the pair. Tested samples on loan from manufacturer.

First, the good news. The JBL proprietary pure-titanium 1" dome tweeter used in this speaker system is absolutely the finest electrodynamic tweeter known to us, regardless of price. That includes anything and everything by Audax, Dynaudio, Focal, Vifa—you name it. On axis, it is flat within approximately ±2 dB up to 25 kHz; it is still quite flat 30° off axis (and more!) up 16 kHz; it handles power like a woofer; it sounds clean, clean, clean—and only JBL has it. (If others could have it, nobody would use those dinky little supertweeters to extend the top end of their domes.)

Now for the not-so-good news. The 2-way system JBL designed around this dream tweeter and a fairly decent 8" paper-cone woofer is okay but far from excellent. All by itself, without a reference, the L40t3 sounds like a respectable high-end bookshelf speaker, but when we switched to the Snell Type C/II in the middle of a Julianne Baird track on Dorian, everybody listening went "ah!"—the midrange became so much more open and uncolored. We tried to nail the reason for that on the lab bench and found that the woofer, although reasonably flat in response, is set 2 to 3 dB higher in level than the tweeter, or so it appeared when we had the B&K microphone close enough to the speaker to avoid serious room effects. That step-like transition in the crossover region could have been what we heard—or possibly just excess midrange energy. The two drivers are in phase, though; square-pulse response is pretty good, indicating some degree of cohenrence, but not spectacular.

The bass response, considered separately, is again quite good for a smallish 8" system; the vented box is tuned to 36 Hz, and maximum vent output is also in the neighborhood of that frequency, i.e., not far from classic fourth-order Butterworth alignment. (The vent is in the back, so it is difficult to obtain the summed response of the woofer and the vent in the nearfield.) Large-signal bass could be a little better; high-level transients tend to create a high-Q response profile as the voice coil starts leaving the gap.

JBL had the means and the opportunity here to come up with a sensational little speaker, but the whole ended up being less than the sum of its parts.

Precise "Monitor 10"

Precise Acoustic Laboratories, a division of Onkyo USA Corporation, Suite B, 200 Willliams Drive, Ramsey, NJ 07446. "Monitor 10" floor-standing 3-way loudspeaker system, $1599.00 the pair. Tested samples on loan from manufacturer.

The last time we looked at a loudspeaker system from Onkyo was ten years ago, and we were not impressed. This time they did the right thing. They created a separate loudspeaker division under a new name, Precise, and hired an American audio engineer with considerable professional as well audiophile cachet, Keith O. Johnson, to design a whole new line of five speaker models. The message is clearly that they are serious about speakers and serious about the audiophile market. As readers of this journal know, we have a lot of respect for "Professor" Johnson without necessarily agreeing with all of his priorities; in this particular situation we suspect there may have been some pressures to finish five commercially viable designs against a deadline, but even so we find much to like in the top-of-the-line Monitor 10. The others—Monitor 3, Monitor 5, Monitor 7, Monitor 9—we have not seen or heard.

The basic format of the Monitor 10 is that of a forward-firing 3-way system with 10" woofer, $6^1/_2$" midrange driver and 1" dome tweeter. The woofer is in a vented box of $2^1/_2$ cubic feet internal volume; the other two drivers are in a separate detachable enclosure. The tweeter is offset rearward from the midrange, and the midrange from the woofer, in the *soi-disant* time-aligned configuration we have always considered to be of questionable value. (Not that we object to it, mind you; it certainly does no harm.) The tweeter level is adjustable by means of a three-position toggle switch. A hood goes over the midrange/tweeter module to create a somewhat KEF-ish appearance.

The performance of the Precise Monitor 10 can be analyzed in terms of a little speaker—the midrange/tweeter module—extended on the bottom by a modest subwoofer. That perspective is suggested by the low crossover frequency, which looked like 100 Hz to us, although the literature says 200 Hz. The crossover network, incidentally, is complex and appears to use slopes of several different orders; we are reluctant to categorize it on the basis of our routine microphone probing. The tweeter takes over at 4 kHz, at least as we see it; again, the literature says 2.5 kHz. At any rate, the little-speaker part of the system is a good one, very flat between 100 Hz and 10 kHz except for a bit of excess energy at 1 to 1.2 kHz; the soft-dome tweeter has a distinct notch at approximately 10 kHz, above which the response is again quite flat to beyond 20 kHz. Off-axis response is also impressively flat. The midrange and tweeter are in phase with each other but out of phase with the woofer; the latter tracks the polarity of the input, whereas the midrange and tweeter reverse it. Square-pulse reproduction is fairly coherent, nonetheless, since the discontinuity takes place at a sufficiently low frequency. The "naked" midrange driver, when measured directly through its separately available terminals, looks a little rough in response, rougher than it is within the system—a somewhat surprising discovery in view of Keith Johnson's known preoccupation with cone materials, resonances, and stress modes. This polymer-laminated paper-cone unit has a 4 dB peak at 1.2 kHz, which constitutes a "gotcha!" when correlated with the aforementioned little problem around that frequency in the overall system response, where it is less obvious. (Yes, we believe we can hear it; we are coming to that.)

The woofer—or quasi-subwoofer—under this very respectable little speaker is another matter. We could hardly believe our instruments when we found its overall output level to be set approximately 10 dB below that of the midrange. What on earth is going on here? The vented box is tuned to 22 Hz by means of an incredibly long and convoluted duct; maximum vent output is also in the neighborhood of that frequency; so this is an attempt to design a genuine 22-Hz box around a 10" woofer enclosed in $2^1/_2$ cubic feet of space—come on! How much efficiency, how much output can you get that way? Of course, as we said, Keith Johnson's priorities are not always the same as ours; it is possible that he prefers *some* output at 22 Hz to a flat, efficient 36-Hz or 38-Hz box which is level-matched to the midrange. Or was it the marketing people?

The sound resulting from this mildly eccentric design is surprisingly good, perhaps because that little-speaker core of the system is well-conceived. If we found ourselves in a situation where we urgently needed a high-quality speaker system and the Precise Monitor 10 became quickly available to us, we would be more than pleased. If, on the other hand, we were offered a number of possible options within roughly the same price range, the Monitor 10 would probably not be our first choice. It represents just another flavor in good, up-to-date loudspeakers. (See our introductory comments above.) What is that flavor? Well, a little more emphatic in the upper midrange than the more neutral Snell TypeC/II, for example—most probably as a result of the small peak we measured—but partisans of the speaker would undoubtedly call that a lively or snappy or open quality, and they would not be totally wrong. The deep bass is occasionally evident but rather tame, not at all like that of a flat-down-to-22-Hz system, and that is exactly what the measurements predict. The overall sound is musical and very listenable but not so compelling that you want to run out and buy the speaker. We somehow have the feeling that the performance/price ratio may be more impressive in some of the other Precise Monitor models, where the working parts are of comparable quality but the design perhaps more conservative.

Overriding our minor reservations about the speaker is our genuine admiration for Onkyo as the first major Japanese audio company to abandon the endemic Akihabara glitz-tech approach to loudspeakers and to come out with a high-quality, businesslike, Anglo-American type of product instead. That line of thinking has a future.

Quad ESL-63 USA Monitor

Tovil Distributors of America Inc., 14120-K Sullyfield Circle, Chantilly, VA 22021. Quad Electrostatic Loudspeaker 63, USA Monitor, $3990.00 the pair. Tested samples on loan from USA distributor.

This is not really a full-fledged review, as there is no need for one. For quite a number of years now, the ESL-63 has been the closest thing to a gold standard in the world of loudspeakers, and the late great Richard C. Heyser's review of the original version in the June 1985 issue of *Audio* leaves very little room for emendations by lesser lights, at least electroacoustically. (Subjectively his evaluation was rather strange, but several high-end/underground/dilettante reviewers have meanwhile counteracted that.)

Our reason for putting in our two cents worth at this time is that the ESL-63 keeps coming up as a kind of undefined reference in the context of our speaker reviews, and that it has been sitting there against the back wall of our studio to be pulled out into the room whenever we are in doubt about the validity of what we are hearing—all that without ever having appeared in our table of contents, until now. In this issue, for example, the other six speaker reviews might not have had quite the same thrust had it not been for that reassuring presence in the background.

The approximately two-year old USA Monitor is a ruggedized, electrically and physically less fragile version of the original ESL-63, so minimally different in fundamental design and execution that the same old manual comes with it, although the input panel configuration in the back is no longer the same. (No voltage selector, 100 to 120 volts being standard; conventional input terminal posts, but spaced just a hair too close together to accept Pomona double banana plugs. This is the USA, chaps; 0.75" here is 19.05 millimeters, not 18.75, because 1" = 25.4 mm, not 25 on the nose. Or did you go by the Queen's thumb?) We can neither confirm nor deny audiophile claims to the effect that the USA Monitor sounds "better" than its predecessor, not having had them side by side and not being in the habit of making such critical comparisons from memory.

The frequency response of the USA Monitor as we measured it corresponds quite closely to Dick Heyser's published measurements of the earlier Quad ESL-63, even though our methods were not the same. The speaker is very flat indeed, and the small departures from flatness are innocuous; on the other hand, the Carver "Amazing Loudspeaker" Platinum II Edition is even a little flatter, so we are no longer as impressed as we used to be. The same statement goes for square-pulse reproduction; nothing we have seen equals the USA Monitor except the 60" ribbon of the Platinum II, which exhibits perhaps even a little less "grass" on the scope between pulses. Dick Heyser did not use that particular test, but we verified and agree with some of his reservations about the speaker: the horizontal dispersion could be a lot better; high SPL's are out of question; floor placement is a no-no. (We are still looking for the ideal stand.) We also have some reservations of our own about the bass response, which is neither deep enough nor tight enough. There is a rather Q-ey bump at around 50 Hz and good-bye Charlie below 40 Hz—and quite limited power handling at the lower frequemcies.

All that fades into insignificance, however, once you become familiar with the way the Quad reproduces instrumental timbres. A cello sounds so much more like a cello than through any forward-firing electrodynamic speaker at any price—witness the Delos CD of Janos Starker playing David Popper encores. String quartets, quintets, trios, etc., sound self-evidently natural, unstrained, true to life on the Quads where other good speakers merely give you hi-fi. It must be the combination of flat response, coherence, low distortion, virtual point source, figure-eight launch, who knows what else—and all bets are off if the dynamic limits of the system are exceeded. A very special, very frustrating speaker—unbelievably beautiful at times, at other times totally inadequate—but very hard to live without when you know what it can do.

Crossing the Quad ESL-63 over to a subwoofer, maybe at 80 Hz or thereabouts, will probably get rid of some of the frustrations, although dynamic headroom limitations at higher frequencies will continue to intrude from time to time. Unfortunately, we have not yet found the suitable subwoofer and crossover for the job; there are some tricky "hybridization" problems to be solved there with an electrodynamic monopole bass transducer.

Meanwhile the Quad ESL-63 USA Monitor, used full range, remains our reference for music with limited bass and dynamics—which includes a great deal and excludes even more.

Snell Type C/II

Snell Acoustics, Inc., 143 Essex Street, Haverhill, MA 01830. Type C/II floor-standing 3-way loudspeaker system, $1990.00 the pair. Tested samples on loan from manufacturer.

This one has been weighing heavily on our conscience. We already had our samples of the speaker when Issue No. 12 went to the printer, yet we had no review of it even in No. 13, except for a one-paragraph mention. Now this model is about to be superseded, although its successor will be only minimally different, we are told. Every now and then a worthwhile piece of equipment falls between the cracks of our slowpoke testing/publishing process, and this was one of them. Most regrettable—and not to be repeated.

Luckily, the Snell Type C/II is so simply and unproblematically good in its own specific category (viz. single-box forward-firing passive-crossover speaker systems) that a highly analytical review is hardly needed—just a general thumbs-up and a few specifics. That such is the case must be credited to designer Kevin Voecks, who had the astuteness to enlist the technical resources of Dr. Floyd Toole's famous acoustical facility at Canada's National Research Council in the development of his design, instead of flying by the seat of his pants like so many egomaniacs in the loudspeaker industry. There's math in them there Snells.

This is an approximately 4-foot tall system with a relatively small footprint of about 1 square foot; even so, it is big enough to make one expect better bass out of its 10" woofer and rearward-firing ducted vent than it delivers. The box is tuned to 31 Hz, but the overall bass response begins to decline at 48 Hz, with the –3 dB point at around 45 Hz. We could be slightly off on that measurement, since the summed response of the woofer and vent proved to be quite difficult to measure in the nearfield, but not by much. By contrast, when your Editor was in the speaker business, his comparably sized 10" system (Fourier 1 or Fourier 1L) pumped out an honest 32 Hz at –3 dB. That is our only serious criticism of the Snell Type C/II, and even that must be qualified by the observation that the Q of the system is quite stable under dynamic conditions—it will not "woof up" on bass transients.

Otherwise the speaker is well-nigh faultless. The woofer, the 5" midrange cone driver, and the 1" soft-dome tweeter are all in phase; the obviously complex crossover (4th-order Linkwitz-Riley, we believe) works beautifully; above 200 Hz the frequency response is uncannily flat, not only on axis but 30° off axis as well—as good as we have ever seen. Of course, with Floyd looking over Kevin's shoulder, he would probably not have dared to leave the premises before everything was perfect. We found it impossible to get a really good square pulse out of the Type C/II despite all that perfection, but that has been our experience with other good 3-way systems using high-order crossovers. It should also be added that the late Peter Snell's trademark of a tiny supertweeter aimed at the back wall is still with us here; you can switch it off if that extra little ambience it adds to the sound is not to your liking. We had it on most of the time.

The basic listening impression made by the Snell Type C/II is one of absolute neutrality. There is never too much or too little of anything; the balance is always right, on all types of program material and with all amplifiers that we tried. Of course, the round-and-mellow partisans will call it a little bright, and the sizzle aficionados will call it a little dull—and that will prove our point. In terms of our complaint about too many flavors in loudspeakers, even the good ones, this one comes reasonably close to the flavor of pure water. One does not tire of it, even after hours of listening. The bass is not particularly impressive, as we said, but what there is of it is of high quality. Indeed, the entire speaker is of high quality.

When a classical record producer friend asked us to recommend an accurate monitor speaker that is not too large and heavy to be transported in his car to recording sessions, we suggested that he try the C/II. He did and he is happy. What more can we say?

Waveform

Waveform Research Inc., R.R. #4, Brighton, Ontario, Canada K0K 1H0. Waveform floor-standing 4-way loudspeaker system, $3995.00 the pair, with dedicated electronic crossover (system purchasable directly from manufacturer). Tested samples on loan from manufacturer.

This review had to undergo a substantial last-minute rewrite because of the announcement of a radical price cut, which completely changes the positioning of the product from the critic's point of view. Let us begin, therefore, with a little bit of history before we address the hardware.

The driving force behind the Waveform loudspeaker is John Ötvös, a Canadian master woodworker of Anglo-Hungarian extraction. (His surname means goldsmith or silversmith in Hungarian, and it is in some ways prophetic.)

John had a dream, which he started to implement four or five years ago: to build the world's finest loudspeaker and sell it to grateful audiophiles via discerning high-end dealers. Yeah, just like that. As a not very far from stereotypical "mad Hungarian"—fiercely idealistic, proud, contemptuous of mediocrity, music loving, impulsive, and impractical—he paid a lot more attention to the product than to the marketplace as he pursued his dream. Obviously, the speaker had to be a masterpiece in wood—he would attend to that—but it also had to be the ultimate in engineering design. Unfortunately, no amount of wishing could produce a totally new and different design concept, but at least the speaker was to incorporate all existing and available knowledge of conventional technology. To that end, he enlisted the help of two outstanding Canadian technologists, Paul Barton and Dr. Claude Fortier, and he made full use of the acoustical laboratories of the National Research Council in Ottawa, under the supervision of Dr. Floyd Toole. It would be hard to find a better team than that.

When the original Waveform loudspeaker made its debut in mid-1986, the cabinet was the audio-furniture equivalent of the Bernini baldachino in St. Peter's in Rome, a dazzling display of woodworking virtuosity. Designed in the form of a truncated octagonal pyramid (which does make some acoustical sense), the solid cherry cabinet had two-inch walls, delicately fluted side panels, Ceylon ebony and 24-karat gold inlays (*ötvösmü*), raw-silk grille, and so forth. In a sense the tail was wagging the dog—the cabinet was upstaging the high-tech innards and fancy electronic crossover—and the three-piece set was priced at $17,000, which of course included the discerning high-end dealer's down payment on his Porsche. For those who would not be ashamed to settle for heavy fiberboard construction with a black, very high-gloss, Steinway-like lacquer finish and no flutes or inlays, the price of the otherwise identical system was only $9800. What John did not fully understand—and what we could have told him had we known him at the time—was (1) that discerning high-end dealers would be most unlikely to take on an ultrahigh-end dream speaker that did not have the endorsement of the dream merchants of the high-end/underground journals, (2) that said dream merchants had previous and very strong political commitments to the likes of Infinity IRS and IRS Beta, Wilson WAMM, etc., and (3) that therefore the advent of "the first technically 'correct' forward-firing speaker" (John's claim in a letter to us and at least arguably valid or near-valid) was actually *bad news* in the corridors of high-end hi-fi power. At first John was quietly ignored in those corridors, then he was politely raped. His speaker was bad-mouthed in both *The Absolute Sound* and *Stereophile*. We were not the least bit surprised; in fact, we had predicted it.

At that point—and that was only a few weeks before you read this—John revised his dream. "To hell with the magazines and the dealers," he obviously said to himself, "I am going to sell the Waveform directly to the audiophile who wants this kind of loudspeaker." In most cases that would be a very bad marketing decision, in our opinion, but in this specific situation we feel that John Ötvös has nothing to lose and the audiophile has everything to gain. To clear his pipeline of existing inventory and to create a cash flow, he is selling the three-piece system in black lacquer at $3995.00, which is just about his cost. That, dear reader, is an almost indecent bargain, an absolute steal, and a safe one at that, since Ötvös Industries, which is John's custom woodworking business, is obviously profitable enough to allow him to subsidize Waveform Research, and he is therefore unlikely to disappear overnight. What guarantees he can provide to back up his direct-to-the-customer sales remains to be seen. The price cut certainly relieves the reviewer of the obligation to rank the Waveform against various world's-best contenders, but let us try to evaluate it without regard for its former or present price.

The foundation of the Waveform's "architecture" is a separately amplified and actively equalized 15" woofer in a vented enclosure. (There are actually two ducted vents, at the 5 and 7 o'clock positions next to the woofer.) We can unhesitatingly state that we have never measured or listened to a better bass system; it is the chief glory of the Waveform, especially as it is so beautifully integrated with the rest of the speaker within a monolithic structure only four feet high (although pretty fat at floor level, to be sure). The design is claimed to effect a 40% reduction in enclosure volume in comparison with an unequalized woofer of identical performance. The vented box is detuned from the optimum Thiele/Small parameters, and resonant peaking in the electronic crossover unit boosts the response back to where it should optimally be—at the expense of power amplifier headroom, of course, which is a lot cheaper than gigantic cabinets. The woofer's nearfield response—as we measured it at a summing junction of the driver and vent outputs, with the "low-frequency coupling" control on the crossover panel centered—had a perfect fourth-order Butterworth profile, i.e., maximally flat, with the –3 dB knee at 24 Hz, maybe 23 Hz with a little squinting. Add to that the long-throw JBL woofer's large-signal capability and low distortion, and you are in low-frequency heaven, exulting in Craig Dory's 32' organ pipes and Jack Renner's bass drums. Hallelujah! (It was one of Mitch Cotter's witticisms that the "sub-" prefix in "subwoofer" had the same meaning as in "subnormal" or "subhuman." The integrated just-plain woofer of the Waveform seems to bear that out.)

The electronic crossover brings in the two vertically deployed 6" midrange drivers at 130 Hz, with a third-order (18 dB per octave) slope and a Butterworthy corner; the woofer roll-off looks a little more subtle, showing some gentle attenuation well below that frequency but ending up with 18 dB per octave also. Of course, the low-frequency coupling peak introduced to help the box affects the overall profile to some degree. The 1" textile-dome tweeter, which is placed between the two midrange drivers, is crossed over at 2 kHz with fourth-order Linkwitz-Riley passive filters, as is the ribbon-type supertweeter on top of the array at 9 kHz.

All drivers are off-the-shelf but strictly top-of-the-line; the supertweeter is particularly costly. (As we said, this is a highly optimized *conventional* speaker.) The high-pass section of the electronic crossover introduces a boost of a little over 3 dB at 16 to 17 kHz to equalize the supertweeter where it begins to roll off. The combined response characteristic of the midrange units, tweeter, and boosted supertweeter is very flat, both on and off the median axis (strikingly so off axis); the large peak at 16 kHz reported by *Stereophile* in their review was nowhere in evidence in our tests. (The most probable explanation of this discrepancy is that the very light ribbon depends on the air load for damping, and that load is much smaller in the thin air up there at 7000 feet in Santa Fe than at altitudes where less light-headed and scientifically more accountable reviewers dwell. All those Sherpa audiophiles, on the other hand, should heed Larry Archibald's and John Atkinson's findings.) The only possible anomaly we uncovered as we explored the semi-nearfield output of the speaker with our B&K microphone (the farfield being too contaminated with room effects) was that the 1" dome tweeter was set maybe, just maybe, 2 or 3 dB too high in level with respect to the midrange. This little "step" in the response profile depended on the position of the measuring microphone; everything remained very flat above and below the crossover. We are told that the very latest modification of the Waveform, which we did not have, introduces a small amount of additional attenuation in the tweeter branch of the network, so our suspicions may have been right—even without the anechoic chamber of the NRC. In any event, we are talking about a rather trivial level balancing decision, involving ±1 ohm of padding—not an engineering error. (Some reviewers are unaware of the difference.)

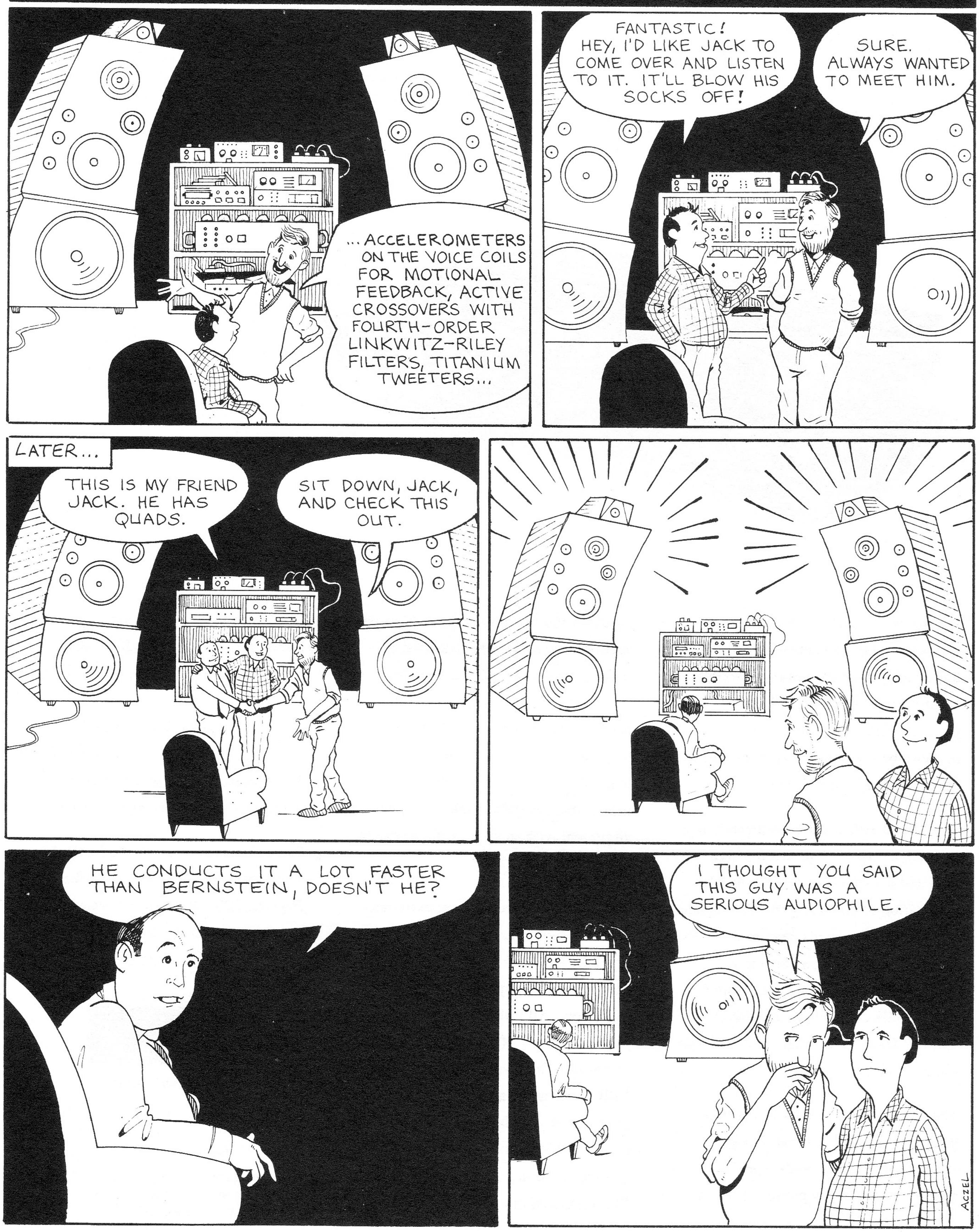
The impedance curve of the unequalized woofer is exactly what one would expect in a vented system; that of the midrange/tweeter/supertweeter section, on the other hand, meanders somewhat intricately, both in magnitude and in phase, but not so much as to constitute a difficult load for a decent amplifier. The nominal impedance of 8 ohms appears to be a judicious average of these swings; the sensitivity (2.83 volts input) and efficiency (1 watt input) are therefore the same: 90 dB SPL at 1 meter. That is pretty phenomenal for a flat wideband system. All drivers appear to be in phase, but we had a great deal of difficulty trying to sum the output of all five drivers, or at least of the upper four drivers, to reconstruct a square pulse. Not surprising and not disturbing—this speaker is coherent to all intents and purposes. Distortion is extremely low; at levels that did not drive us out of the room the THD readings were minuscule, almost amplifier-like. We should also add that John Ötvös supplied us with a set of NRC printouts documenting all sorts of measurements at their state-of-the-art facility, and none of these contradicted any of ours, except possibly the tweeter level matching.

What kind of sound results from all of this elaborate engineering? Incomparable bass, as we already stated; stupendous dynamics, without even a hint of stress on the loudest symphonic and operatic climaxes; almost totally neutral tonality, probably a little on the dry side for those accustomed to the sloppy, rounded-off "warmth" of various speakers; also, a mercilessly revealing, unforgiving character when reproducing juiced-up, overbright, edgily recorded program material, of which there is still no dearth, alas. We had no problem whatsoever with our favorite recordings, but it is possible that the slightly reduced tweeter level in the latest mod will make the speaker more tolerant of the average studio production. On the other hand, listener position is relatively uncritical with the Waveform; there may be a best seat in the house, but there are many very good seats. Imaging and ambient nuances are not exactly the speaker's long suit; it is quite satisfactory in those areas but not nearly as stunning as, for example, the little Sapphire from Audio Concepts. Emulating the latter by covering everything but the diaphragms with acoustical foam would probably do wonders, but try to persuade John Ötvös to give up that Bernini-in-wood shtick and put a black hood over his masterwork.

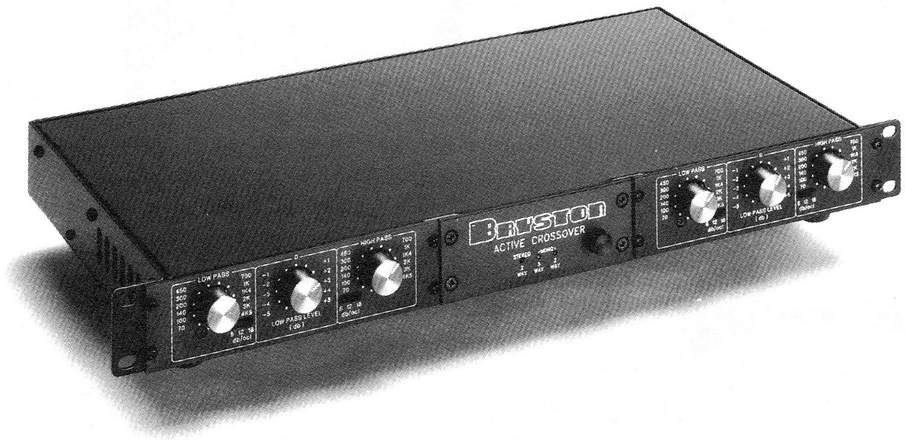
So, everything considered, this is indeed as good a forward-firing box speaker as there is (if you can call that fancy pyramid a box); it may have its minor weaknesses but it does all the big things, the difficult things, brilliantly. It sounds like real music, even when the music is complex, loud, and dynamic; it can fill the largest domestic spaces at realistic SPL's; and it tells the truth about the program source. How many other speaker systems, at any price, can make those claims? Of course, it still sounds like a forward-firing box speaker and not like a dipole or various other multidirectional designs. For that reason, it may not be everybody's dream speaker, maybe not even ours. But we feel that a forward-firing box speaker of this caliber *had* to be designed before one could decide whether or not that time-tested format is still the way to go. Remember, some rooms will not accommodate any other format. Dipoles, for example, have to be pulled far into the room to sound really good—let no one tell you differently—whereas the Waveform does not. Our judgment is that the Waveform fulfills the needs of a large number of audiophiles better than any other loudspeaker system, and especially so at $3995.

A few addenda and tidbits: The dedicated electronic crossover was driving two identical Boulder 500AE power amplifiers in our listening tests. The circuit design of the crossover, although basically excellent, is not quite up to the Boulder level of sophistication. Occasional and not too disturbing RFI was clearly traceable to the crossover unit. The speakers and the crossover we tested were the very same samples that *The Absolute Sound* had pooh-poohed without the courtesy of a full review. Nasty scratches in the black lacquer finish bore witness to careless, or possibly even contemptuous, treatment by slobs. In fact, the flat, mirror-smooth top of one of the cabinets had some interesting parallel scratch marks on it—as if someone had been cutting lines, man, with a razor blade. *O tempora, o mores!*

In Your Ear


...ACCELEROMETERS ON THE VOICE COILS FOR MOTIONAL FEEDBACK, ACTIVE CROSSOVERS WITH FOURTH-ORDER LINKWITZ-RILEY FILTERS, TITANIUM TWEETERS...
FANTASTIC! HEY, I'D LIKE JACK TO COME OVER AND LISTEN TO IT. IT'LL BLOW HIS SOCKS OFF!
SURE. ALWAYS WANTED TO MEET HIM.
LATER...
THIS IS MY FRIEND JACK. HE HAS QUADS.
SIT DOWN, JACK, AND CHECK THIS OUT.
HE CONDUCTS IT A LOT FASTER THAN BERNSTEIN, DOESN'T HE?
I THOUGHT YOU SAID THIS GUY WAS A SERIOUS AUDIOPHILE.
ACZEL

THE IDEAL CROSSOVER FOR MULTI-WAY SPEAKER INSTALLATIONS



Bryston 10B electronic crossover

Bryston's Model 10B Electronic Crossover combines ideal signal-handling with an enormously flexible control function. Simple, direct front-panel switches allow any crossover curve to be set instantly, and the signal purity is always maintained.

The Model 10B features independently selectable crossover points for high-pass and low-pass, in case the speaker installation requires slightly overlapped, (or slightly staggered), response curves for the drivers. You can also independently select crossover slope, from 6, 12, or 18dB/Oct., where one driver requires faster cutoff than another in the same system.

The crossover may be used in any of three internal connections: 2-way stereo, 3-way mono, and a special configuration, 2-way mono. This last cascades the low-pass and the high-pass sections and allows the selection of unusual crossover curves, including, "dual slopes", where the crossover point is effected at a shallow rolloff, and the stop-band is rolled off rapidly thereafter. It also permits the increasingly popular Linkwitz-Riley alignment with steep rolloff curves, 24 or 36 dB/Oct.

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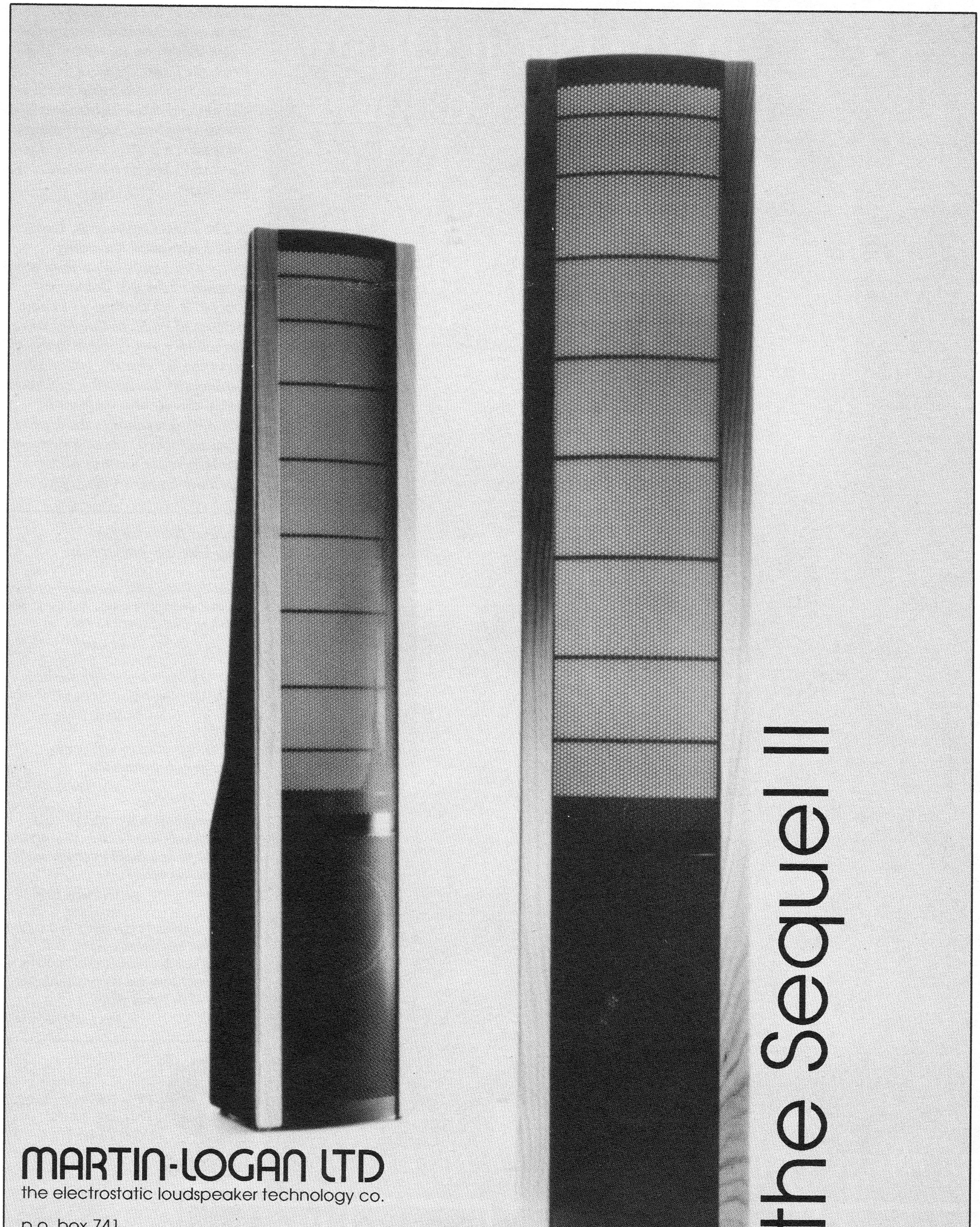
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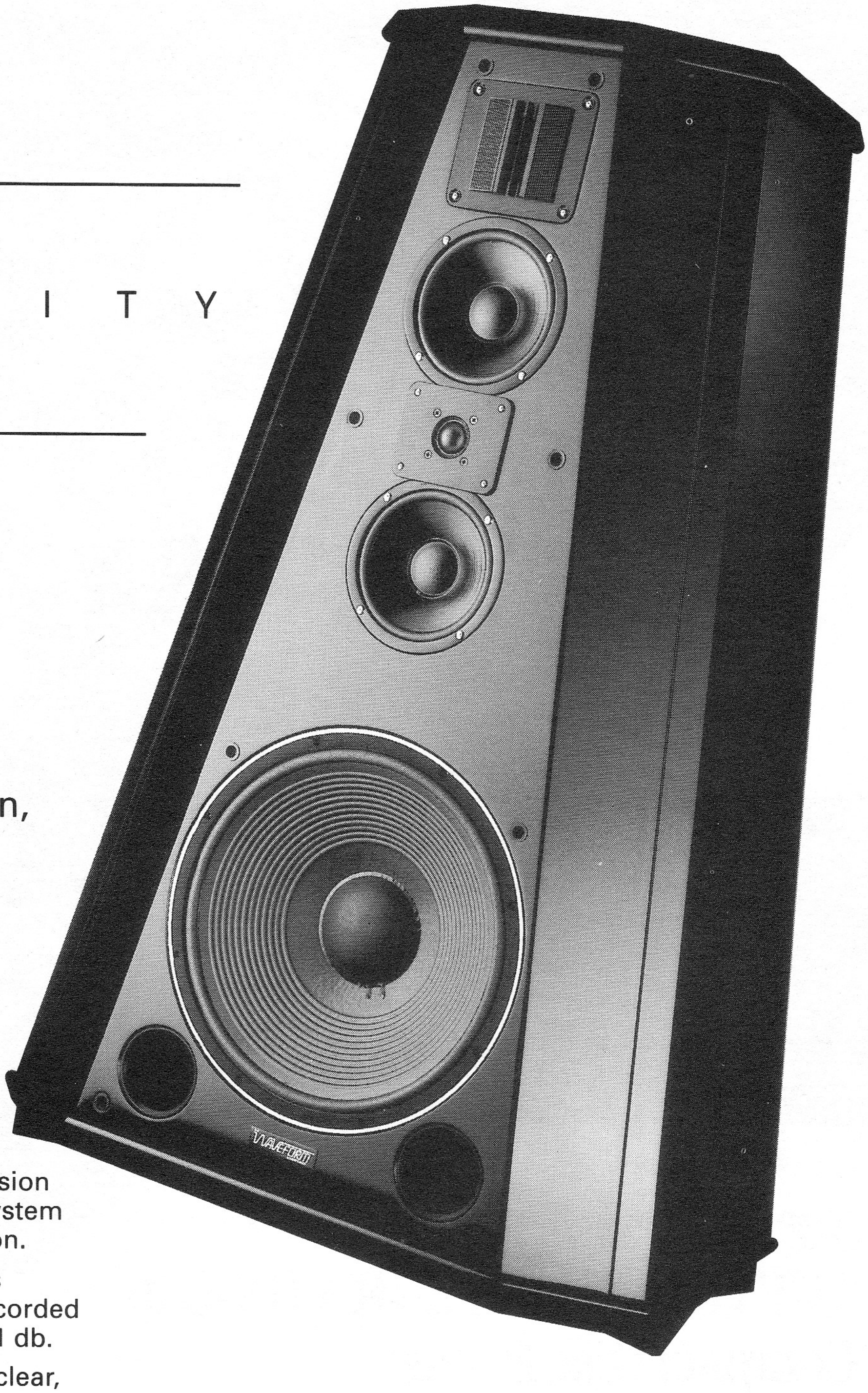

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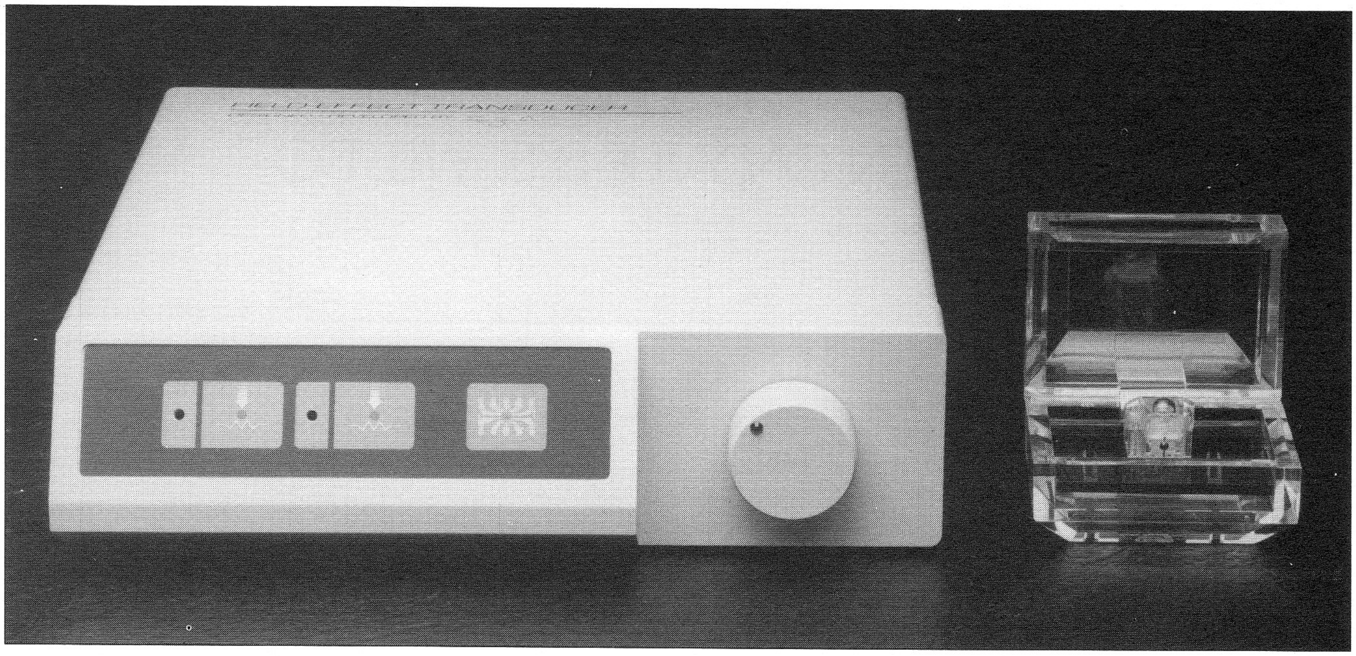

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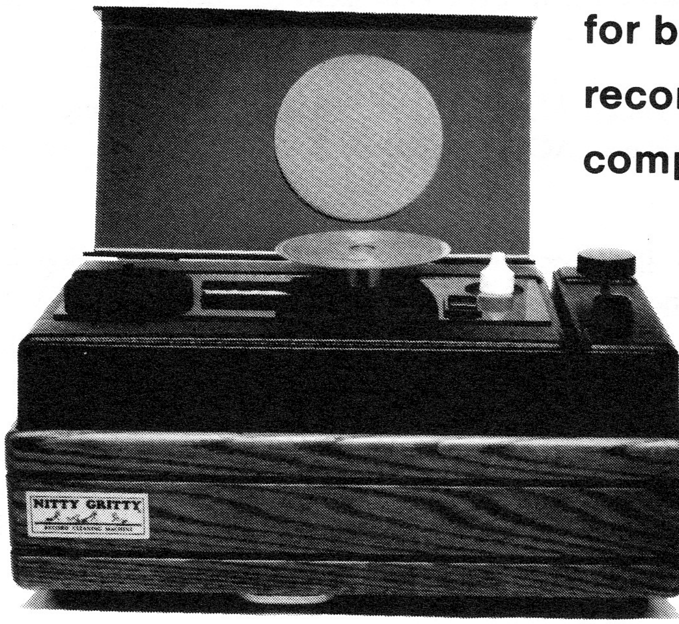

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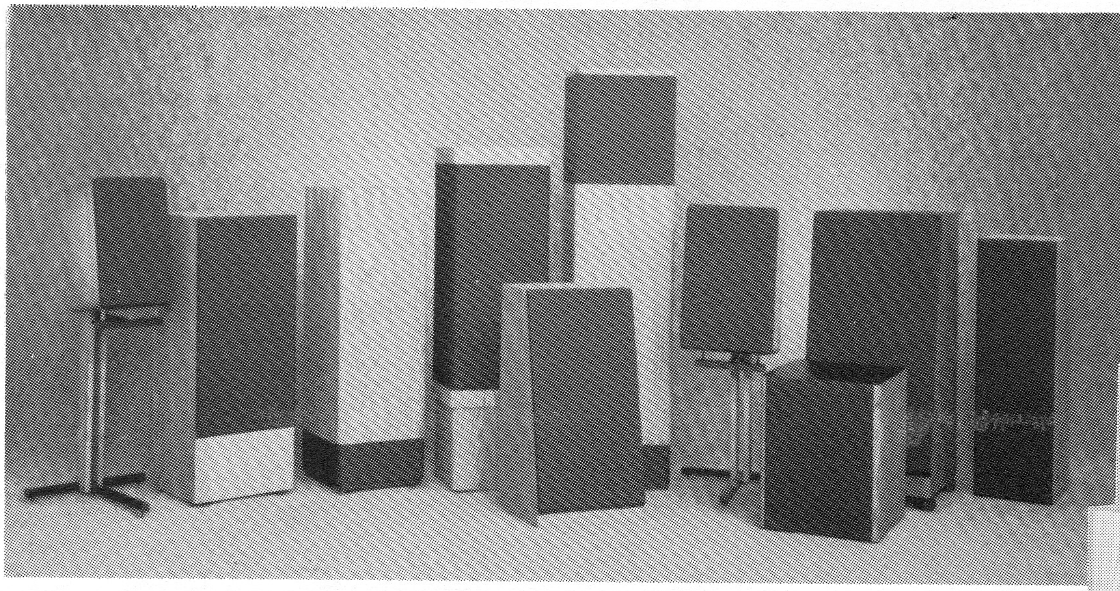

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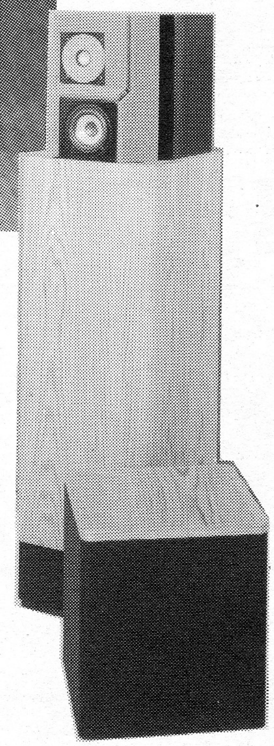

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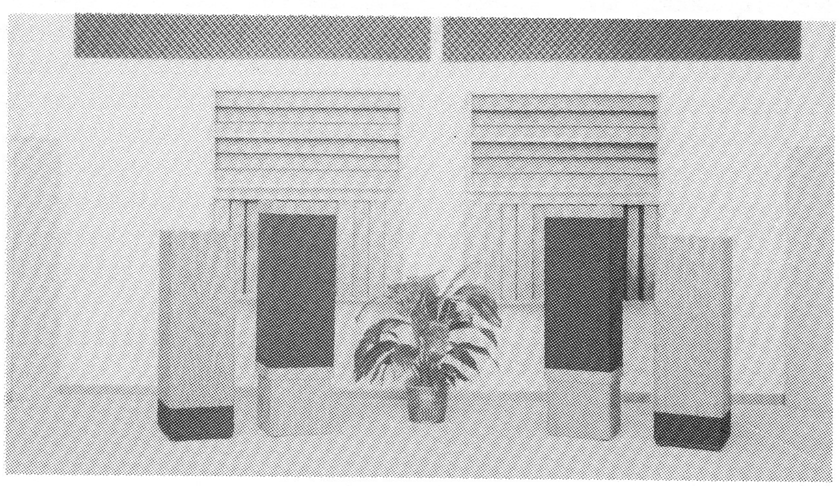

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Seminar 1989:
Exploring the Current Best Thinking on Audio
(Part II of the Three-Part Transcript)

The complete transcript takes up so many pages that we have decided to publish it in three parts, instead of two as originally announced. To get the most out of Part II, you need to have read Part I. Have you?

All the introductions, opening remarks, and prefatory throat clearings were taken care of in the last issue, at the beginning of Part I. We refer you to the capsule biographies of the six panelists published there; we certainly want you to be aware who is who when you read who said what. You will recall that the transcript ended where the participants adjourned for lunch; it resumes here where they returned to the conference table. The juicy gossip in the dining room was unfortunately not captured on tape, probably a blessing in disguise from the legal point of view...

EDITOR: Let's start discussing the program material in general and recording techniques in particular. I would like to ask each of you—three of you being very active recordists and all of you having strong opinions on the subject—I'd like to ask you severally and together what you think the right recording technique is to achieve the kind of results that we've been talking about, that we all want. Who would like to start?

EARGLE: Okay, the thumb is pointing in my direction. First off, I think we have to put this in some perspective by saying that, in this country at least, less than 5% of the recorded material sold is classical or based on a natural model, a natural acoustical model of some kind that we want to translate over, the rest of it being entirely multitrack and open to a new set of laws or lack of laws. It's purely whatever is created over the loudspeakers in the control room. The healthiest thing to happen in classical recording, which is really what we're talking about here, happened about ten years ago, and it was the fact that going to digital recording forced you more or less back to two tracks—because there were no digital multitrack machines at that time that were practical. The Soundstream machine could always lay down 8 tracks at once but with only 20 minutes playing time for the whole pass. So that encouraged a lot of people to throw away their multitrack machines for classical and go back to even earlier than where they had been in the early '60's, when I began in this business. They were using three and four tracks in those days. And I might say parenthetically that the worst thing to try to figure out a use for was a three-track recorder back in 1962. It was an invitation to put the soloist on the center track and the violins on the left and the cellos on the right, without any real sense. In other words, the only stereo was really whatever leakage there was between all three of these. So if you ever tried to play with the level of the soloist, you were really playing with the level of the middle of the orchestra at the same time, so it really wasn't a satisfactory solution.

LIPSHITZ: You put the Mercury recordings in that category?

EARGLE: The Mercury stereo recordings were basically the two outside tracks, with just enough of the center channel, the mono channel, to give you a little bit of center fill, and it was usually run about 6 dB down.

LIPSHITZ: All right, but it was three widely spaced microphones, and each went to one track.

EARGLE: They weren't that widely spaced. There was a group of microphones that might have been panned together; they were sent to the two outside tracks. The middle track was always used for the mono disc alone. I mean nine times out of ten; that was a single mike that somebody had figured a good place to put. But I really think that clearing the house, so to speak, was the best thing that ever happened in classical, and a lot of people are still recording two-track classical, even now with multitrack available. Not everybody, however; in fact, Deutsche Grammophon regularly uses a Sony 32-track for recording classical music. They have their own editor that will edit that many tracks at once, and after the *Aufnahmeleiter* has done the editing, it's then given to the *Tonmeister* to do a mixdown, and it's entirely in his hands from that point. But in this country, many of the smaller companies have gone to a basic two-track format and as such have sort of wanted to simplify the microphone array, maybe limiting it to no more than 8 to 10 microphones, maximum.

LIPSHITZ: But the fact that the recording might be done on two-channel machines doesn't mean it will be only a few mikes.

EARGLE: No, it means that you've got to make the right decisions.

LIPSHITZ: You can't change your decisions.

EARGLE: That's right, you can't change them later, and it will make you go for a safer perspective on the ensemble because you know you can't fudge it later.

LIPSHITZ: Now I believe Decca/London historically always have—except for their big opera recordings—mixed down live to two channels.

EARGLE: Well, the few Decca sessions that I've been to have been basically direct to two but with a multitrack backup. And the way they do it is really very, very sensible. The have a console, a homemade console that has several sections in it. Each section is really a two-channel board. One section will be devoted to the main array across the front; another section will be devoted to house microphones; all of the accent mikes will be routed to another section—and these will all be fed to their own left/right outputs, which individually go to pairs of tracks on the multitrack. But at the same time all the L's and R's are sound and feed directly to a two-track machine. Now, if they find that they made a mistake in the balancing—in the on-the-fly balance of the two-track, whatever that mistake might have been—they always have the option of pulling out the multitrack tape and setting it up again with individual calibration points and duplicating the two-track output, except they can now go in and ride gain on these various sections. Now, that's limiting because you can't go in and change one microphone at that point, but you can change the entire section, the entire subgroup—which seems to me to be the most sensible way if you really feel you need backup to cover yourself for a potential artist problem or balance

problem in the eyes of the conductor. That's a very good way to do it. But overall I would say that we're seeing a lot more sanity in the miking of all kinds of classical music, and we're seeing, more than anything else, a new generation of microphones. Remember that you never heard of B&K instrumentation mikes being used for this purpose until the first digital recorders came along—because they had flat power bandwidth whereas many analog machines of the day would actually have high-frequency saturation.
LIPSHITZ: But the B&K mikes would not always be used correctly.
EARGLE: Well, when you say correctly…
LIPSHITZ: Well, I am thinking of a specific example. Some of Marc Aubort's recordings, for example, where he used B&K 4134's, which is the pressure mike, not the free-field mike.
EARGLE: Okay, yeah. A lot of people didn't understand the difference between a flat-on-axis and a flat random-incidence microphone. You're right; many times the wrong mikes were used. They were also quite a bit noisier than most of the mikes, but they at least didn't have a high-frequency peak.
LIPSHITZ: Well, wait a minute. Wait, wait, wait.
McGRATH: Depends on the axis, how you point it.
LIPSHITZ: I asked him about that; I asked him how he had used his 4134's—were they pointing at the musicians or were they pointing up at the ceiling?
EARGLE: Well, granted, he was misusing them in terms of…
LIPSHITZ: And they were pointing at the musicians, which meant they were rising above 5 kHz.
EARGLE: But if they were properly used, they didn't have the peak that the current Neumanns and AKG's had at that time. That's the point I'm trying to make.
LIPSHITZ: Right. That's smooth, but not flat the way he used them.
EARGLE: But if properly used, they were smooth and flat.
LIPSHITZ: Yes.
EARGLE: And I think that today we see everybody using all redesigned microphones, things that are the product of the last ten years. Schoeps, the Sennheisers, the newer Neumanns, AKG's—and I think that that's made a drastic difference in the texture of our recordings, of everybody's recordings.
LIPSHITZ: For example, Decca/London, at least according to what I've read in the magazines, shortly after they started coming in for some criticism about a rather harsh, strident high end—I think this was actually before the CD was out but when some of the digitally recorded tapes were put on LP—bought a whole batch of Schoeps microphones to replace their omnis with the rising high end; they bought a batch of omnis with flat high ends.
EARGLE: Yes, they had been using the M-50's, which have the rising high end. That's a hybrid microphone that is omni at low frequencies and becomes directional at high frequencies; it's got a mike mounted on a little spear inside, about that big, if you've ever seen the internal structure of it. That was a product of around 1950 or so and was designed by Neumann for rather distant pickup in concert halls, for broadcast, where they couldn't maybe have the mikes right down on top. And Decca adapted that, and that was their mainstay, and with the old analog recorders it didn't sound so bad.
CARVER: What did you mean when you said, change "the texture of your recordings"?
EARGLE: Well, okay. There are two words that I use, *structure* and *texture* in the recording. Structure, to me, is being able to determine quite unambiguously where something on the soundstage is. I feel the two are of almost equal importance, and one without the other is a very unsatisfactory recording. Structure is what tells you, you know, where the violins are, where the winds are, and the brass…
CARVER: So that would be our, or my, "imaging," perhaps.
EARGLE: Well, imaging in that sense, in the sense of left/right, front/back. Texture, to me, implies an overall coloration or glow to the sound. In my own miking, for the most part, there's a near-coincident pair in the middle, an ORTF pair usually, which provides all the structure that you're going to need in a recording. Flanking that, and operating anywhere from 6 to 8 dB down, might be a pair of omnis.

"But overall I would say that we're seeing a lot more sanity in the miking of all kinds of classical music, and we're seeing…a new generation of microphones."

CARVER: Those give you the texture.
EARGLE: Well, they help in that regard, and they also give you a little bit more of the string sound up front, across the front of the orchestra.
CARVER: They supply the bloom.
EARGLE: That's right. You're using the word "bloom," you see, and I could ask what that means, except I think I probably already know what it means.
LIPSHITZ: An ambient field.
EARGLE: Yes. Then there's also a pair of mikes in the back of the hall, if I'm operating on the stage. If you're operating well out into a room that has a lot of reverberation, then you may not need those. And all these subsidiary microphones are really operating lower enough in level so as not to dominate. I'm sure that you operate in similar ways.
McGRATH: Well, simpler. It depends on what I'm recording. Probably two thirds or three quarters of what I do is orchestral recording for broadcast or for just concert recording, which are live musical events, where what I can do in terms of the positioning of the orchestra, or indeed the microphones, is very limited. So I have to go with the best that I can do, and having worked now for about 12 years in one particular hall that I enjoy working with, I've found what is a very hot "sweet spot" for my spaced omnis, which gives me the kind of perspective that I'm very comfortable with.
EDITOR: Which ones do you use?
McGRATH: I use the Schoeps. I've been using them for the last six years.
LIPSHITZ: These are the Colette Series capsules?
McGRATH: No. Well, they could be on Colettes, but they're CMC-5's with the MK-2S capsule.
LIPSHITZ: 2S?
McGRATH: It's called the 2S. They also have the MK-2, and then there's a new MK-2 which I'm not sure I like. I've got them all.
LIPSHITZ: What's the difference between the 2 and the 2S, Peter?
McGRATH: The 2S probably has a little bit more bite.
LIPSHITZ: It's up at the high end?
McGRATH: I don't know if it's brighter; it just sems to have a little bit more focus. Perhaps it *is* simply a rise; I don't know.
LIPSHITZ: Yes, I think it is.
McGRATH: Probably is; it's probably a bit peaky, but it sounds very, very good.
EDITOR: You used to be into B&K's, the instrumentation mikes.
McGRATH: I used to use the 4133, not the 4134. I didn't like the 4134. But the 4133 did sound good.
EDITOR: And what about the new ones that are for music, expressly?
McGRATH: I bought the 4003, and I didn't like that microphone. I found that it was appreciably quieter than the 4133 that I had used, and that in itself was a plus, but then…
EDITOR: That's why they came out with that series, right?
McGRATH: Yes, but then the Schoeps MK-2S was brought to my attention. I kind of glommed on to that mike, so to speak, and I have really not moved from it. I like it a lot.
LIPSHITZ: Peter, when you used the 4133's, you used them with the little grids on?
McGRATH: I used it with the regular grid on.
LIPSHITZ: Pointing at the musicians?
McGRATH: Yes. So that was rising on axis? By how much?
LIPSHITZ: No, no. The pressure mike, the 4134, would have been rising. The 4134 is flat, pressure, if you remove the grid…
McGRATH: I tried both capsules.
LIPSHITZ: …and if you were using the 4134, as Marc Aubort did, you should have pointed them at the ceiling, so that the sound came at grazing incidence. That gives them flat direct and flat random-incidence pickup; that would have, in fact, been better, in that sense, than you could get with the 4133, which would be flat, direct, but rolling off, random incidence.
EDITOR: That's right in the instruction manual, for that mike, if you use it for measuring.
EARGLE: But who ever reads the instruction sheet on a microphone?

EDITOR: John, you use a Japanese condenser mike, don't you?
EARGLE: Yes, the Sanken CU-41.
LIPSHITZ: That's a two-way microphone.
EARGLE: Yes, it's a woofer-tweeter microphone. And what I like about it, in particular, is the frontal response of a cardioid—this microphone is zero on axis and exactly –6 dB down 90° off, all the way out to 12.5 kHz. And the pattern is absolutely uniform over that entire frequency range in the front half; the back half is pretty choppy, as it is on all cardioid mikes. The reason I do that is that, when you have a near-coincident pair splayed out—and the splay angle may be well beyond 110°, which ORTF recommends—the fact is that the middle of the orchestra comes in with lack of coloration, of off-axis coloration, and I find that to be very, very important.
LIPSHITZ: I think the B&K cardioids are probably also very good in that respect; at least according to the curves they look very good.
EARGLE: Yes, they are.
McGRATH: I've not found cardioids that I like, for that very reason. I haven't tried the Sanken, nor have I tried the B&K, but the ragged quality of the middle is horrendous.
LIPSHITZ: I quite honestly don't like cardioids.
McGRATH: I don't, either.
LIPSHITZ: You know, I'm not involved in making commercial recordings, and essentially everything I record is live concerts.
EARGLE: You're more or less in a documentary mode.
LIPSHITZ: Well, it's the local chamber music society. We put on annually about 50 concerts, and they're all recorded for broadcast on the university radio station. We have a weekly radio program, and with 50 concerts a year we have the material to do that. In fact, we have been recording digitally for about six years now, I guess, and broadcasting live from the digital master tapes. I suspect we may have been the first North American station to broadcast live digital recordings over the radio. We must have 250 or 300 master tapes.
EDITOR: Stanley, I understand you believe in single-point miking.
LIPSHITZ: Well, I believe in single-point miking, and we can say more about those ideas later, but single-point cardioids—the ORTF is spaced, slightly spaced, quasi-coincident cardioids—I don't like single-point cardioids.
EARGLE: Well, you have way, way too much middle, unless you splay them way out.
LIPSHITZ: The reverb is all bunched up in the middle of the soundstage between the speakers. Edward Tatnall Canby in *Audio* magazine a few months ago was talking about this; he had a number of articles on the Denon Mahler cycle with Inbal, in Europe...
EDITOR: I reviewed one or two of those.
LIPSHITZ: ...and they used the B&K cardioids even before they were released. He made some comments about imaging and ambience and recording techniques, and so on—some things I disagreed with, and in fact they just recently published a letter of mine commenting on some of these things—but he made one interesting point that I hadn't really appreciated that clearly till he made it, and that is that coincident cardioids are inphase for the full 360° pickup. There's no out-of-phase rear lobe, and the result is, when you reproduce that over speakers, that all the reverberant images are between the speakers.
CARVER: That's right.
LIPSHITZ: And if you analyze it, the reverberation sort of bunches in the center, so you have a center-weighted reverb. It's closed-in sound.
CARVER: The reverb should have a lot of out-of-phase components.
LIPSHITZ: Yes.
CARVER: It drives me nuts when I hear recordings that are made, and the reverb is inphase.
EARGLE: I agree. You mean predominantly inphase?
CARVER: Predominantly inphase. And that bloom that I like a lot just evaporates.
LIPSHITZ: So you see—just to complete this little thing, then I'm sure you'll all want to comment—I don't like coincident cardioids. I far prefer coincident figure eights or coincident hypercardioids, both of which have a full-level or –6 dB out-of-phase rear lobe, both of which give a significant amount of incoherent reverb pickup—pickup of directions other than the direct sound. In fact, the figure eights, of course, because you've got a $\sin^2 + \cos^2 = 1$, they have equal pickup in all directions in the horizontal plane, so that they're totally neutral for direct and reverb pickup, and both side quadrants are out of phase.

"I believe in single-point miking...but I don't like single-point cardioids. The reverb is all bunched up in the middle of the soundstage between the speakers."

EARGLE: I can give you a problem in using a single pair of microphones for picking up both reverberation and direct sound. A lot of times... well, it depends on the hall you're working in, and it depends on your ability to reseat players. If you're on the floor of a large English church that's been deconsecrated and converted to a recording studio, then you can pretty much move microphones...
EDITOR: In that order? *(General hilarity.)*
EARGLE: Yeah, yeah. The players will be consecrated at the first break. If you know English players, they'll find the nearest bar. Anyhow, the thing is that, when you're at the beginning of a session, you've got to make a lot of very, very quick decisions, and having separate quantities in the recording available on separate faders is a godsend. Because if you wanted more reverberation, for example, or less reverberation, with a crossed pair of figure eights, you'd have to move in or out of the orchestra. So in changing one perspective, you're changing the other one, too. And that gets to be a problem because, sometimes, the right texture on strings and their balance with the rest of the orchestra depend upon being in a very narrow front-to-back zone of—and I'm not kidding you—around two feet.
LIPSHITZ: I think you're right. There are trade-offs. What I'm saying has nothing to do with the commercial realities of having to make a recording. My point was simply that—given that one was making a quasi-coincident-type, ORTF-type recording—the reason that ORTF is preferred to coincident cardioids, I believe, is the slight phasiness due to the spacing...
McGRATH: It gives that open quality.
LIPSHITZ: ...because you have no phasiness with coincident cardioids, since everything is inphase, the direct sound and the reverb. Now the direct sound should be coherent, should be inphase, but you want the rest to have the kind of relationship it has *live*, which is incoherent. And you do get that with figure eights; you get it with coincident mikes with out-of-phase rear lobes. I think a lot of people who have tried coincident or quasi-coincident recording have done one of two things: they've either used mikes with very poor polar patterns or not good enough polar patterns—because, as you pointed out, John, you've got to have very good off-axis performance if you want to be free of coloration in the pickup—or they have used cardioids and didn't like the sound and ruled out the technique. There are all these other reasons why one might not be able to manage with that sort of configuration, but what worries me about some of the techniques, for example Dave Griesinger's equalization of coincident mike recordings, where he boosts the L – R signal below, say, 500 Hz by 6 dB or more...
EARGLE: Or more—sometimes more.
LIPSHITZ: ...now boosting L – R below 500 is not the same as cutting L – R above 500. When you boost L – R relative to the L + R reference, what was a coherent recording becomes phasy in the low-frequency regime because, when the L – R is added to the L + R to get back left and right, the L – R can dominate the L + R, and you can figure out that with a coincident pickup what was inphase direct sound then becomes antiphase direct sound for sources more than a certain amount from the center. So he is adding phasiness on direct sound, and that is a great pity in order to produce some kind of ambient, airy, phasy sound quality altogether. Whereas if you cut L – R above 500 Hz—of course you just make things more mono—you don't add the antiphase. So I don't see that as a solution to what is perceived by some people as a problem. I guess that's enough for this moment.
EDITOR: What I hear so far is that there is no substantial disagreement among you about the kind of sound you would like to achieve; there is some minor disagreement as to the tools that you would use to achieve that. Do I misunderstand you; are

you after different kinds of sound?
McGRATH: Well, I think that's an assumption that may or may not stand up. If Stanley were to play a recording that he liked on a system that he approved, and then I would play one that I like on a similar system, we might then get into some very serious disagreements as to what we like and what we don't like.
LIPSHITZ: I think we would find we disagree because your playback system sounds—on account of the speakers—very similar to mine. We're talking Quad ESL-63's, which are speakers where, because of the polar pattern of the speaker, you've got less room effect on the playback, so that you're listening analytically to what's in the recording—and I think we would probably disagree.
McGRATH: We'd probably disagree on what we like. But what figure eight do you use? What coincident figure eight capsules?
LIPSHITZ: Well, we had no good microphones available initially. We had a pair of U-87's at the university radio station that we used and built a jig to hold them. The only polar pattern by the way for the U-87 that's at all good is the figure eight. The most difficult polar pattern to make by the way is a cardioid. Most mikes that are omnis or figure eights will be better than the corresponding cardioid. So we used those and experimented with them for a number of years, and then in 1978 Anton Kuerti, Toronto pianist, one of Canada's finest pianists, played a complete Beethoven sonata cycle for the chamber music society—eight concerts. And we broadcast him live from the Theater of the Arts straight to the FM transmitter and bought some custom equipment to do it. We applied for an Ontario government grant for some microphones, and we got it. So halfway through that cycle, after a great deal of soul-searching and investigating, we went for Schoeps—a couple of bodies and a set of capsules. The capsules we bought were the MK-41 hypercardioids, the MK-40 cardioids, and the MK-8 figure eights, which have only recently become available. The second half of those concerts were broadcast using the MK-8's; the Theater of the Arts is fairly dead. So the figure eights I'm talking about that are good are the Schoeps MK-8's. The Schoeps cardioids are also quite good; the Schoeps hypercardioids are very consistent; but more recently we use a Calrec Soundfield microphone as a stereo mike, and there the polar patterns are all extremely accurate because they're synthesized from the outputs of four capsules in the head, the array, of the microphone, which are individually equalized for frequency response, so you're talking figure eights that do not roll off below 100 Hz but are equalized flat to 30 Hz.
McGRATH: That's what I was going to ask. On the MK-8's you had to use some kind of EQ, didn't you?
LIPSHITZ: Not for chamber music, usually. No, we didn't.
EDITOR: Dave, you wanted to say something?
CLARK: I'll make a statement about using recordings. The only recordings I've been involved with are studio-type recordings. But in using recordings, the work that I do at this time is in automobile sound systems, which I guess aren't up to the level of what we're talking about here...
EDITOR: Conceivably they are.
CLARK: ...however, getting back to sitting off center in a living-room situation, you have the same problem—the biggest problem of listening in a car is that you're not in the center. I feel that a lot of these techniques are evaluated by sitting on the center line between the speakers, head in a vise so to speak, and you say I've got imaging from here to there, and I can pick things out—for me that's not good enough. In a way it's better than it needs to be; I can get better imaging—or structure, as John says—of the soundstage, sitting in the center between two speakers, than I need...
EARGLE: Sometimes it's too good, yes...
CLARK: ...but as soon as I move off center, it degrades to a level that is not up to my standards. Namely, everything piles into the near speaker. In a simplistic way I tend to blame that on coincident miking because it almost always has elements of the sound, of any sound on the soundstage, coming from both left and right speakers.
LIPSHITZ: Which does? Sorry?

"...I can get better imaging sitting in the center between two speakers than I need... but as soon as I move off center...everything piles into the near speaker."

CLARK: Coincident.
LIPSHITZ: I'm not quite sure I understand.
CLARK: Without a time delay.
LIPSHITZ: You're saying that those speakers carry all the instruments.
CLARK: Right.
EARGLE: But not in the same amount, except along the median plane.
CLARK: Not in the same amount.
LIPSHITZ: Of course not. If you're using figure eights, any speaker on the right-channel axis would be in the null of the left microphone—so that would *not* be true for hard right and hard left, which would be the direct sound, single channel only.
CLARK: That's right. You *can* find them, just as with a pan pot in a studio you can pan all the way from one side to the other.
LIPSHITZ: And such a source would stay in the right loudspeaker no matter where you listened from because there would be no left-loudspeaker direct sound for such a source on a coincident recording.
EARGLE: Dave, I think what you are addressing here is a tendency of anything that is absolutely common mode between the two channels—in other words, equal information between channels—to really flop over to the nearest loudspeaker when you're off axis.
CLARK: That's exactly what I've been saying.
EARGLE: And for that very reason, I find that in a recording made with three spaced omnis—left, center and right—I find that this hard center, which is absolutely in-phase between channels, just collapses over. And it's one reason why I never pan anything in the middle—even if I have a soloist. I always have a stereo pair on the soloist. And those mikes are fed hard left and hard right, and it's just the fact that the soloist is off axis on both that creates a center image that's not pinpoint—it has a little bit of area to it.
McGRATH: But it feels real. I found that technique works very well.
EARGLE: It feels very real. Also—a very, very critical thing for me—if I'm seated right on axis, you can postulate that that's the best place in the world to sit, but if you're a little bit off, that common-mode information is going to get into some pretty broad comb filtering.
CLARK: Yes, it does.
EARGLE: Until you get maybe a couple of feet off axis. One way you can diffuse that problem is to avoid it by having quasi-coincident microphones which don't pick up that tight a center-pan-potted middle.
CARVER: So that the vocalist has a little stereo spread.
LIPSHITZ: A little phasiness.
CARVER: A little L – R.
LIPSHITZ: Let me put a contrary view on this, to what Dave was saying. With coincident recording methods, the direct sound is essentially inphase in the two channels.
EARGLE: It's like pan pot.
LIPSHITZ: Yes. And the reverberant sound done properly should be in random phase—that's why I was making the case against using coincident cardioids. That being the case, as you move away from the center the image moves over and fairly rapidly moves into the nearer loudspeaker. That's true. So central images will move over; any hard left image would stay in the left-hand loudspeaker, and any hard right would stay in the right-hand loudspeaker when you move. So what happens is that the image stretches. The left-hand end stays where it was; almost all the rest moves over to the nearer loudspeaker. Now let's look at an example of, say, spaced omnis, the sort of thing that Peter uses.
EDITOR: And John does.
LIPSHITZ: John adds them to a quasi-coincident central thing at a slightly reduced level.
EDITOR: Oh, I see.
EARGLE: An organ recording might have a pair of spaced omnis because of the special ambience that you're trying to pick up, but that's not an image problem.
LIPSHITZ: Well, it's not clear what the image of an organ is because different organs have different images depending on where they put the pipes.
EARGLE: That's right.
LIPSHITZ: But anyhow, the point was this—that with the spaced omni recording there isn't any clear image. You know, I can go into more detail about that, but my great objection to spaced omnis is that the

theory of imaging doesn't exist. For classical-type recordings, the microphones being relatively distant from the musicians, the main difference for sources near the center—musicians near the center, woodwinds and so on—is time difference, time-of-arrival difference at the microphones. With coincident recordings, of course, there are no time-of-arrival differences; it's all level differences in the two channels, the two mikes. Time-of-arrival differences at the mikes produce level differences at the listener's ears. Level differences at the mikes produce time-of-arrival differences at the listener's ears, the reason being that each ear hears both loudspeakers. This is a very misunderstood thing, and I can't prove it for you here, but...
EARGLE: Well, it can be proved.
LIPSHITZ: It can be shown that over the region from 1 kHz down, or at least 500 Hz down, which is half the audio bandwidth in terms of octaves, that is the case. Now the dominant hearing mechanism is time-of-arrival differences at the ears, and coincident recording produces recordings which match that—in other words, which are natural in that sense. Spaced-mike recording produces level and polarity differences at the ears. And what you actually find is that—take a slightly off-center source, spaced mikes, and let that instrument play up the scale—and what you find is, as the instrument plays up the scale, you have...
EARGLE: Rotating vectors...
LIPSHITZ: ...yes, at your one ear the signal level will change, go through a null, reverse polarity, change, go through a null, come up again. And the rate at which it changes with frequency is different at the two ears. So what you're getting is inphase and antiphase signals at your two ears. In other words, total phasiness. You never, live, get signals that remain inphase over a band of frequencies and then antiphase over a band of frequencies.
EARGLE: On the other hand, if you take a coincident recording and sit off axis...
LIPSHITZ: You add some of that *to* the other things. Yes. So my point is this. You see, time-of-arrival differences at the ears produce phase shifts, but a phase that varies linearly with frequency, so they are delays. You don't get a phase that stays 180° over this whole band of frequencies and suddenly flips to zero and then back to 180°—that kind of thing. That is what I think a lot of people are crediting with ambient, airy warmth. It's not ambience. It's phasiness on direct sound. That would be fine on ambience, but it's there on the direct sound. Now, just to complete my point—because you're all going to want to comment at length on what I just said—when you move away from the center of a coincident-mike recording, you add, as John has just indicated, some of this phasiness problem, which you don't get at the center. The image moves over, collapses into the one speaker, and phasiness is introduced. When you start with a spaced-mike recording, you have that phasiness already at the center listening position; you don't have any kind of precise imaging; things near the center are really quite anomalous and move around with frequency. And when you move away from the center, you don't get the same impression of collapse because you didn't have anything really to collapse in the first place. But things are still happening the same way. It's just that the aural impression is of less of a degradation because you've got less to degrade; it's already degraded in the middle. There's real things to attack now.
McGRATH: Well, the response I would give to that is simply this: I'm not disputing the validity of anything that you've said. The only thing that I can offer is that I own the Schoeps MK-8's and I've used them in maybe 30 different halls, and I've used the MK-2S's in a spaced configuration and I've used them in a variety of different musical situations ranging from solo piano to Mahler symphonies. And in many cases I've not only done that but I've also hung a pair of spaced omnis and then critically positioned a pair of MK-8's and have run parallel recordings—and without injecting my personal bias on it, I've played both back, and in every situation that I've been able to conjure up, I've gotten a more generally preferred result with the spaced omnis. That's it, simply stated. I don't dispute you. And if I could get an MK-8 or a coincident mike or indeed a figure eight that gave me the tonal characteristics that I achieve on a consistent basis with my omnis, I probably wouldn't hesitate to use it for all the reasons that you cited.

"That is what I think a lot of people are crediting with ambient, airy warmth. It's not ambience. It's phasines on direct sound."

LIPSHITZ: Can you tell me what the criteria are? You see, I think we probably wouldn't agree on what we mean by stereo.
McGRATH: Well, I think that the criteria are, you know, very simplistically speaking, that one sounds more like what most people's perception of beauty is. One sounds pretty thin, one sounds pretty hard, one sounds pretty unlistenable, sounds pretty constricted; the other one sounds maybe very vague and phasy, but it has palpable beauty in it, and people generally tend to like it.
LIPSHITZ: Now the MK-8's do roll off below 100, 150 Hz gradually, whereas the omnis are much flatter—but that's not what you're talking about.
McGRATH: Also, the string tone sounds always very harsh to me; it's always very gritty and aggressive.
LIPSHITZ: Are these good acoustics that you're talking about? Because, of course, the omni has a worse direct reverb characteristic than the figure eights.
McGRATH: Well, we're dealing with the reality of going in and making a recording in a hall, and in different hall situations I keep coming up with this consistent result. That's all I can argue. I mean, the results speak for themselves.
EDITOR: Maybe we could bring this down to the level of most audiophiles if each of you could state what you like among commercially available recordings and what you don't like, and use those as an illustration of what you're talking about. Because this is highly theoretical; in fact, Dave just showed me your article here, Stanley, which is in the September 1986 *Journal of the Audio Engineering Society*, where you provide mathematical proof of what you just discussed.
LIPSHITZ: Well, I mean, it's not original, please! There are at least seven or eight references there. You'll find some of the proofs elsewhere. It's just—I was getting frustrated that people would apparently misunderstand that thesis.
EDITOR: But could you provide some commercially available examples of these various characteristics that we're talking about?
McGRATH: You mean in terms of recordings that substantiate one point of view versus the other?
EDITOR: Yes. Right.
CLARK: I have a hard time finding a recording I think was made with two spaced omnis.
McGRATH: I've got 27 of them that I brought with me.
EARGLE: Listen. I can tell all of you gentlemen here—and anybody who disagrees will have to substantiate the disagreement—on a typically good stereo recording I'm hard put to tell exactly how it was made. On a bad stereo recording I can cite chapter and verse, probably, on how it was made.
McGRATH: That's a very profound point. I would agree with that a hundred percent.
EDITOR: That's a striking thought.
EARGLE: So the thing is that there are many, many nice recordings about which I'm shocked when I find out how they were made. There's a recording of Arleen Auger that's a Grammy nominee for best female vocal, classical.
EDITOR: This is your recording?
EARGLE: Yes. And you will all agree when you hear it that this is a very natural, very nice concert setting of a singer. Horsefeathers. It was not. It was done in a chapel, where I used the natural ambience of the room, but because of the music-making conditions I could not mike her standing in front of the piano. First off, you all realize that when you have a vocal recital, the piano is usually on half-stick, and the singer nests herself in the crook of the piano. And it's because of half-stick that you get the proper balance from instrument and singer going out into the hall. Now, when you're trying to document this as a recording, a piano doesn't sound good on half-stick.
McGRATH: Sounds hard.
EARGLE: Sounds terrible.
LIPSHITZ: Loses its brilliance.
EARGLE: Sounds terrible. Okay, so if you raise the piano cover all the way up...

McGRATH: She's lost.
EARGLE: ...she's lost. So you have to move her out a little bit; they lose communication...
McGRATH: Then they can't play together. Always a problem.
EARGLE: ...and then they can't play together. Okay. So the way we solve the problem is very simple.
LIPSHITZ: Stand to the right of the keyboard.
EARGLE: No, not even that. You want her in the middle. You want her in the middle because that is an expectation for many listeners. It's an expectation for broadcast. So what you do, you mike the piano as an event, and you put the singer about eight feet away facing the piano and put a mike in front of her, in this case a cardioid, about four, five feet away, and balance to that way.
LIPSHITZ: If you'd used figure eights, you could have put them one in the front and one in the back, except one batch would have been out of phase, out of polarity.
EARGLE: That's right. I'm not sure how audible...
McGRATH: How do you correct that?
EARGLE: You really can't.
CLARK: She could sing into a reflector...
LIPSHITZ: She could breathe *in* while she sings.
McGRATH: She could sing backwards! *(General merriment.)*
EARGLE: The point is—and I also had a pair of mikes in the house to pick up a little bit more room sound and mix in—that I defy you to spot any of these ingredients when you hear the recording. Anyhow, the best recordings—of course, as in the case of anybody who's in it professionally and has product, you can all go out and listen to my records if you want to—I would say that among the work of others some of the more recent English Decca things have been masterpieces of multimiking. A recording that I've liked rather recently is one that Jimmy Locke did with the San Francisco Orchestra. It sounds to me like they've gone away from the M-50's and the KM-84's that they've been using for years.
EDITOR: On what label is that, John?
EARGLE: English Decca. London in this country.
McGRATH: Some of the Dutoit recordings are rather exquisite, and I don't know what they do, but they're thoroughly enjoyable.
EDITOR: Would you make that statement about the Dutoit recordings from day one or just the very recent ones?
McGRATH: I haven't listened to them all.
LIPSHITZ: They've all been done in the same church near Montreal. I haven't visited it but I did read somewhere that the ambient sound you hear is not entirely the church; it's been assisted electronically. I believe that's the case; I may be wrong.
EARGLE: A little bit of that is all right. If you have a natural base to work from, you can tail out just a litle bit more with the Lexicon and have it sound very, very natural. You wouldn't want to do that in a dead room.
LIPSHITZ: No. I believe they take foam things and put them in some of the seats. I think they actually erect a new platform in the body of the church, where the musicians sit. You know, it's not what most people probably visualize when they think of a recording being made in a church.
EARGLE: Well, Decca always—and they do it in San Francisco, they did it in L.A.—where the orchestra is seated on a stage, they will build a platform out from the stage, extending it maybe 20 feet out into the house. And one thing that you get when you do that, you find that the entire orchestra excites the same acoustical space. When the orchestra is back inside a shell, the back of the orchestra is in one environment and the strings are in another, and that's a problem.
LIPSHITZ: What's best for listening is not necessarily best for recording.
EARGLE: Absolutely. Because, in concert-hall listening, you may have 2500 people listening to one watt peak power coming out of the orchestra, and you need all the assistance you can get from a very sympathetic and reflective shell. And that plays havoc when you're trying to record. It gives you very unnatural, dry sounds.
CARVER: Dry sounds?
EARGLE: Yes. Well, dry in the sense of having a very high direct reverberant ratio.

"[As for] the best recordings ...I would say that among the work of others some of the more recent English Decca [London] things have been masterpieces of multimiking."

EDITOR: John, among your own recordings, which are you the proudest of?
EARGLE: Well, I would say that the *Petrouchka* recording that came out recently from the Seattle Symphony, with the "Scherzo Fantastique" and "Fireworks" and the "Russian Easter Overture"—it's a real bargain, you know, approximately 70 minutes' worth of music—that one would certainly be an example.
LIPSHITZ: That's on Delos?
EARGLE: Yes. There's a new recording that's coming out on the Pro Arte label that I did; it was a recording of the Strauss *Le bourgeois gentilhomme*, done in Studio A at RCA with the New York Chamber Orchestra. I haven't heard the recording yet in its edited form, but I suspect it's going to be good. An orchestra of that size fits beautifully in a room that size, without artificial reverberation, just using the room itself.
LIPSHITZ: But you're not suggesting that the sound would be different in the mixed two-channel digital master you prepared and in the edited version of the CD surely?
EARGLE: No. Absolutely, the sound will be the same; it's just that I haven't had the chance to hear it at home.
LIPSHITZ: Okay.
McGRATH: There are people who would suggest that's a possibility, though.
LIPSHITZ: Yes, but I was hoping that John presumably believes that they're not going to modify his work.
EARGLE: That's true.

* * *

LIPSHITZ: Peter, you probably have examples of recordings you want to bring up.
McGRATH: I would have to divide them into two camps: recordings that I've made that I'm most satisfied with in terms of the music and, indeed, in terms of the sound. I think the Leonard Shure recordings that I've done of the Schubert sonatas are the most satisfying musically. I've never witnessed anything more profoundly moving to me in my life musically than the time I spent with him, doing those things. The sound is very, very good but it's certainly not the most up-to-date.
EDITOR: Those are strictly analog, right?
McGRATH: Those are LP's now. We're in the process of bringing out CD's of those.
LIPSHITZ: And that is on...?
McGRATH: Audiofon. And then, probably, sonically—and also musically it's not a bad performance at all—our recent *Water Musick* on Harmonia Mundi with the Philharmonia Baroque Orchestra.
EARGLE: And that's the San-Francisco-based group?
McGRATH: That's correct. Yes. And that's a lot of fun, and I'm very, very happy with the sound of that.
LIPSHITZ: I don't have any commercial recordings out, so there's nothing people can get hold of.
EDITOR: Well, you must be aware of some things that you like.
LIPSHITZ: I'll make some comments. If people want to experiment, there are certain labels which have a house recording policy—or had. And one can therefore be fairly certain—if one picks out recordings of a certain vintage—one can be pretty sure how they were made. For example, a lot of the early Delos recordings that were recorded by Marc Aubort have two widely space omnis—that's it. If you, for example, take almost any recording—I believe *any* recording—on the Nimbus label, they are ambisonically two-channel matrixed coincident-mike recordings. So they *are* coincident. There are a few other labels that have recordings of this specific kind. I think the Opus 3's are usually coincident recordings; I'm not positive about that. There are some others. But if somebody wants to try one thing versus another, I would suggest for example the following comparison. I've used it in a few lectures I've given. The Equale Brass Quintet CD on Nimbus, a coincident-mike recording with the Soundfield ambisonic mike, in a stereo reduction. It is ambisonically encoded but doesn't have the slight phasiness that is introduced by the two-channel matrixing of the three-channel original. But you are immediately aware when you listen to it of the depth—goodness, the trumpet is really much closer than the trombone, or whatever, in the recording. Take a totally contrary recording, which is the American Brass Quintet on Delos.

EDITOR: That's an early Delos, isn't it?
LIPSHITZ: Oh yes. It's a much more reverberant acoustic as the recording was made, and you're immediately aware of the fact that it's very reverberant and ambient. But think about the question of how you can tell on the Nimbus recording that the one instrument is much further away from you than the other one. It is also very ambient, but it's not blatantly ambient. The Delos is a widely spaced omni recording with a great deal of phasiness and no real precision in imaging.
McGRATH: What you're saying is, one is a blurred image in an ambient space and the other is a sharp image in an ambient space.
LIPSHITZ: I'm saying, one is a blurred image in an ambient space, with the blurring giving the impression of heightened ambience, and the other is a precise image in an ambient space, which doesn't sound ambient because it doesn't have the blurring. And I would suspect that if we sat down and passed judgment on these, we would disagree as to which was the more desirable.
McGRATH: I'm not sure, but I probably would.
EARGLE: On the other hand, we would all probably agree that we're hearing the same thing. But the judgment that we attach to what we hear would be different, depending on where we all come from and what our expectations might be—and what we just happen to like.
LIPSHITZ: That Nimbus is quite spectacular, by the way, from the point of view of dynamic peaks. They never released it on LP.
EARGLE: On the other hand, there are some Nimbus piano recordings, again using ambisonics, that are so far removed from the piano that they're hard to listen to.
McGRATH: I would agree with that.
LIPSHITZ: I'll agree. Some of them sound too reverberant.
McGRATH: Empty classroom sound, as they say.
LIPSHITZ: They may be fine—I don't know—they may be fine on surround-sound playback.
EARGLE: It could be. But the ears will help you lock in your...
LIPSHITZ: But I think they're not really right for stereo.
EDITOR: You haven't heard these decoded and played back that way?
LIPSHITZ: I have. I have decoded some of them ambisonically, but I have not when the system was set up properly; I have not assessed these.
McGRATH: You know, it's interesting. Without the prejudice of the discussion that just preceded this, I would wager that if you took my most recent recording, which is of the Mozart horn concertos with the baroque horn, and you took the Nimbus recording of the same music also with the baroque horn, and if you were to present both of these recordings, let's say, to another "august" group having the same discussion, and they listened to these two recordings, they would draw the opposite conclusions as to what microphones were used. I mean, I would just wager that would be the case.
LIPSHITZ: Yeah. I don't know.
McGRATH: Theirs is very vague and very, very ambient, and it's swimmy, and you have a hard time hearing the horn, particularly when he goes up and down the scales, and it's very disjointed from the orchestra and swimmy, and mine I think, if anything, it almost—well, I wouldn't say it suffers—but it's analytically right there. Absolutely crystal clear in focus. They used the Soundfield; I used a pair of spaced omnis.
LIPSHITZ: Of course, the tools are only tools in the hands of the user. It doesn't guarantee anything.
EARGLE: And the room is so important.
McGRATH: It's everything.
CARVER: And as John says, he challenges you upon listening to the finished product to determine which system was used.
LIPSHITZ: And you can't tell easily. Sometimes you just can't tell.
CARVER: I have a question. The kind of recordings I like paint a picture that extends slightly beyond the speakers, with nice pinpoint imaging—I can close my eyes, and it's there, it's there, it's there, it's there...
CLARK: It's too wide.

"True coincident [miking] would mean there's no time-of-arrival difference in those two channels, and that I know I don't like. When I hear that, it sounds flat."

CARVER: Huh? No, no, no. I put my speakers close together.
LIPSHITZ: What angle do your speakers subtend?
CARVER: I like the soundfield to subtend about *this* angle. *(Bob must think he is on TV.—Ed.)*
LIPSHITZ: Well, that's about 60 degrees. *(Ah! Stanley to the reader's rescue.—Ed.)*
CARVER: My speakers are about *this* angle.
EARGLE: Is this with your sonic thing in there—the hologram working and that business—or without it?
CARVER: I'm just describing the soundfield I like. Never mind how I get there.
EARGLE: Okay.
CARVER: Because I'd like to know what microphone...
McGRATH: ...best enhances that image?
CARVER: That's half of it. And then the other half is the ambient soundfield, the bloom, all of the out-of-phase stuff that I respond to so, you know, in my heart; I just love that, I love it, I love it. But it's *everywhere* and it goes "shhht" and it goes higher than the speakers, it goes deeper than the speakers, lower than the speakers, wider than the speakers—and I *love* it when it happens. And I search, and I search, and I search, and so many...
EDITOR: You like ketchup on your sound.
CARVER: No, that is not ketchup on my salad. That's just wonderful, it's wonderful...
EDITOR: Sound, I said, not salad.
McGRATH: No, that's a wonderful dressing on a good lettuce. That's what it is.
LIPSHITZ: The only way you get direct-sound images that extend beyond the speakers is antiphase signals...
CARVER: No, I know how... I understand that.
LIPSHITZ: ...so you would not get that with a coincident mike arrangement unless you go beyond the inphase pickup angle or you process it to increase the L – R signal. You can get that with other mike arrangements.
CARVER: Too often, when I bring home a new CD and I put it on my system, I hear a flat curtain of sound strung between the two speakers. And it's distressing when I know that I've heard recordings that have this palpable depth that you talked about, where you knew that—was it the trombone?—was behind...
EARGLE: Or the horns, especially. And the horns really come out of back there because they're playing in that direction.
CARVER: Now, I understand exactly what the sound vector field has to be in my ears so that I hear that. But what's not clear to me is how you get it with microphone technique. I mean, I could map the soundfield that's required to generate that impression in my mind, but if I were to make a recording, I wouldn't have a clue. How do you get the imaging, and then superimpose on that the bloom—and make it work?
LIPSHITZ: You can't get the imaging without having a coherent pickup in the two mikes—assuming you're using two mikes. So that essentially forces coincidence, or near coincidence, if you want the imaging. If you want the bloom...
CARVER: Now near coincidence means—this is really the important part to me...
LIPSHITZ: Spaced no more than your ears are apart.
CARVER: True coincident would mean there's no time-of-arrival difference in those two channels, and that I know I don't like. When I hear that, it sounds flat.
EARGLE: I don't like it, either.
CARVER: Okay. So near coincident—we move them apart a little bit, at least a good fraction of an interaural time delay—is that right?
McGRATH: Like a little bit pregnant, right?
CARVER: Huh? Is that right? We move them apart a little bit.
LIPSHITZ: Yes. It would have to remain less than eight inches apart.
CARVER: Ah! Okay. Now we have something that approximates an interaural time delay, and that probably will sound very good, since...
LIPSHITZ: No.
CARVER: Why not? He says yes; you say no.
LIPSHITZ: Because you're assuming that moving the microphones apart introduces an inter*aural* delay. It introduces an inter-*loudspeaker* delay.

CARVER: It introduces the distance... no, no, no.
LIPSHITZ: It introduces a delay between the signals at the two reproducing loudspeakers, but that doesn't mean it introduces a delay betwen the signals at the listener's two ears.
CARVER: No. That's right.
LIPSHITZ: And that is precisely the point I was trying to make before. It introduces *phasiness* at the listener's two ears.
CARVER: Well, maybe phasiness isn't so bad. Maybe we'll interpret it as something...
LIPSHITZ: I didn't say it's bad. I said phasiness on the *direct* sound is bad. I said phasiness on the ambience is nice. You see, you're caught in the following dilemma—and this *is* a dilemma. The only way to prevent phasiness on the direct sound to get imaging is not to do this...
CARVER: I know.
LIPSHITZ: ...the only way—not the only way, one way—of introducing an ambient airiness is to add phasiness to the direct sound and lose the imaging. What you really want is no phasiness on the direct sound and plenty of ambience.
CARVER: That's right.
LIPSHITZ: The only way to get that in a live recording is to use coincident mikes which pick up a lot of ambience and *have* ambience to pick up.
CARVER: Okay.
LIPSHITZ: But it must be nice ambience to pick up, which means you've got to record in a good venue that's suitable.
EARGLE: There are a lot of those, right?
McGRATH: Oh, they're common.
LIPSHITZ: There are two. *(Laughter.)*
CARVER: To get this straight: so, to do it, we've taken a coincident mike and made it near coincident, so they're spaced about eight inches or so apart...
EARGLE: Well, I have, but he hasn't.
CARVER: You have. Okay.
EARGLE: Because there's one aspect of having it absolutely coincident that I don't like, and that's when I wander a little bit off the median plane, there's that sudden shift.
LIPSHITZ: I'm never very aware of it. I know what you mean. Yes, I know the effect.
EARGLE: I mean, it *has* to be there.
CARVER: It also makes the center image too hard; it nails it too hard in the center.
LIPSHITZ: Not much, though.
CARVER: And real life isn't like that.
LIPSHITZ: But, Bob, careful! You move your mikes six inches, eight inches apart; that's the distance we're talking, no more than that.
CARVER: Yes. That's a lot.
LIPSHITZ: It adds phasiness only in the frequency range above 1 kHz or so. About 500 Hz.
EARGLE: And the ear doesn't really hear phasiness per se up there.
CARVER: No.
LIPSHITZ: You're only sensitive to phasiness below 1 kHz, really. So you're adding it in the region where it doesn't degrade the low-frequency imaging; the sound is still essentially coincident below 500 Hz, so you retain the low-frequency imaging; you add some phasiness at high frequencies, which broadens and diffuses the image a little bit...
CARVER: I understand.
LIPSHITZ: ...and, well, I'm not quite sure what it does, but it makes a slight difference...
CARVER: It changes the time relationships a little bit.
LIPSHITZ: But it's nothing at all like what you get when you move your microphones three feet or six feet or nine feet apart.
CARVER: Oh, sure. Yeah.
LIPSHITZ: That is phasiness over the whole audible spectrum range, you see.
CARVER: Now, with that, when we've taken the coincident microphones and moved them apart, have we built the complete soundstage picture, or are we only halfway there?
EARGLE: It depends on the room you're working in.
CARVER: Do we have to add the flanking microphones? Do the flanking microphones make the...
EARGLE: The flanking microphones come under the heading of local seasoning at the hands of the master chef.
CARVER: So, you use them for two purposes: to enhance the pinpoint imaging that I was talking about...

"...the left-center-right spaced-omni approach has a lovely texture to it. I give it a B minus on imaging —not as analytical as what I like— and an A plus on texture."

McGRATH: No.
EARGLE: No.
McGRATH: No. On the contrary.
CARVER: So they're used strictly to make the bloom?
McGRATH: To get the color of the hall.
EARGLE: To pick up a little bit more hall sound and to give a little bit of an assist to a string section that may otherwise be swamped out by high-pressure brass and percussion.
CARVER: Okay. All right. So the...
CLARK: What's the timing? You bring them back a little farther?
McGRATH: You put them on the same plane.
EARGLE: I usually set them up on the same plane.
McGRATH: I do the same. Yes.
EARGLE: And they're always a good 6 dB down, often 8 dB down.
CARVER: Uh-huh. I understand.
LIPSITZ: I still disagree with your basic statement.
CARVER: I asked a question; I didn't make a statement.
LIPSHITZ: No, no, no. You did.
CARVER: I stated what I liked.
LIPSHITZ: You said that you maintain the imaging, and now this adds bloom. I say, as soon as you've moved your microphones a few feet apart, you have lost the imaging.
CARVER: No, no, no. Either you misunderstood me or I misspoke. I said what I *like* in my soundstage, and then I described it. And the question was, how do I get it?
LIPSHITZ: You called those images, though. By image, do you mean something whose location isn't a function of its pitch?
CARVER: That's right. Whose location is a function of its physical position in space.
LIPSHITZ: Physical position. All right. I don't believe you can get that with spaced microphones.
CARVER: I think that's right. I mean, I don't see how you could. You have to have the timing cues just right to produce the illusion of that sound being ten feet back or less... But how do you do it?
EARGLE: There is one area of recording that we ought not to leave unspoken about here, and that's Jack Renner's approach of three spaced omnis. And the spacing we're talking about is roughly maybe nine feet between microphones.
McGRATH: Collectively?
EARGLE: No. Eighteen feet between the outside ones. That's roughly about it.
CARVER: You say he puts the microphones 18 feet apart?
EARGLE: That's right. But there's one in the middle.
LIPSHITZ: Mixed.
CARVER: Okay. Mixed.
EARGLE: Now, the middle mike normally is a little bit down. It's normally 4 to 6 dB down, and the way Jack normally sets that is to pull it out and then bring it in until this vast hole in the middle now has a...
McGRATH: A fill.
EARGLE: ...a fill to it. So, when you do that, you have avoided the whole middle because you've just filled it in. Okay, in terms of stereo, you're now talking about two stereo pairs. One left and center, another one center and right, being subtended by the listener over narrower angles. And the net result of that is that left center information in the orchestra does sort of come out on the left center; right center information comes out on the right center, and you have very good localization of the middle of the orchestra, the left of the orchestra, and the right of the orchestra. And a lot of sins of spaced microphones are alleviated by that. Now, I can normally spot that in a recording because I can shut my eyes and really lock in—the winds, for example, all sound like they're right in the middle. Normally, in my recordings, or those that are done with a coincident or quasi-coincident pair, you hear the flutes over here, oboes over here, clarinets over here, and bassoons over here—because the winds, depending on how many there are, can be seated the length of this room here. Sixteen winds would occupy this space—and two rows deep. Anyhow, the left-center-right spaced-omni approach has a lovely texture to it. The imaging is not as analytical as what I like, but it's good. I give it a B minus on imaging and an A plus on texture.
EDITOR: When I sit in the concert hall, I don't get anything better than that B-minus imaging, as you call it.

CARVER: That's right.
EDITOR: In a typical seat in the concert hall, I have a sense of left, center, right, front, back, and under the very best conditions one point in between these.
LIPSHITZ: Right. But the microphones, of course, are not as far back as you would be in the concert hall.
EDITOR: That's right.
EARGLE: They wouldn't work that far back.
LIPSHITZ: They would not work that far back. And this is, of course, stereo versus surround sound. There is no way you can put the microphones in there for stereo recording where you'd prefer to sit in a concert hall. If you could do a true surround-sound type recording, then in principle at least you'd want to put the microphones where you like sitting, and then when you reproduce that sound you would have the same statements that you could make about it. I can't tell exactly where anybody is in the orchestra, and there's the hall ambience all around me.
EDITOR: But why aren't you happy with that kind of presentation in ordinary stereo?
LIPSHITZ: Because we're not reproducing the ambience around you. It's all coming from the front.
CARVER: That's right. Now...
EDITOR: Wait a minute. Do you mean to say that under those circumstances...
LIPSHITZ: You've got to move the microphones closer.
EDITOR: ...you need more precise information in order to get an illusion of reality?
LIPSHITZ: Yes. The reason why I reject the argument that imaging is an overrated aspect—because you don't normally hear the location with that kind of precision from where you normally sit in a concert—is that the recording is made from a much closer location, from which there would be no doubt, if you sat there, where the instruments were.
EARGLE: Well, the conductor himself normally...
LIPSHITZ: Has no trouble.
EARGLE: ...has no trouble telling where everybody is.
LIPSHITZ: Yes. You know, in the third, so-and-so was off key on such and such.
CLARK: Why should you be trying to duplicate what the mikes hear?
LIPSHITZ: Because, if you can recreate accurately a replica of the original perception that you would have gotten from there...
CARVER: If you were close...
LIPSHITZ: ...then you can do other things.
CARVER: Yes, that's right.
LIPSHITZ: Then you can do other things. Then you can make it wrong—in many ways. But if you can't make it right in *any* way, you're worse off.
McGRATH: I agree with that. That's a profound thing for me, and I agree with that a hundred percent. Yes. That cuts to the core of the illusion.
CARVER: Let me give you an example, an example of what Stanley just said, in actual practice.
LIPSHITZ: You can control the illusion.
McGRATH: That's exactly right.
CARVER: During the development of sonic holography, one of the things I had to undo was the spatial distortion associated with loudspeaker reproduction. During its development I worked very hard, and I could do complete ventriloquism. I had a person walking around banging castanets together, and I could bring the castanets right up to my right ear, and I could have her whisper in my right ear.
LIPSHITZ: How did you mike this?
CARVER: With two speakers. And...
LIPSHITZ: No, but how did you mike it?
CARVER: I'll tell you in a minute. Another thing I could do was—I made a recording in which I fly combat model airplanes, where we have two model airplanes buzzing around the sky, trying to outmaneuver the other and cut off this ribbon that he tows, and the combined speed is about 240 miles an hour. It's really exciting. You know you've been alive when you've flown a combat match. *(General tittering and* sotto voce *wisecracking.)*
LIPSHITZ: What size are these?
CARVER: About this big. *(There you go again, Bob.—Ed.)* The engines are two horsepower.
LIPSHITZ: Three feet long. *(Thank you, Stanley.—Ed.)* Okay. *(Wisecracks continue in the backgound.)*

"...that's a very powerful demonstration that you've correctly undistorted the spatial distortion, if you can ...walk a sound around the room anywhere at will."

CARVER: And when the airplanes do a loop—how round is a loop?—well, it's a large, round loop that goes up, overhead almost, and then back down again. That was the most difficult to achieve. I'd bring the recording inside my house and I would play it back through my processing, and the poor little loop would be squashed—tsoo-tsoo, like this, pfoom, pfoom, pfoom—so I could never, and I haven't yet succeeded to this day, to make my round loop in the house; it's always squatted, squished, but I can, horizontally, have a person come around and whisper in my ear.
LIPSHITZ: How high is your ceiling? That's the problem! You should raise your ceiling. *(Laughs.)*
CARVER: Believe me, I've done it; I've put my speakers outside; I mean, I've gone through the whole thing...
LIPSHITZ: I know.
CARVER: So... I've sort of forgotten what I was...
EARGLE: Well, you can answer a question about how you recorded this—when a woman came up and whispered in your ear. You were going to tell us later.
CARVER: When she did that, I had her come right up to one microphone, talk into one microphone.
LIPSHITZ: But what were you using? Coincident recording? Spaced mikes? What mike technique were you using?
CARVER: Oh. I had a piece of cardboard, a large piece of cardboard; I glued a pillow, a large pillow, on one side, and I had—this was, now remember, 10 or 12 years ago—two Nakamichi microphones glued on either side of the pillow, and I had cut some holes in my large piece of cardboard.
LIPSHITZ: These were omnis? Well, it doesn't really matter very much. Okay, so you had a baffle between your two mikes.
CARVER: Uh-huh. I had a baffle between my two mikes.
EDITOR: This is Bob all over. I love it.
CLARK: It's great.
EDITOR: This has your signature.
CARVER: Yeah.
LIPSHITZ: Well, you know, there are people who record with that sort of arrangement. I mean, not the pillow and the holes...
McGRATH: Yes. The Jecklin baffle between omnis. What do you think about that? Does that give us some coincidence with omni characteristics?
LIPSHITZ: No, it does not give you coincidence. No way. It is similar to those weird baffles that some of the people with PZM's are using, where they have these PZM's on opposite sides of a piece of Plexiglas. Because they are just a quarter of an inch apart, the Plexiglas being a quarter of an inch thick, they say this is coincident recording. Not at all. The acoustic path length from the one to the other is many feet. It's a baffling; it introduces level differences as a function of angle and frequency due to the diffraction effects, so that you produce level differences and time differences at the mikes. I haven't tried it.
CARVER: I remember what my point was in that story. It was, if we can learn to do that kind of ventriloquism, then perhaps we have a handle on how to undo the spatial distortions in the recordings. Because that's a very powerful demonstration that you've correctly undistorted the spatial distortion, if you can completely do ventriloquism and walk a sound around the room anywhere at will. Except I couldn't in a million years take that and record an orchestra and make the orchestra come out sounding real. It's a puzzle to me just what should be done—how to get the ambience component of the soundfield and the direct sounds and have it all work out. Now what I've heard so far—and I'm learning about making recordings—is two nearly spaced, nearly coincident microphones, two flanking microphones, and there are more microphones in the back sometimes?
EARGLE: If you need them.
CARVER: If you need them. Okay. I understand. I got it. And the two flanking microphones really do contribute mostly to the ambient soundfield, the out-of-phase components of the soundfield.
EARGLE: Probably. Yes.
LIPSHITZ: Well, let me give you an example of one recording which I did of a brass group in a large church. The recording is extremely reverberant, but the imaging is still quite precise—because it's a coincident recording. But the venue is very reverberant. I doubt whether people would

say that it's not reverberant enough if they heard it. I think you'd probably feel that it was reasonable—or at least you might agree that it was a reasonable balance. Most venues are not reverberant enough for a coincident-type recording to sound reverberant.
CARVER: That's right! And that's why they're so disappointing sometimes.
LIPSHITZ: That doesn't mean it's not reverberant; that Nimbus Equale Brass CD I suggested to you does not *sound* reverberant. But when you ask yourself what causes you to be able to tell that there are enormous differences in distance between you and the instrumentalists, it *is* the direct/reverb ratio in the recording. That's the only way you can tell that so-and-so is closer than so-and-so. So it *is* reverberant.
CARVER: So the clue or the cue that allows you to understand that something is soft and close instead of loud and far is the...
LIPSHITZ: Direct/reverb ratio.
CARVER: ...cue or clue associated with the ratio between the direct sound and the reverberant sound.
LIPSHITZ: Yes. Now normally, in a concert hall, you don't sit down and say, when the orchestra plays, oh what wonderful acoustics. You're not normally aware of the reverberant sound. If you are aware, it usually means something is wrong.
EARGLE: There's too much of it. Yes.
LIPSHITZ: Too much, or nasty things, echoes, or some features that are not nice. So, the things you're consciously praising about recordings are things whose presence you are consciously aware of. I believe that in many ways they're wrong. But I understand the desire for them. What I'm reluctant to give up in order to achieve that sort of sense of involvement in the sound is the precision of the direct sound—because in no natural situation do you have phasiness on direct sound. You can't. And that's what worries me.
CARVER: No, there shouldn't be any phasiness on direct sound. I agree. I have one more question, something I just want to understand. If I had a hard L + R signal, mono signal, between my speakers, and it was a really hard signal, and imagine that I had an echo associated with it but it was completely in an L – R channel, and I had a control so that I can turn off the L – R or turn up the L – R at will—and let's make it a toy duck quacking, a mechanical toy duck going quack, quack, quack, quack—and there were some echoes associated with it, and if I began to turn the echo up, my subjective perception would be that the duck would begin to move further away?
LIPSHITZ: Oh yes. With a Blumlein recording, coincident figure eights at 90°, and this means really good figure eights—the Schoeps figure eights, by the way, I've always used at 85°, the criterion being the fact that they get slightly sharper at high frequencies, and you've got to trade off a solid central image versus one that's sort of receding a bit and giving you a slight hole in the middle.
McGRATH: How far back do you position the mike? Do you allow the spacing of the musicians to exceed a 90° angle?
LIPSHITZ: Oh, I wouldn't mind because that means the image will go slightly beyond the loudspeakers. That's fine; you can go, you know, 10° more, either side, and within a certain range of antiphase signals in the two channels you get imaging beyond speakers—then it starts becoming oppressive phasiness on the ears.
EDITOR: You haven't finished your thought, Stanley. What about the Blumlein...?
CARVER: What about my duck?
LIPSHITZ: I haven't finished my point. With a good Blumlein recording, with let's say textbook-perfect figure-eight microphones, your volume control is a distance control to a very large extent.
CARVER: Okay. I believe that.
LIPSHITZ: And you can try that. As you turn the thing up, you'll find that it's almost as if you were moving closer to the sound.
CARVER: Well, in this case, as I turn the L – R up, I move further away.
LIPSHITZ: In fact, Peter Walker is perhaps the first person who talked about that, if you remember the instruction books that came with his preamplifiers...
CARVER: Yes, I remember that.
LIPSHITZ: ...or perhaps even come with the current ones; I'm not sure.

"I would like somebody to tell me why the microphoning has to be more location-specific, more image-specific, than what we are accustomed to in the concert hall."

McGRATH: The most important control.
LIPSHITZ: He would say that there is generally a correct volume setting.
CARVER: That's right, that's most realistic.
LIPSHITZ: And that is true. If you turn it too high, the lateral perspective is wrong for the apparent distance perspective. If you have it too low, again it's too wide for the apparent distance. There's an "about right"—and if you did the recording yourself, you either agree with that or you can fool yourself into believing that you hear what you know is about the correct perspective.
EDITOR: Does that phenomenon still exist with spaced microphones?
LIPSHITZ: No.
CARVER: The effect, though, of volume? The more out-of-phase level, the more distant it becomes...
LIPSHITZ: Not in the sense that I'm talking about it, though. You don't think of the volume control as being that kind of distance perspective control, that things click in just correctly when you set it.
EDITOR: Is there still one correct volume for spaced microphones?
CARVER: Yes.
LIPSHITZ: I don't think so. I don't think so.
EDITOR: You don't think so.
CARVER: My final question with my duck, Stanley. If I begin turning my L – R signal up, the duck begins to move further away. Now, obviously I would think it's a louder duck, so in my mind my mechanical duck would move further away and be quacking louder. Obviously, if he is moving away, and I certainly haven't changed the volume of the L + R, then it must be that he is quacking louder. So it's sort of the acoustic analogue of what artists have done for years with railroad tracks. Perspective.
EARGLE: Well, they show an object the same size against a perspective. Yes. And the foreground object is only maybe three units high here. But the room perspective looks like this, you see. So that same object, the height back here...
LIPSHITZ: Looks enormous.
EARGLE: ...looks much larger.
EDITOR: Nobody can see your drawing, John, on the tape.
CARVER: Exhibit B is a classic drawing of an artist's perspective with telephone poles and a street.
EDITOR: Okay. I would still like one of you to tell me—maybe Stanley, since I think you're the most vociferous advocate of this particular approach—I would like somebody to tell me why the microphoning has to be more location-specific, more image-specific, than what we are accustomed to in the concert hall.
LIPSHITZ: Because you have no sense of sight when you're listening.
CARVER: You have to make it come alive.
LIPSHITZ: You're listening in an acoustic which is not a concert hall.
EDITOR: Suppose I'm blind.
CLARK: Stanley said one of the reasons was because the microphones *are* closer and the microphones *can* hear those positions better. I don't buy that, but I think that's what he said.
EDITOR: I know he said that, and that...
EARGLE: Well, okay. Can I offer a way around this? Let's say we take anybody's modern recording of an orchestra, where there is good hard left and hard right information, either by virtue of being on the major axis of a figure eight and the null of a figure eight and that sort of thing or whatever—whatever technique you might use. Who in this room would take that recording and take something that we used to call a blend control and pan that in, to make it sound the width which you might hear in Row M?
CARVER: Nobody.
EARGLE: Not a soul!
CARVER: End of argument. That's it. That's true.
EDITOR: This is the best-seat-in-the-house argument. Right?
EARGLE: Yes, yes, yes.
CARVER: Yes. You get a better chance.
EDITOR: But there's more specificity of localization and imaging in some of these recordings than in the best seat in the house.
LIPSHITZ: Well, no, but John's point is this—that if you sit at a distance such that you have a 60° subtended angle for the or-

chestra, you're not in the best seat of the house; you're much too far forwards. What he is saying is, if you sat that close to the orchestra, the specificity would be pretty good. I think that's what he is saying.
EARGLE: Uh-huh. Yeah.
McGRATH: I'd like to hang from my light bar that I put my mikes on in the concert, sometime; I'd just like to hang from there, just to see what it's like. The sound must be extraordinary!
LIPSHITZ: The sound the conductor gets must really be quite spectacular.
EARGLE: Oh, it is.
EDITOR: And deafening.
EARGLE: I've been out there. I've listened from the podium a number of times, you know.
McGRATH: It's extraordinary.
EARGLE: Anyhow, can I register a very strong complaint against a lot of pop/rock product that's being generated today?
McGRATH: One? Only one?
EARGLE: Well, the main one is this—that all musical values aside, some of the stuff that is mixed down from 24, 32, and 48 tracks is so center-heavy that it just drives me crazy.
CARVER: I know.
EARGLE: I mean, my God, you're paying for stereo—why don't you get stereo for Christ's sake! You know, it's all piled in the middle so much of the time, and maybe you'll hear a lead guitar over here part of the time and something over here. And if you look at it on the scope, it's just really sort of a 45° blob. There's almost no original stereo recording in there; it's all panned mono.
LIPSHITZ: Mono. Yes.
EARGLE: Maybe the output of a Lexicon is in...
CLARK: John, I don't know how you can say that. I've recently searched for a bunch of recordings that were extremely phasy, had very little correlation between them, and I had no trouble finding them—and they all had to be pop recordings.
EARGLE: Were they new recordings?
CLARK: Yeah, like Tiffany, for instance—big, swishy, swirly things, and a little voice in the middle...
EARGLE: Well, good for them.
CARVER: They're getting the hang of it now.
EARGLE: It's about time.
CARVER: The last year there've been a lot them that have come out.
CLARK: They vary all over the place.
EDITOR: Okay. Name a few intelligently recorded rock/pop albums.
EARGLE: I decline because I really don't know. I mean I'm not that familiar with the scene.
McGRATH: Dire Straits?
CLARK: Yes, Dire Straits is another whole thing.
McGRATH: Yes, it's wonderful.
EARGLE: Now, there are a few jazz things that I've done. I wouldn't do what I do if I didn't like it, basically, but that doesn't mean that I praise everything that I've done; Lord knows, we've all made mistakes. But to me the real mark of a good pop or jazz recording—a natural acoustical recording—is that it's a very nice mixture of real stereo recording plus close-panned images when you need them. You have to mike a soloist at a distance of about 18 inches to two feet.
CARVER: That's where you put the microphone for soloists?
EARGLE: And a bass has to be miked; a guitar has to be miked; otherwise, if you rely upon natural perspectives, they're going to be drowned out by the drums and everything else. But the drums can be miked in stereo, beautifully; a piano can be miked in stereo...
CARVER: Oh, I hate pianos miked in stereo. I hate it; I just hate it.
LIPSHITZ: When you say miked in stereo—what is it that you hate about a stereo piano? You mean you don't like a 60° wide piano?
CARVER: One mike stuck on the treble end and one mike stuck on the bass end—that's what I hate.
EARGLE: Well, you don't quite do that.
CARVER: Okay. Stereo back away is fine.
EARGLE: A piano is normally panned left to center and not from left to right...
CARVER: That's fine.
EARGLE: ...and the drums are normally panned from center to right. And you get a very nice cohesion between them.
CARVER: John, the microphone is 18 inches away from the soloist—do you do that also with opera singers?
EARGLE: No!
CARVER: No? Much further out—okay. Just pop.
EARGLE: Well, I've never done an opera, but I wouldn't be that close.
EDITOR: Let's talk about some of your other equipment in your recording sessions—your digital processors, consoles if anything—what is the current state of the art *past* the microphone?
EARGLE: For classical—unless you're doing actual panning and need a pan pot—all you need is a summing amplifier on the output of a string of mike preamps, and you need one for the left and one for the right, as simple as you can get it. The only filtering, the only EQ I ever do on a classical date—ever have done—is to roll off the extreme low end if there's a discrete rumble or something like that—and that's a bad room.
McGRATH: Air-conditioning noise, blower...
EARGLE: Yes. But otherwise never any putzing around at the high end.
McGRATH: I hold a similar philosophy. I basically don't even have a summing network, though I've been using it now on my concert recordings, but basically I've just been using—specifically—a Jensen two-channel thing...
EARGLE: It's a lovely preamp.
McGRATH: The mikes go into it, and the signal of that goes right on out into the tape recorder.
EDITOR: Those are JE-990's, right?
McGRATH: Yes. And that's it.
EDITOR: Well, what about digital processors? There's been a lot of talk about the Colossus and similar stuff. Any comments on one kind of A-to-D conversion against another?
LIPSHITZ: Well, I think the differences are questions of accuracy. If you want an accurate A-to-D conversion, then make sure that the system is either properly dithered or there's enough noise in the mike feeds to do that.
EDITOR: There's a lot of inaccurate stuff around, isn't there?
LIPSHITZ: Oh, yes, the A-to-D conversion is much more difficult than the D-to-A conversion.
CLARK: Before you throw out 16 bits, make sure you've got all 16, right? Before you go to 18, make sure you've got your 16.
LIPSHITZ: Well, 16 good ones is better than 18 bad ones.
EDITOR: That's for sure.
LIPSHITZ: And the same is true of the D-to-A's, mind you, where the number of bits is proliferating, and that doesn't mean they're better. But that being the case, I think we're talking engineering details, which in principle I don't believe have significant effects on the sound.
EARGLE: They're all vectoring in on the same solution. I mean, if they're all perfected, whatever they're doing, the answer ought to be the same.
LIPSHITZ: Yes. You've got single-spot CD players and three-spot laser optical systems—that's neither here nor there.
EDITOR: So you don't go for this "hey, man, of course it's better, it was made with the Colossus, man"? What do you think is better?
EARGLE: The only difference between the current processors right now would be the filtering, the monotonicity of the devices and so forth, and the dithering. Normally, there's enough dithering caused by the self-noise level of microphones.
LIPSHITZ: For classical recordings, I think invariably. The digital system's noise floor is below the mike feeds. I mean, you could try that on PCM-F1 type processors, where you can switch from 14 to 16 bits.
EDITOR: There is hiss on nearly all of my CD's. Where does it come from? It comes from the mikes.
LIPSHITZ: Yes, of course. But there's hiss everywhere. And there ought to be. Because, quite honestly, if you had a digital recording that had hiss while the music was playing and silence between notes, you've got noise modulation. I mean, that's an indication of something not being right. There ought to be a steady, thermal, white-noise floor in almost every recording.
EDITOR: But not so high in level that you're aware of it when you're not listening for it.
LIPSHITZ: Well, the mike noise may have been rather high. But anyhow, the point I

"...if you had a digital recording that had hiss while the music was playing and silence between notes, you've got noise modulation... something not being right."

was making about the PCM-F1 type processors is apropos of this question. You can switch from 14 to 16 bits. The dither is just done by the inherent noise in the circuit, and the noise floor doesn't change when you switch the number of bits. It's properly dithered at 16—slightly overdithered—and underdithered at 14. If you ask me the question which should you use in a recording, I'd say to you this: Have your mike faders up at the normal level at which you'll use them during the recording, crank up your monitoring level, listen to the sound, and switch between 14 and 16 bits. And if you cannot hear any change in the noise floor, the digital noise floor is dominated by the noise coming in on your mikes—use the 14-bit mode because that gives you greater error protection and it's being well dithered by the incoming signal on the mike lead. If you hear any change in the noise floor when you switch from 14 to 16, the digital system's floor is dominating the mike feeds—use the 16-bit mode because now you need the dither in the recorder to dither the system properly; the mikes' hiss is not going to do it. But here we're talking about noise that's close to the digital floor; what Peter just mentioned there was a noise floor in the recording that's way above the 16-bit floor. Well, that's being limited by something that's not the digital recorder, unless there's something vastly wrong with the digital recorder.

McGRATH: Generally, that's mike preamps—and microphone self-noise, too, to some extent.

EDITOR: What about consoles? There have been a lot of bad consoles.

EARGLE: Well, of course, the minute you speak of the necessity of a console, you're talking about pop recording because for most classical you're really looking at, at the very most, two sets of summing amplifiers—a whole bunch of preamps feeding into one sum and a whole bunch of preamps feeding into another in the other channel. But the trouble with consoles is that console builders play with architecture the way I noodle an eclectic factory in Los Angeles. Anytime they want to change direction—I go around the corner—they put in a summing junction to go from here to there. And they overdo it—there are so many active devices in a console that it's almost frightening. And I guess the reason why that happens is that once you have enough of these, you can double the number and get a certain decrement in performance; then you double that entire number for the same decrement—so the more you have, the more you can use with impunity. At least it seems that way. The real test of a console is to have a hard-wire bypass around it—around as much of the architecture as you intend to use—and switch it. The way I've done this in the past is to take a little L pad with a 200-ohm light across the middle to simulate the source impedance of a microphone and to feed into this a stereo program and to bring the level down so it really corresponds to what a good capacitor microphone might be giving me. The figure I take is 94 dB SPL, in other words one pascal, to give you 7 mV out. And I simply run that through the board, bring it up to zero level, and then have a path around the board. Then take a good preamp and have both of these coming in, set levels, and then switch between the two—put headphones on or speakers, whatever you want to listen to at that point. If you hear a difference when you do that, that's not attributable to level, then you've got a problem.

CARVER: Does that happen often, the problem part of it?

EARGLE: Well, the answer is that I've found very few consoles, at least using the part of the architecture that I normally use, which is very, very simple.

McGRATH: Okay, that's a big difference.

EARGLE: I try to bypass all the monitor section, the mark and mix, and all that bullshit; I try to avoid that.

McGRATH: So you're really barely using it.

EARGLE: That's right. I'm using mainly the mike preamps, up to a pan pot.

LIPSHITZ: I don't use consoles...

EARGLE: You don't have to use them...

LIPSHITZ: ...because I have very simple recording things; I built some of my own mike preamps and so on...

EARGLE: But most of the problems are the ones I've run across.

"The real test of a console is to have a hard-wire bypass around it—around as much of the architecture as you intend to use—and switch it."

LIPSHITZ: ...but I would do exactly that. Take it out as soon as you can.

EARGLE: As soon as you can. And bypass as much as you can internally. But when I don't hear a difference in the sonic quality, what I do is turn off the music and crank the gain way up, and then listen to the noise of the system. There are a lot of consoles that are prone to local RF problems, TV sync coming in, and things like that, that will drive you crazy. There are some consoles that will take long microphone cables coming in, and you'll have a problem. The same cable on another console won't have a problem. You disconnect the cable—the problem goes away. You're looking at electrically unbalanced inputs or transformer inputs. There's nothing wrong with a good transformer input, I'll tell you that much.

McGRATH: It's just good engineering versus bad engineering.

CARVER: Aren't the transformer inputs the best?

LIPSHITZ: There are clever transformer inputs, too, you know. There's an idea that I think Peter Baxandall came up with, for a transformer input with feedback around the transformer to keep it operating in the zero current mode.

CLARK: Barry Blesser did that years ago.

LIPSHITZ: So that it can have zero distortion, no magnetic saturation things, and frequency response to as low a frequency as you like—you know, hundredths of a hertz.

CLARK: A question. You mentioned some allegedly bad consoles. How did you verify, or how does anybody verify, that they're bad? Now, I just heard what John said about RF problems and noise floors and noise quality and so forth—is that what we're talking about? In that case I can agree with it; there are differences. But I have designed quite a number of consoles—I used to work for a console manufacturer and designed them, and they had many, many parts, not *so* many in those days—but what is a bad console and how do you know? What is this bad sound? Is it bad imaging or something big like that? In other words, I think a lot of them are given a bad rap without really testing them.

EARGLE: Like so many things in this business, you know, like anything large and expensive—the first thing you do is go up to it and kick it, or pick it up, or touch it; you feel how cold it is to make sure it's really metal and not plastic, and things like that. In other words, a lot of it has to do with the quality of the switches, the feel of the switches—the tactile qualities, as opposed to the audible qualities.

CLARK: Well, that certainly makes my point.

EARGLE: The point is that the word bad is defined by all of these things. I mean a switch that will put a click in the signal when you engage it or disengage it—some will, some won't, it depends on how they're made—some consoles you switch in the middle of the recording and you'll have a level shift going out or a click in your output; others you won't...

CLARK: But these audiophile recording companies—like Telarc—are making a big deal of what kind of console they use and what kind of op amps are in it and all of these exotic things...

EDITOR: And what kind of wire in it...

CLARK: ...and what kind of wire in it—not seemingly the practical kinds of things you're talking about. And also people say, as you have, there's so much in the signal path. Now, there is a topology or call it a configuration of mixing buses, when you have a large number of inputs, where you can do separate summing junctions summing into other ones and come out with less noise than feeding them all into one summing junction—in which case I would say it's clear-cut that you can use more parts and achieve better performance in that area. You go through a hundred of these console-grade op amps and you come up with less distortion, far less distortion, than some of the most highly reviewed tube-type audiophile amplifiers. So where's this badness? Have we actually found it, or are we just sure they are bad?

EDITOR: Well, I'm not an active practitioner, as you know. When I said "bad consoles," this was hearsay, and the hearsay was based on early op amps that were apparently of less good performance than the more recent ones.

McGRATH: Would you agree with that?

CLARK: Twenty years ago, they used op amps with a slew rate of 0.5V/µs, and in a system that has a nominal operating level of 1.23 V *[1 to 3 V?—Ed.]*, that can under extreme conditions cause a problem.
McGRATH: But if those extreme conditons were not met, would said console using those op amps be less good than a current one, designed today?
CLARK: Yes.
McGRATH: It would be less good—or as good?
CLARK: Less good.
McGRATH: Why?
CLARK: Because you could come up with a worst-case scenario instrument, a real instrument with fast rise times or an electronic instrument plugged into it that would exceed the 0.5V/µs slew rate of these 20-year old op amps.
McGRATH: But assuming that you're simply implementing it with just microphones, recording music. Would it sound less good than a current, contemporary design? And if so, why?
CLARK: My belief is that with distant microphones, classical miking techniques such as you talk about, you probably would not be able to hear the difference using these 0.5V/µs op amps.
LIPSHITZ: The current op amps are better, I think, in all measurable respects. Like the output stages of the best ones are quite low in crossover distortion, without negative feedback, so that the nonlinearities inherently, I think, are lower. But your point, Peter, is that these would not be stressed, probably, under normal conditions—but the modern ones would also certainly be quieter, probably less prone to RFI, especially the FET input types. That must be a major headache in consoles.
CLARK: The better old ones, the kinds that I designed, were just as good as modern ones in noise and RF and distortion. They used discrete-component op amps with very wide bandwidth products.
LIPSHITZ: Ah. Okay. I was automatically thinking integrated op amps, I guess.
CLARK: Well, you know—what's bad? I take the worst, which is 20-year old op amp consoles—I don't want to make a case for those. I don't like those at all. Those weren't the kind that I designed.
EDITOR: Aren't they still in use, though, quite widely?
CLARK: Are they? Audio Designs—was that who made some of those?
EARGLE: I don't remember.
LIPSHITZ: You mean the consoles using 741's?
CLARK: Yeah, and 1558's and that sort of thing.
LIPSHITZ: I don't know whether they're still in use. They probably are in some places, but they're not being used by any of *us*, and I doubt whether any of the recordings by any of the companies that...
CLARK: I'd rather keep them out of the conversation; I don't wish to defend *them.*
EDITOR: Let me ask you this. Let's take a console that you totally approve of—that there's nothing wrong with according to your lights.
CLARK: Okay.
EDITOR: And you put every control in neutral, so that the console isn't doing anything except passing the signal through its amplification stages—but not altering the signal, at least not deliberately. And then you bypass it in the manner that John just mentioned. Would you say that it would pass that test?
CLARK: If you turn the gain up, or had a condition where you could hear the noise, it undoubtedly would have more noise than a bypass, almost by definition. If the noise issue was sufficiently low in level, I don't think that in a scientific test you would be able to hear a difference between the bypass and dozens of amplifiers in a console.
EDITOR: So the so-called minimal console recording or no-console recording, according to you, is a tweako fantasy?
CLARK: In my opinion, yes. I think all of this discussion about where to put the microphones is decidedly not a tweako fantasy—that's the real thing.
EDITOR: Oh, nobody even suggested that.
LIPSHITZ: Peter, I think, look—John was saying a little while ago that he would take a signal out as early as possible, use as few stages as he needed to use...
EARGLE: Well, I think that's good engineering practice in any discipline.
LIPSHITZ: ...and I think I would do the same. Not because I believe that the other stages make audible degredations; they will add noise, even if it's not audible noise—there's just no point in going through them unnecessarily.

"So the so-called minimal-console recording or no-console recording, according to you, is a tweako fantasy?" "In my opinion, yes."

McGRATH: It's a lot cheaper to do without them.
LIPSHITZ: And if you're buying, you wouldn't buy the console with those features if you didn't need them. I think there's a difference between your question, whether one could demonstrate that these were degrading the sound, or whether one therefore, being unable to demonstrate that, must necessarily use them because they do *not* degrade the sound.
EDITOR: Nobody is saying *that.*
LIPSHITZ: I agree with Dave. I suspect that many of the cases where people have ciriticized consoles are probably totally unwarranted in the way they were being used, but they were not consoles that are as good as ones that we can make nowadays.
McGRATH: But there is a pattern. Even if one accepts the premise that they're not audible, the evidence does speak to this—and perhaps it's just a function of a consistency of philosophy—that people who are inclined to position microphones carefully and use the proper microphones and perhaps minimize their number are also people who are generally inclined not to have a lot of miscellaneous electronic stuff. So that it naturally evolves that people put the credit where it really isn't necessarily due; where the real credit is that they're just doing the basic things correctly and leaving all of the crap out of the picture.
EARGLE: There's a lot be said for operating as simply as you can, if it's consistent with your miking philosophy. Because it's less stuff to schlepp, believe me, and it's less stuff to go wrong.
LIPSHITZ: Yes.
EDITOR: But there's this broader issue here; consoles are just one example. I mean the prejudice on the part of purists against a multiplicity of active stages in the signal path. The philosophy of—let's remove from the signal path as many active stages as possible.
LIPSHITZ: But I suspect that if you put out an audiophile recording saying, you know, made with custom-designed microphone preamplifiers feeding directly into custom-modified blah-blah-blah...
EARGLE: That's a record-jacket hype that's been around for 30 years.
LIPSHITZ: ...and you put out the same recording, identical disc, with a jacket which says, recorded on an ABC console fed into a factory-standard XYZ digital recorder...
CARVER: Forty-seven op amps.
LIPSHITZ: ...most people would find the latter decidedly inferior in sound to the former, even though they're identical, in other words the same disc.
EDITOR: All right. I can tell in what direction we're headed. *(Laughter.)* And we'll dig into that.
EARGLE: I remember, once at RCA back in the old days, back in the vacuum-tube days, I once counted the simplest path from a microphone to a stereo master lacquer—the path of that single input from that microphone, how many transformers it had to go through. And it was well up in the 20's, I mean something like 23 or 24. By the time it went through the console, and the console combining networks, and then into the next thing—the tape machine didn't have one, but the remix board had several, and the lacquer cutting chain had its share of transformers—and yet the end product wasn't that bad. I mean they were all fairly good transformers, let's face it—and you can make them.
LIPSHITZ: You can make very fine transformers, absolutely.
EARGLE: Yes, superb.
LIPSHITZ: They're better in many ways than active circuitry...
EARGLE: Yes, they are.
LIPSHITZ: ...because they don't have the ground problems and common-mode things. But there's one point that might be worth making, and that is that the additional circuitry in the console, which may not be audibly degrading sound in the sort of aspects we've been referring to now, will probably be passing it through a large number of coupling capacitors. Now, I'm not talking about that because I'm saying or implying that there's something about a capacitor that is nasty; I'm not implying that at all. I'm saying that there is probably a large number of high-pass filters. They may all be set at 5 Hz, but there is a very

large accumulated phase shift that is introduced in the lower part of the audio band, getting up to a few hundred Hz, by those high-pass filters, those coupling circuits. And there is evidence, published in the *Journal of the Audio Engineering Society*, some experiments that Laurie Fincham of KEF did, for example, that those things potentially are audible; those phase shifts are audible. Maybe we need more people to do some more experiments, down at the bottom end. Phase effects normally are very subtle, but the implications of this paper are that, yes, we want high-pass filters—you don't want to be feeding 0.5 Hz and noise and junk and room pressure changes to your subwoofers—but perhaps we need linear-phase high-pass filters that will roll off below 5 Hz or 10 or 20 or whatever but maintain the phase response as linear phase down to very low frequencies, so that the phase distortion isn't a factor. You see, a high-pass is much worse than a low-pass. Almost any low-pass filter is essentially linear-phase in its passband. A linear-phase filter is not time-dispersing; it's waveform-preserving, almost totally. And it doesn't matter whether the low-pass filter accumulates quite a large phase shift; it's linear phase shift, pure time delay, a few microseconds or milliseconds—totally nondistorting. But a high-pass filter with the same amount of phase shift, because it's happening in the inverse direction as a function of frequency, is very phase-distorting, very waveform-distorting, and potentially audibly so.

CLARK: I've read that KEF thing; in fact I was there when he gave it, and I've studied that. I think that the audibility of that phase shift at low frequencies possibly is better looked at as an audibility of the time dispersal rather than the degrees of phase shift...

LIPSHITZ: The group delay.

CLARK: Yes, the group delay. Because 45° or 90° or some such small amount of phase shift at 20 Hz amounts to many milliseconds. And it's not too surprising when you start talking about the milliseconds involved that it might be audible. Again, I think group delay wins out as a concept over phase shift in terms of what's audible and what isn't.

LIPSHITZ: Well, as long as you know how to interpret the numbers, both of them are ways of quantifying it. An interesting point, of course, is that making linear-phase analog circuits, or phase-corrected analog circuits, is relatively expensive. Digitally it's very easy to do these things.

EDITOR: Dean Jensen put out a pretty good paper on what you just said, Dave—what kind of phase shift is relevant and what kind is not. Basically that flat group delay is not audible.

LIPSHITZ: With one proviso, one qualifying comment. Yes, you're talking about meaningful and relevant parameters, or something, in high-frequency phase distortion...

EDITOR: I have it around somewhere.

LIPSHITZ: Yes, I know the paper you mean. And, yes, that is all perfectly correct. The implication, however, that these high-frequency phase shifts are audible, which is in that paper, is quite unsubstantiated.

CLARK: That's what I thought. He always implies that.

LIPSHITZ: Yes, he implies that, and I've asked him, and he can't substantiate it and won't even try to. He'll say, for example, I know you disagree with me on that and don't want to discuss that issue. That's all very well and good, but it doesn't prove anything one way or the other. *(The recent tragic death of Dean Jensen leaves this controversy without a conclusion.—Ed.)* All the evidence is that phase shifts above a few kHz—phase shifts of modest amounts, which is what we're talking about—are inaudible. But phase shifts of modest amounts below 500 Hz can be audible—and there's indication that perhaps significantly so at low frequencies—and need to be thought about. Something of interest—I haven't had a chance to listen to the disc yet—but there's a very interesting CD that has just been put out by the Acoustical Society of America. It's available for 20 dollars, 17 each if you buy five or more—that's for members at least—and it's audio demonstration tests, 39 tracks on this disc. It's been produced in conjunction with Philips/Polygram, the Institute for Perception Research in the Netherlands, and Russing at—is it Northern Illinois or whatever?—and they have digitally regenerated all the Harvard psychoacoustic tests, on tapes released by Harvard in '78 and rapidly distributed—none of them are available. They digitally generated all these things, and it contains demonstrations of many psychoacoustic phenomena. And that could be extremely interesting to many people; you will find demonstrations of interchannel level differences, interchannel time differences, the audibility of reverberation; for example they have an anechoic and a reverberant voice—I think it's voice—track, where you're aware that this is a reverberant voice—it's in a room, it's not an anechoic voice—but you're not normally very aware of the reverb. And then they play the voice backwards.

"All the evidence is that phase shifts [of modest amounts] above a few kHz are inaudible. But phase shifts of modest amounts below 500 Hz can be audible..."

EARGLE: And you hear the tail coming in.

LIPSHITZ: Yes, the tail comes in first. You're absolutely aware of it then—this is reverb there, absolutely.

CLARK: I have that disc. I've listened to about the first half of it; it's fascinating. A lot of it is pretty dry; it's not really entertaining; it's instructional material.

LIPSHITZ: But if people are interested in some of these psychoacoustic things, there is a 93-page booklet that comes with it.

EDITOR: Acoustical Society of America? How long has it been available?

LIPSHITZ: A few months; it's dated 1988.

* * *

EDITOR: Let's talk about storage media—LP records, analog cassettes, CD's, digital audio tape, and so forth. What do you think the future holds? What do you think is going to happen to these various media? Are they all going to survive?

EARGLE: Can I tell you? The data as I see it tells me that the LP—outside of a few specialty houses like Reference Recordings and Dave Wilson and people like that—is really doomed. I think major companies will put it out for certain market segments, like jazz, or rhythm and blues, or country and western, where the population base hasn't had the infusion of CD players that the rest of the world has. But I think the classical business in the hands of the majors is going to go basically compact disc and cassette. The Philips cassette remains the strongest medium that we have.

EDITOR: That's what you're being immortalized on right on.

EARGLE: Right. And I think that you're going to find more and more people duplicating cassettes from a digital source and hard memory, instead of having a running master tape, a duplicating master. I think that the vote is still out on the DAT as a possible newcomer, or certainly as a replacement for the Philips cassette. First off, the Philips cassette is much better than the LP was relative to what wants to replace it. In other words, the LP was replaced by something that was much better, much more easily manufactured, quieter, and so forth.

EDITOR: Many tweaks would disagree with you, but...

EARGLE: That's right, that's right. But by and large the Philips cassette, well duplicated, especially in Dolby C, really rivals, for quick A/B checking, a CD, except on certain kinds of music—you know, a piano recording, something like that. So the need for the DAT, as a carrier for prerecorded music, is pretty slim. As a device at *our* level, for recording, I think the DAT is really going to just take off; it already has taken off. I think it's going to replace a lot of machinery out in the field. But I don't really know that it's ever going to make it as a carrier of prerecorded program material.

McGRATH: It might be doomed to the same oblivion that the F1 was.

EARGLE: Right.

McGRATH: Who knows? I mean, it may fall into the hands of those of us who adore its functionality and its qualities, but it may...

EARGLE: Absolutely.

LIPSHITZ: Of course, the F1 was intended, I think, for the semiprofessional market. It was not intended as a consumer product.

McGRATH: Well, interestingly, I was a dealer for Sony, and that wasn't even marketed through their ES division, which was their high-end home division. It was strictly a consumer product.

LIPSHITZ: Well, in Canada now, for example, the PCM-601 is available from Sony,

but only Sony professional. It's not available as a consumer product, as far as I know. But if you want prognostications, Peter, on these things, I think the LP is clearly going and will soon be gone.
EDITOR: Do any of you regret that?
CLARK: No.
EARGLE: I don't.
LIPSHITZ: Well, not greatly...
McGRATH: I do.
EARGLE: There's a little nostalgia, but...
LIPSHITZ: ...because I don't believe that the new media are inferior, you see. I suppose you regret it in the way you regret something you're long familiar with...
EDITOR: I don't think that's what Peter means.
LIPSHITZ: I know.
McGRATH: I'm still one of those backward flat-earth or black-vinyl types.
EDITOR: You're a tree-worshiping analog druid.
McGRATH: That's it, exactly.
LIPSHITZ: I do disagree with John, though, about cassettes. But I guess I'm a bit of an unusual individual here—I do not own a cassette deck. I do not own a cassette deck because I've had a lot of experience with cassette decks. Our chamber music society every year, in order to make up for deficits, gets permission from some of the artists—the best concerts we've had during the year—to make copies on cassette and sell them to our members for donations.
EARGLE: In what way do you not agree with me? You don't think it's going to hang on as a strong medium?
McGRATH: They sound terrible—please! They're awful.
LIPSHITZ: They're not flat; they change from day to day; after a few months they start wow-and-fluttering...
EARGLE: Oh, of course.
LIPSHITZ: I mean, just a month ago, we had a violin-piano recital; the violinist lives in San Francisco, and he got his cassette copy—you know, we give a copy to each of these people free—and I get a call from him: it's a semitone sharp! A copy a semitone sharp that's duplicated directly from the digital master tape? On all our cassette decks we match carefully with a test tape, so that when we do our dubbing one is not going to run out ten seconds before the music ends for side one and that sort of thing. And I measured all our cassette decks, and none of them was more than 1% off in speed, which is 1/6 of a semitone. But this person has absolute pitch—maybe he didn't mean a semitone; maybe he meant a significant fraction of a semitone; I'm not sure. A 1% error for a cassette deck is par for the course. And after a while these things, wow and flutter, become intolerable. The president of our chamber music society has reached the stage where he can hear wow and flutter on cassette decks; it's an acquired ability. Oboe is perhaps the best instrument to demonstrate that. He can even hear wow and flutter on *live* oboes. *(Laughter.)*
EARGLE: Aha. He's really good.
LIPSHITZ: It takes great skill. But for a perfectionist it's a flawed medium. For most people it is a very convenient, good-quality medium.
EARGLE: Yes. True. I agree. Okay, let me say this about it, since I'm the one who implied that it had no problems. The thing is that there are more cassettes sold—more prerecorded cassettes sold—than any other single recorded carrier of sound. So that the consciousness level out there must be geared into what the cassette is offering.

* * *

The question we have to ask ourselves is, if there were dissatisfaction with the cassette as a medium out there, then it would bid fair to be replaced by DAT or something. I don't think that level of dissatisfaction exists with the prerecorded cassette because if it did, it wouldn't be selling the way it is. Now that may be a circular argument; I don't know.
LIPSHITZ: But I agree with John that the future of DAT for the consumer is highly questionable because I don't see its clear function. It is recordable, and that is continually made great weather of, yet the LP survived for 40 years and it was not recordable. I believe the cassette is where it is now because there was no other feasible medium for use in cars, and certainly for portable use when the Walkman was invented. The CD may never be as impervious to vibration and so on as a DAT or cassette can be; I don't know. But it seems silly to have double inventory, one for your home and one for your car.

"They [cassettes] sound terrible—please! They're awful." "They're not flat; they change from day to day; after a few months they start wow-and-fluttering..."

EARGLE: Record companies hate it. They hate the promotion of extra inventory.
LIPSHITZ: So, really, it makes much more sense to me to have a CD player or changer in the trunk of your car and use the same discs at home, and to have some kind of portable CD player, than to have two sets of software. And that would probably not mean buying two sets of software but buying the one and transcribing to the other. Now is it worth really paying $20 or $15 for a two-hour blank DAT cassette to transcribe a $15 CD? And that requires a $2000 recorder as well, in order to enable you to do it, plus presumably a player, another $1000 DAT player in your car, so you can play it. It seems to me it's a technology which the professionals and semipros will be using. Many artists already, I understand, would request copies on DAT to take home to listen to, to approve or otherwise...
EARGLE: Most of the ones I know would take a cassette of that.
EDITOR: The hardware is still a gray-market product. Nobody has really authoritative information on this—whether that can be expected to change or...
(Remember, this seminar took place before the great mid-1989 compromise in Athens, Greece, between the recording industry and the DAT hardware makers. Some of what follows here is therefore out-of-date, but the opinions expressed are still of interest.—Ed.)
CLARK: I have some authoritative information regarding professional units. There are Technics units on the shelf—they're by Panasonic—that have XLR connectors; they are imported; they're for professional use. So it's here for the pro use, but anybody can buy them.
LIPSHITZ: But you have to buy through a pro dealer, don't you?
McGRATH: No. There are, in fact, two or three gray-market companies now that are seling them to anybody over the counter. Even home or pro units.
LIPSHITZ: In Canada they're available through Sony.
CARVER: The Carver dealers have pretty much killed it. They say no, it doesn't work right to the customer's interest. Of course they don't have them to sell.
LIPSHITZ: What doesn't work properly?
CARVER: DAT's.
LIPSHITZ: You have a DAT machine?
CARVER: Yes, I have a DAT machine.
LIPSHITZ: I didn't know you had a DAT machine.
CARVER: Yeah.
LIPSHITZ: Oh. Is it advertised?
CARVER: No. I don't make one. Carver Corporation doesn't. I have one.
LIPSHITZ: You want to be sued then by this conglomerate? Wait a minute, is he saying that he personally owns one?
EARGLE: Yes. He personally owns one machine.
CARVER: I personally only have one machine.
LIPSHITZ: Oh! I thought he meant Carver Corporation was offering one for sale.
CARVER: No, no, no.
LIPSHITZ: Ah! I thought, goodness, he's going to be sued. Then we'll see what happens.
McGRATH: But Nakamichi will have one out. They showed it at the show and they are intending to bring it in.
EDITOR: And they're willing to be sued.
McGRATH: We're going to take it on.
CARVER: The success of the DAT machine, at least now when it's expensive, depends a lot on how the dealers get behind it.
EDITOR: Well, more power to them. I think somebody should take on the RIAA and beat them. And I think they will if they take them on.
LIPSHITZ: Well, Sony in Canada has been selling R-DAT's for a year and a half now—the consumer *and* the professional R-DAT's. But only through Sony Canada Professional, and that means you've got to buy it through a professional audio dealer.
CLARK: But that means you have to walk in off the street, plunk down your money, and buy it—the same as retail.
LIPSHITZ: Probably. I don't know. I haven't tried it.
CLARK: My office is in with a pro audio retailer, and anybody with, well, $2600, $2700 can walk in and buy this professional Technics DAT. Actually a couple of dif-

ferent models—and the Sony. There is no problem.
LIPSHITZ: The thing that riles me about this whole DAT controversy—and I am sympathetic to the whole copyright question, and there's no doubt that a lot of these machines will be used for copying copyrighted material, although the U.S. Supreme Court has already ruled on the Betamax situation, and it's not clear that for personal use that is not acceptable—but what annoys me is the way this is treated as a new, unprecedented threat because it's a high-quality copy. They're implying that low-quality copies are all right.
EDITOR: Exactly. Not only that; they're implying that a very good copy is all right, but a very, very, *very* good copy—that's bad! That's the implication and that's nonsensical.
McGRATH: It's absurd.
LIPSHITZ: And on top of that, having the double lockout—that means the following two things: One, you can't do a digital-to-digital copy because the consumer R-DAT machines will not digitally copy at 44.1 kHz sampling rate. That's locked out two ways. There's a copy-prohibit flag in the R-DAT format, which would automatically prevent copying at *any* sampling rate if that flag is set. Virtually every CD made has in its digital subcode a copy-prohibit flag, so you couldn't copy a CD onto your R-DAT even if your R-DAT were to allow digital dubbing at 44.1 because the CD digital output says "copy protected," and your R-DAT will say, "Sorry, I won't dub it for you." But on top of that, they have prevented recording at 44.1 and decided it's got to be 48. Now that means, we have—the chamber music society that I'm part of has—250 or 300 Beta-format master digital tapes. We could not transfer them to R-DAT format.
EARGLE: I'm sure you can find a way to do it.
LIPSHITZ: Oh, we could find a way to do it. But that's not my point. It's a double lockout; it is pointless. That lockout was already present in the digital interface format. Why put an additional one in to appease an industry that weren't appeased? They weren't appeased—let's go over...
EDITOR: To the best of your knowledge, Stanley, how would the president of RIAA answer that one question?
LIPSHITZ: Oh, I've no idea. He probably couldn't really answer it without making some other sort of similar things...
McGRATH: Right there. *(He shows his Sony professional DAT recorder deck.)*
LIPSHITZ: But this is the professional...? This is yours?
McGRATH: Yes.
LIPSHITZ: We have in front of us the... Well, you better say what...
CLARK: Exhibit C.
McGRATH: No, it's legal. It's legal. It's a pro unit.
LIPSHITZ: It's perfectly legal. It's the Sony portable pro R-DAT machine, which will do these things.
CARVER: They're all legal.
LIPSHITZ: But, for example, not all the pro units will do what you think they do. Professional people have a need to transcribe material for legitimate purposes.
McGRATH: You are right.
LIPSHITZ: I mean some recording studios are being sent CD's by the copyright holders and told, "Here's the digital master. I'd like you please to rearrange the titles, the songs, in the following order." If they have a CD player with a digital output, they can't dump this down onto digital tape and rearrange the order because that CD has got a copy-protect flag on it, and they can't dub without doing various things...
McGRATH: By the way, this will not accept a 44.1 signal in the digital domain. It will record in 44.1; it has a digital input; it will take 48 and 44.056, but it will not accept 44.1.
LIPSHITZ: If it's like the Sony R-DAT, the 2500 I had experience with, it will take 44.1 if it does not have a copy-protect flag.
McGRATH: But I tried it from another R-DAT to this, where I've never recorded the copycode.
LIPSHITZ: Oh. And?
McGRATH: In fact, what I did was, I recorded a master on this, from my analog, and then before I mailed the master off I wanted to make a 44.1...
LIPSHITZ: Which digital input format are you using? Which interface format?
McGRATH: I think it's...

"...it isn't likely that a company is going to keep a 48 kHz master for making R-DAT's, and then transcode that down to 44.1 and edit it, and then make CD's."

LIPSHITZ: Sony-Philips?
Mc GRATH: ...the EBU.
LIPSHITZ: The EBU? The XLR thing?
McGRATH: Yeah. Exactly.
LIPSHITZ: Ah. Well, that's not the case with the other Sony professional R-DAT.
EDITOR: I've heard through various sources that there are some consumer-type DAT machines that will record at 44.1 with a very minor unauthorized modification.
McGRATH: There are dealers, in fact, that are advertising that service.
LIPSHITZ: I think you must be quite careful in what you decide to publish in your magazine about this whole issue. I think we wouldn't, perhaps *you* wouldn't, want to be seen to be advocating such a modification of equipment.
EDITOR: I'm not advocating; I'm just reporting what I've heard.
LIPSHITZ: That is true. One can indeed make modifications, depending on the internal architecture of various equipment, to get around some of these things.
EDITOR: That still doesn't get rid of the copy-protect flag on the CD itself.
LIPSHITZ: No, but in principle somebody could make a black box that takes the digital feed in, finds the copy-protect flag but modifies that bit, and puts the same digital feed out. I mean, one can make those things; it's not something that the average person can easily do. There are ways of defeating almost anything you want to defeat. My point is, the people for whom I see a clear use for the R-DAT are the pros and the semipros who do their own recording. If you're doing your own recording, it's a magnificent recording service. It's more convenient to carry around than a video recorder and a digital adapter, or an open-reel format digital recorder.
EDITOR: What about editing?
LIPSHITZ: It doesn't have editing capability at the moment. But for our use, the chamber music society's use, these are a lot of functions we don't need. We don't have multiple takes; we can't edit.
McGRATH: It's the same for all the broadcast tapes I do. It's not a requirement.
EDITOR: You have editing facilities on that machine?
McGRATH: Not at all.
LIPSHITZ: You can edit between tracks by just, you know, record pause, just overwriting. It's not glitch-free, but if it's silent when you do that, that's fine.
EDITOR: Or you could copy onto another machine, digitally.
LIPSHITZ: For the average consumer, I don't see the need to do the recording. You see, my point was, the cassette was successful because they needed a compact format for car or portable use. It needed to be recordable because there weren't prerecorded cassettes available initially. So people become used to a recordable format. If you give them a recording-type R-DAT, I can see it could conceivably be used for dubbing material, possibly for use in car or portable environments where people don't want to use CD's. If it did not record and was a playback-only medium, I really don't see it. It doesn't really have advantages over the CD; it has disadvantages in most respects with respect to the CD.
McGRATH: Really.
CLARK: It's better for the car.
LIPSHITZ: It might be cheaper to duplicate in the long run than the CD is, but I doubt it. The CD has had quite a few years lead time to get its process down.
EARGLE: Yes, the tape cost alone is not likely to go down very, very much...
EDITOR: There have been suggestions to the effect that the higher sampling rate produces a minor improvement.
LIPSHITZ: But you won't get the higher sampling rate because the material that would be released on prerecorded CD's will have been...
McGRATH: It's all 44.1.
EDITOR: But the prerecorded DAT, conceivably...
LIPSHITZ: But the 44.1 dubs that people get will come from the CD masters.
EDITOR: ...where, say, a Mitsubishi master is made at the 48 kHz sampling rate and then duplicated digitally on a DAT.
EARGLE: Peter, it isn't likely that a company is going to keep a 48 kHz master for making R-DAT's, and then transcode that down to 44.1 and edit it, and then make CD's.
EDITOR: dmp does that.

EARGLE: Well, dmp is a very special little company, and one man runs it; he can do anything he wants to do. But this will not become an industry standard because it would require two different edits.
LIPSHITZ: And not only that. You've got to watch it, though, because there are many professional digital recorders that are switchable 48/44.1. But it does *not* follow that if you switch it to record at 48, the analog filters are switched, so that your bandwidth increases. It may *still* use the 20 kHz filter it uses at 44.1.
EARGLE: In which case you gain nothing by doing it at 48.
EDITOR: All you're saying is that it's not designed as a total two-format machine.
LIPSHITZ: No—and quite honestly, if I'm given the following choice, even assuming the analog filters are switched, to record at 44.1 or record at 48, for CD release I would record at 44.1. There's no point in doing digital arithmetic unnecessarily when you're digitally filtered to get it down to the lower bandwidth. It's pointless.
EDITOR: That makes a lot of sense.
McGRATH: Back to dmp, though, what he does—he edits in the 48 mode and then number-crunches to the 44.1, so he doesn't have to edit in both domains.
EARGLE: Is he doing razor-blade editing?
McGRATH: Yeah, on his Mitsubishi. Yes.
LIPSHITZ: But there are the Mitsubishi 96-kHz sampling machines that are now available. What on earth are people going to do with those?
EDITOR: I think that happens to be a very good product, if you like the music—dmp I mean.
McGRATH: I think it's wonderful. I think he makes the best-sounding product I know of.
EDITOR: I suggested to Tom Jung that he dig up some of these rapidly fading jazz greats that hang out in various places in New York, half forgotten. They're not being replaced; there is no new generation of jazzmen who play like that, and they would deserve his kind of recording, but so far he is only doing this New Age stuff—or whatever they call it.
LIPSHITZ: Well, quite honestly, you know, with a portable R-DAT machine and a good mike/preamp thing—you could always carry it all in under your arm—you could record in clubs and produce pretty high-quality masters, I think, if you can get permission to do that.
EDITOR: Sure.
CLARK: Can I ask a question? Why do we even consider a flawed medium like the cassette—amongst ourselves, why do *you* —or LP for that matter, that has demonstrable flaws in it, when at the same time you're really questioning a console, with these series of amplifiers, that's demonstrably okay?
McGRATH: I'm not sure I follow.
EARGLE: Would you ask that again? I missed the line of the question.
CLARK: Well, we have something that specificationwise—and to me soundwise—is great, like digital audio tape...
EARGLE: Okay.
CLARK: ...and you were saying that the cassette is sort of okay, but this is...
EARGLE: Well, I would say, look—I ought to be ashamed, but I have never bought a cassette tape. I use it for making copies for those people who need them. Everybody that I know who wants a copy of something wants a goddam cassette.
McGRATH: That's right. And I don't own one either, and I live with them for exactly the same evil purpose.
EARGLE: Okay. And the only reason I put forth the cassette as a viable medium—not likely to be replaced by the expensive R-DAT, which is the only thing that would replace it in kind—is that there are more prerecorded cassettes sold than any other thing in this country. They must be doing something right—either the expectation of the masses out there is so low that they will accept a cassette with all of its garbage, or we're being a little bit too critical of it.
LIPSHITZ: Isn't the answer, John—are you not actually implying—that if these people go off their cassettes at some point, they'll probably go on to CD, not to R-DAT?
EARGLE: Well, I know they're not going to go to R-DAT because of the expense, the high cost of buying prerecorded material. You can pick up a prerecorded cassette

"...we have an unfortunate situation... The R-DAT was never intended as a professional digital format and may become a professional, not a consumer, format."

for three or four dollars.
CLARK: Yes, but that's like saying computers are expensive.
LIPSHITZ: But the R-DAT material would come down if it were mass-produced.
EARGLE: It's not going to come down. The tape cost will not come down. The duplicating cost will.
LIPSHITZ: But it's not clear to me why you couldn't produce a 75-minute R-DAT and sell it for the same or less than the price they sell CD's for. I'm sure you could.
CLARK: I didn't get that point I was trying to raise satisfied.

* * *

EARGLE: Dave, what is your question again?
CLARK: The question, which turns out not to be really much of a question after it's been brought up, is why do we accept the cassette? Because, well, we really don't accept the cassette. But I'd like to make a point because we're talking high end, state of the art in audio here, right? I would just like to make the point that the cassette does not represent the state of the art in audio, and it never did.
EARGLE: I think we all agree to that. And I think we all say that it's an inferior recorded medium that just so happens to have elevated itself to the highest figures in the country.
McGRATH: But the other half of your question was, how could someone criticize consoles...
EARGLE: ...rail against the console and accept cassettes.
McGRATH: And LP's are so bad, too—I mean, was that the way it went?
CLARK: Yeah, I was throwing that in. But can we just say, when cassettes are so bad?
EARGLE: Well, it's a matter of accepting. The thing is that the public doesn't accept consoles; only professionals accept them. The cassette exists because of the vast public acceptance, and we just have to acknowledge that. The question that originally got this going was Peter's question about mediums, about recorded media. And any question about recorded media must indicate the largest-selling one, even though it's terrible.
LIPSHITZ: I think we have an unfortunate situation with the DAT, in the same way that the cassette was unfortunate. It was never intended for music and became the largest music character that we know, in terms of numbers—of sales. The R-DAT was never intended as a professional digital format and may become a professional, not a consumer, format.
EARGLE: Well, that's the only area where it's been allowed to grow. It does it so well!
LIPSHITZ: It does not have the time code capability that one would want; at least it's not formalized in some standardized way yet...
EARGLE: You don't need it at that stage; you dump it to an editor.
LIPSHITZ: One could. But if you were designing a professional R-DAT, you probably wouldn't design it the way the R-DAT format is...
EARGLE: Well, it's an another funny screwup. I mean it's brilliant engineering, improperly brought to the market.
LIPSHITZ: ...so it's another case of this situation. It's a beautiful thing—I mean, I initially was very skeptical about its robustness, but apparently it's surprisingly robust. It's very nice. And it probably is going to be around for some time. I don't know whether it will become *the* portable professional format; I suspect it will because so much money has gone into the development of the technology, the standardizing of it and so on. I think the Japanese will tend to stick to it.
McGRATH: How old are your earliest F1 masters? When did you start using it?
LIPSHITZ: When did it come out? Was it '81?
McGRATH: Yes. I think I got one of the first in the United States...
LIPSHITZ: Probably '82.
EARGLE: '82 was when I got mine.
LIPSHITZ: I think '82.
EARGLE: Seven-year period.
McGRATH: Mine are imported from Japan. And the masters I made from them...
LIPSHITZ: No problem.
McGRATH: No problem. They still play. Absolutely no problem.
LIPSHITZ: I always used high-grade types. I had bad experiences with other types be-

cause of head clogging.
McGRATH: But everyone said that they're wonderful—*but*... And wait a few years or wait a couple of hundred plays... I made maybe a thousand masters.
LIPSHITZ: Oh yes, but Doug Sax also said that you couldn't make a single digital-to-digital copy without errors. I mean, people have said all sorts of things. I've reached the stage now where I don't care what people say—I care what people can prove.
EDITOR: Peter, I would like to ask you, before we quit this subject, to state the "tree-worshiping analog druid" party line.
EARGLE: The what, now...?
McGRATH: The tree-worshiping analog druid party line. Whatever that is.
LIPSHITZ: The initials spell TWAD.
McGRATH: On what issues?
EDITOR: You still consider the very best analog achievable to be better than the very best digital recording achievable? I believe that's it.
McGRATH: That's essentially my position.
EARGLE: Are you talking about finished product for the consumer or a master tape which you made?
McGRATH: I'm talking about having made recently, using the analog master tape as a starting point—which may or may not be a valid premise for some people—but taking that tape and taking the original machine that recorded it, using "brown rice" type wires, and dumping it straight into a Sony 1630 with no intervention, just finding where the peak level is and then making a transfer...
LIPSHITZ: Nothing wrong with that.
McGRATH: That's exactly what I've done.
LIPSHITZ: I mean, I've been buying a whole lot of these old historic releases...
McGRATH: No, no. Let me finish the whole thing.
EARGLE: He's not through yet.
McGRATH: I'm building a case. And then we've taken that tape and dumped it into a 1630. The 1630, then—all that's done to it at that point is just time code is put on it, and the CD...
EARGLE: It's making it from analog to digital at the same time. It's a very important step.
LIPSHITZ: A-to-D conversion, certainly.
McGRATH: No, no. What I'm saying is, after that transfer we just put the time code on it. Then CD's are made. Then that same original master is taken to Doug Sax and it is cut. And then, comparing...
CARVER: Cut? An LP?
LIPSHITZ: From the CD? From the digital master?
McGRATH: No, no. Cutting from the original analog master—we are then making an LP. And then comparing the finished LP product to the CD product, back to the original analog master tape.
LIPSHITZ: Played on the same machine every time?
McGRATH: Played on the same machine throughout.
LIPSHITZ: Even at Doug Sax's?
McGRATH: No, it was cut on a different tape recorder there. I sent something like 12 different reference tones and so forth, so I'm pretty sure that the EQ was right—and I'm not exactly sure what he does there.
LIPSHITZ: Was he able, by the way, to cut the disc with a nonpolarity?
CARVER: I have a lot to say about this stuff. *(Muttered under his breath.)*
McGRATH: We tried both, in terms of a comparison...
EARGLE: Well, what are your objections to this? *(Addressed possibly to Lipshitz or Carver; not clear.—Ed.)*
McGRATH: ...and he inverts.
LIPSHITZ: He inverts?
McGRATH: The bottom line is, when we played the LP and compared it to the compact disc using a—you name it—an Accuphase with a Wadia D/A converter, or an Accuphase straight, or any one of a variety of different CD players, and using the LP on one of my better turntables, such as a Goldmund, the LP more closely mirrors what's coming off the analog tape than does the CD. And based on that, I would say I still would have to consider myself an analog druid tree worshiper or whatever.
LIPSHITZ: What do you conclude from that experiment, then?
McGRATH: Well, what I conclude is that if the digital technology cannot indisputably mirror even the deficit-laden analog original, then far be it from doing so with live original music.

"What I have done is that I've compared this [professional DAT] making duplicates of analog masters to the original analog master, and you can still hear a difference."

EARGLE: Okay, can I ask you another question? Have you taken a 1630 and performed the following experiment? Because the 1630 is the only element in the chain that could really be in question—or it's certainly one element that could be in question—the other element would be the CD player, the Wadia and the Accuphase. Have you ever jumpered from the composite digital in and out on the back of the...
LIPSHITZ: Is the Wadia a player?
McGARTH: I'm sorry. This is a Wadia D/A converter.
LIPSHITZ: Yes, I know of it.
EARGLE: Okay. But the point is that if you take the digital encoding part of this whole thing and run its output into its input and use it as a link in a comparator and A/B through it or around it, you think you're going to hear much of a difference?
McGRATH: I don't know.
EARGLE: Okay. Have you ever taken your F1 unit and A/B-ed it like that?
LIPSHITZ: Yes. And I've challenged Ivor Tiefenbrun to that experiment, and he could not tell the difference between A and B.
EDITOR: Yes. That was written up in the Boston Audio Society newsletter.
EARGLE: I can't either. And I'm sure you couldn't tell the difference there.
CARVER: I sympathize, Peter, I sympathize a lot with your experience...
LIPSHITZ: I have an explanation for what you're saying, a potential explanation for your conundrum, and that is—as John has mentioned, there are two possible places where the change could have occurred. Because when you transcribed your analog tape to digital, you used your analog tape recorder...
McGRATH: Sure, there is a possibility...
LIPSHITZ: ...when you listened to it for the comparison subsequently, you used your analog tape recorder.
McGRATH: Correct.
LIPSHITZ: Forget the LP; we'll try to compare the digital copy with the analog original—and you're saying they sound different.
McGRATH: They do.
LIPSHITZ: So the problem is either the 1630 or the CD player, as far as I'm concerned. Now I can say this about the Wadia CD player *[he means D/A processor—Ed.]*—it is not flat in frequency response.
McGRATH: Well, the differences are also audible when just coming straight out of the DP-70, which is presumably a pretty good CD player. That's the Accuphase. And, if anything, the sound gets closer to the subjective impression of what's on the analog tape when going through the Wadia.
LIPSHITZ: But forget that. Why not play it on your 1630 and see if *it* sounds like the analog tape? Because that eliminates the one extra step.
McGRATH: What I have done—for the sake of argument, what I have done is that I've compared this *(pointing to his Sony professional DAT deck)* making duplicates of analog masters to the original analog master, and you can still hear a difference. And this at either 44.1...
LIPSHITZ: And how did you set your levels and so on, Peter?
McGRATH: What you do is, you simply take a 1 kHz tone, which I have on the analog tape, and run it into here; then, when running this back, we match the two levels precisely—because I have a trim on the output of the Stellavox...
LIPSHITZ: That's measured at the loudspeaker terminals?
McGRATH: Yes, exactly. And you hear a difference. Now, I'm not sure if this phase-inverts—there's no documentation...
LIPSHITZ: I'm sure it doesn't.
McGRATH: I don't know that it does or doesn't.
LIPSHITZ: I doubt it.
McGRATH: I hear a difference. I mean, it's there. I mean, in terms of what I've... Maybe, again, I've not done an ABX...
EARGLE: Okay, if you walked into the room, gone out of the room, had somebody toss a coin, and...
McGRATH: Yes, yes. Because what happens, I come in and I sit down and I say, "Scramble!" I don't want to know what I'm listening to. And I can pick it every time. It's very obvious to me.
CARVER: I sympathize with that. I've had similar experiences—but from the up side of the coin. My experience has been like yours when I listen to the LP and listen to the CD—the LP is much nicer to listen to.

But I've not been able to have your experience, in which the analog copy compared to the digital copy is superior. I've always found that they're indistinguishable, whenever I've done experiments such as you've described. So my results are different than yours, and I can't explain them. All I know is—*both* experiences are anecdotal. But what I did do is a series of converging experiments, and it actually was written up. When my CD didn't sound as good as my LP, I set about to make them sound the same. And it wasn't very difficult; it wasn't very hard at all. I took a very pedestrian approach. I didn't have the kind of equipment that you had; I just went out and bought a bunch records—LP's and a bunch of CD's—and tried to make them sound the same. And I found that I could, if I simply modeled into my signal chain the specific frequency response of my cartridge/LP interaction—and also the fact that the vertical information coming off the LP was significantly larger than the *equivalent* vertical information coming off the CD. And so I had to make a little circuit that...
McGRATH: Introduced some phase shift?
CARVER: No. It didn't introduce phase shift.
LIPSHITZ: Increased the L – R.
CARVER: Increased the L – R. And also, I had to do some other funny things. I had to equalize the L + R channel a little differently than the L – R channel. That introduced a little time delay. Then I had to make a time-delay compensator—sort of the same thing we talked about earlier—before it went back into the matrix. By the time I was finished, I had a circuit board that had a lot of stuff on it. I was looking at it—shit, it takes all that...
LIPSHITZ: To degrade the CD player to what your cartridges do when you play an LP—to what cartridges and cutters do to the original signal.
CARVER: Yeah. The thing of it is, I genuinely enjoyed the LP better. The things that I like about my hi-fi set—which is sort of a warm bass, and a nice curtain of sound, lots of bloom—my cartridge did all of that. And it didn't take long to figure out that it was my cartridge/LP interaction that produced the soundstage that I liked.
EARGLE: I hope you're not putting forth the idea that that's more accurate.
CARVER: Heavens, no!
EARGLE: Okay.
EDITOR: He's saying he likes inaccurate sound.
CARVER: I'm saying that I liked it better, but it's not difficult. I mean, in the next breath—you know, unless I sit on my hands and bite my tongue—it would be very easy to say, you know, it must be more accurate, it sounds more lifelike. Because it does sound more lifelike. I mean it has depth—and more depth. But it can't be more accurate; obviously it isn't.
McGRATH: Well, generally, when I get that kind of customer coming into my store—saying that CD's sound like garbage, and this and that and the other—I generally take a very hard-line posture and just simply ask, on what basis are you saying that? Here I'm just simply offering what I have experienced. That's why I said, about making these comparisons...
CARVER: The point of what I was saying is that the LP is not superior to the CD. I can make the CD sound like the LP; I could never make the LP sound like the CD. So, by definition, that means that as a recording medium...
McGRATH: Well, I'm still not convinced that *that* isn't true *[viz. that the LP is superior to the CD—Ed.]*. I still feel that an LP, meticulously manufactured, using what hopefully is a state-of-the-art lacquer, state-of-the-art cutting techniques, and then played back on a state-of-the-art turntable...
CARVER: Oh, yeah...
(Everybody talking at the same time.)
EARGLE: It's hard to get all of the state-of-the-arts to happen at the same time.
LIPSHITZ: You see, it's rather like Mitch Cotter telling me...
CARVER: It is possible. I agree; it's possible. It's amazing how good an LP can be. I mean, it can be virtually indistinguishable from a CD.
LIPSHITZ: So can an analog tape.
CARVER: Sure.
McGRATH: Well, again, all I'm saying is that the LP produced from that resulted in sound more closely mirroring what was on the analog tape. That to me makes it better.

"The point of what I was saying is that the LP is not superior to the CD. I can make the CD sound like the LP; I could never make the LP sound like the CD."

CLARK: I agree than an LP is better than a cassette.
LIPSHITZ: Well, I don't know the answer to the difference you heard. We're second-guessing; we can't answer that question. Sounds like a perfectly reasonable thing you did, and if you indeed heard the difference, I can't explain it now. However, I would...
EDITOR: There is a possible explanation.
LIPSHITZ: ...say—my belief—there is an explanation; I don't believe it's esoteric, and I'm sure that if...
CARVER: If we were there, we would sort it out.
LIPSHITZ: ...we could repeat that experiment, we'd be able to find the reason. And whether it relates to frequency response or a polarity thing, a distortion thing, an overload, I don't know what it conceivably could be, but it's one of them.
CARVER: But we would find it, and it wouldn't be mysterious.
McGRATH: Well, I'm not suggesting that it is.
CARVER: And it wouldn't mean that the LP storage medium is somehow more magnificent or superior. That's what it is...
EDITOR: Here's a possible explanation. Your digital recorder is more bandwidth-limited than your pure analog system.
CLARK: That's true. Maybe it's...
LIPSHITZ: But so are his ears, I think. How high do you hear?
McGRATH: What?
LIPSHITZ: How high do you hear?
McGRATH: What? *(Everybody laughs at the joke except Lipshitz.)*
CLARK: Almost up to voice range.
EARGLE: How old are you?
LIPSHITZ: Very good. I didn't guess that you were being silly.
McGRATH: I have no idea. I would guess 10 kHz, 14 kHz.
LIPSHITZ: Yeah, but I mean you're not claiming more than 20 kHz, like Laurie Fincham—21 kHz he used to claim—I don't know if he still does.
McGRATH: I'm not claiming anything at all. No, no. I've no idea what my ears do. I'm 40 years of age, and I suspect that it's normal...
LIPSHITZ: I'm below 15,000. I wouldn't be able to listen to the TV whistles...
EDITOR: Okay. One could still suggest that there's a lot of out-of-band energy in the analog program material, which, when properly reproduced, may not be directly audible but may somehow interact with the audible spectrum.
EARGLE: No! You'll have to explain that one. That's a long article.
McGRATH: Come on, Mitchell!
LIPSHITZ: A nonlinearity. Okay, I will give you an example that happened three, four months ago of exactly that. A friend of mine at the university comes to me, and he says, "I borrowed a CD from a friend. I tried to transcribe it onto cassette tape—because I liked it. Can't do it," he says. The CD is perfectly fine, he says. The cassette tape has got this horrible distortion on it. So I started asking various questions. Yes, even recording at lower level, it's still like that. Problem. So I asked him to bring me the CD and the cassette copy. I listen to the CD; it's a pop recording with a female vocalist, and so on...
McGRATH: There's a 12 Hz signal in there or something...
LIPSHITZ: No. No. Perfectly fine-sounding but clearly gimmicked recording—the cassette tape has got this ghastly, gritty, horrible, twittery sound on the voice. Things going on. Ha! Sounds like an intermodulation thing—something beating with the bias? Because analog recording is not very linear, certainly not with cassettes. So we do a spectrum analysis of the CD—oh, goodness gracious, look what's happening above 5, 10 kHz—I forget exactly where the corner was going up. So a student of mine says, "Oh," he says, "looks like they might have used an Aphex Aural Exciter on that." Now you know. You listen to the voice—it's got this excessive breathiness to it. I guess this is Aural Excitation. There is a *huge* high-frequency thing there; cassette decks cannot record that without overload and beating.
EARGLE: Well, if you bring the level down, it ought to disappear.
LIPSHITZ: It did not.
CARVER: By 20 dB. You have to bring it down 20 dB.
LIPSHITZ: I think his cassette deck was bad. I took it home. I've got my modified

Revox B77. I've been doing digital recordings for the last seven years or whatever, but before that, you know, I've got dozens of master tapes, and I would challenge many people to tell the difference between the analog and the digital, except on the very loud passages. When you set the volumes within tenths of a dB—and you have to be very accurate on that—they can be *very* close. But no problem recording it on my machine, so I think it's just his cassette deck, and/or the cassette medium couldn't handle that. But there's an example where there is no obvious explanation for the horrible distortion. And it wasn't out-of-band stuff because the CD produces no out-of-band signals, nothing above 20-odd kHz.
EDITOR: You said something, Stanley, that I would like you to explain. It doesn't really belong in the discussion of storage media, but it came up and we might as well handle it. You said that our ears are even more bandwidth-limited than the digital system and therefore the bandwidth of the digital system is perfectly adequate. I have heard suggestions to the effect that the ear can detect rise times corresponding to higher frequencies than we're able to hear as pure tones. Do you...?
LIPSHITZ: I know many hypotheses. I know many people who say, *may* be able, or *could*. These are all possible. I just happen to believe they're not true. I mean...
EARGLE: Well, nobody has ever proven them at least.
CLARK: Some of those experiments—I've heard some of that stuff. What it turned out to be when it got into the fine print—somebody says this in an offhand manner, and I followed it up—they're talking about presenting two things almost simultaneously, and when could you detect a shift, and the amount of change required in the two presentations.
LIPSHITZ: At your two ears, you say?
CLARK: And the time, when translated to a frequency, was above the audio band.
EARGLE: Oh, yes. Well, the ear, you know—in terms of microsecond-resolving of a very sloppy onset of the signal—you know, it will shift...
LIPSHITZ: Interaural is one thing. But monaural or intra-aural...
EARGLE: If you do what you're saying and translate that into an equivalent bandwidth...
CLARK: It gets very high.
EARGLE: ...it gets very, very high.
CLARK: Like 40–50 kHz or something.
EARGLE: It's a remarkable property of the ears acting in concert with each other and improving on the bandwidth.
McGRATH: Couldn't there be some relation to that?
EDITOR: Couldn't there be that type of energy in actual program material?
CLARK: No, we're not talking energy; we're talking about a difference with which low-frequency energy is presented to the ears, and then the very mathematical, artificial thing of saying, "What's the time difference?"—and then looking at that time difference, the reciprocal of it, as a frequency. That's the only explanantion; the one thing that came up that I understood with these things. That's the best I could do as a translation.
CARVER: You know—and 50 microseconds is a long interaural time space...
LIPSHITZ: Oh, no, no, no. You can hear a couple of microseconds. What you can hear is down in the less than 10 microsecond range.
CARVER: That's what I said—50 microseconds is a *long* time. It would be very easy...

"If you present a signal to both ears with a small time offset between the two, that time... can be less than the rise time of the ear, and you can still detect the timing difference."

LIPSHITZ: 50 is very big. But now 20 kHz bandwidth would correspond to—what's your .35 over... You know, what is the standard first-order rise time of a 20 kHz bandwidth signal? I forget the number offhand, but it's...it's...
(A somewhat tentative calculational discussion ensues between Clark, Lipshitz, and Carver. They end up getting it close, but no cigar; the correct answer is 18 μs.)
CARVER: ...It's 20 microseconds, and 20 microseconds is a big time for your brain to...
EARGLE: A long time, yes.
CARVER: I mean, your brain will be able to pick up a small fraction of 20 microseconds, so...
LIPSHITZ: So the point, I think, that's being made is that if you present a signal at one of our ears, the mechanical system that lets it into the inner ear has a frequency response cutoff—an incredible plummeting low-pass filter that chops off, if your hearing is still good, at maybe 20 kHz—and the inner-ear transduction mechanism converts all this into neurosignals that go off to the brain, and the rise time of the system based on that frequency response cutoff is not very fast. However, if you present the signal to *both* ears with a small time offset between the two, that time offset can be less than the rise time of the ear, and you can still detect the timing difference.
EARGLE: The firing takes place at different times.
LIPSHITZ: Because the signals are cut or delayed going into both ears the same amount, but the one was additionally offset by a small amount towards the other. No contradiction there.
EDITOR: But are you suggesting that this kind of phenomenon cannot be reflected in actual program material?
CARVER: No, no, no. It can.
CLARK: It could be.
CARVER: It can, easily.
EARGLE: If you could derive a binaural experiment to do this...
CARVER: It's a binaural... It's not a test of how we hear rise times or high frequencies; it's a test of how we hear binaurally. That's what it is.
CLARK: And you could do it with a 20 kHz band-limited system. You could run this experiment.
LIPSHITZ: Yes, exactly. That's a good point.
EDITOR: I see. Okay.
(Unfortunately we must stop in medias res, *having run out of space. To be continued, and concluded, in Issue No. 15.)*

Letters to the Editor

(continued from page 9)

than it actually is.

You remind us of a series of old jokes that took the form of "I'll never forget what's-her-name" and "Support mental health or I'll kill you." Your own protestations on the subject, in your own letter, are the proof that you are bratty, self-indulgent, undisciplined, etc. Any brat with an IQ of 90 can learn the words "intellectually and emotionally bankrupt" and then throw them like stones at anyone he gets mad at, regardless of the context. (Were you a spitter and biter in your fourth-grade fights?) As for your imputation of financial bankruptcy, it is more than bratty; one has to be mindlessly irresponsible to put something like that in writing without having seen a financial statement specifying assets and liabilities. Where do you get your facts?

One would expect a Cornell graduate (1) to be aware that "provocative" statements tend to provoke people and (2) not to react to the inevitably ensuing criticism as if it were an outrage crying to heaven. If you call digital recording "a sick joke" and its endorsement by the press "out-and-out lies," then you have be prepared to take a little heat yourself. You overstate your case so grotesquely that serious people will not take you seriously. You come off as an attention-seeking lightweight.

Note that we are not entering the analog vs. digital debate here all over again. It occupies a good many pages of the seminar transcript in this issue; we have little to add at this juncture. We do want to point out, however, that we never said that The Absolute Sound *had turned down your article, only that they would not have published it even if you had submitted it. Also, the missing part of your old subscription has meanwhile been fulfilled. You just had to ask, like several thousand others; we had advertised the fulfillment offer for many months.*

—Ed.

The Carver "Silver Seven" Clones: Who Needs Tubes for the Tube Sound?

Whether the shamans and vested interests of the High End like it or not, these solid-state Carver "t-mods" accurately (and without magic) replicate the sound of the $17,500/pair "Silver Seven" tube amp.

The subject of Bob Carver's controversial t-mods, and of his amplifier design philosophy in general, is far too involved to be rehashed and reargued every time we write a new Carver amp review. We refer you to Issue No. 10, pp. 32-44, and Issue No. 11, pp. 28-29 (the premiere review of the Silver Seven), where all the necessary background information is presented without the knee-jerk anti-Carverism of the ultrahigh-end establishment.

Carver M-4.0t

Carver Corporation, P.O. Box 1237, Lynnwood, WA 98046. M-4.0t Magnetic Field Power Amplifier, $799.00. Tested sample on loan from manufacturer.

This review should have appeared in the last issue, where we ended up running only a capsule summary on the M-4.0t for lack of space. (See Issue No. 13, p. 56.) Meanwhile Carver's clock has been ticking faster than ours, and the M-4.0t is about to be phased out in favor of the virtually identical but cosmetically slicker TFM-42 and TFM-45. No matter; the M-4.0t is the original Silver Seven t-mod, and all qualitative statements about it are true of the entire family it engendered.

As we explained in our original Silver Seven review, the whole point of that somewhat grotesque though brilliantly successful exercise in vacuum-tube cultism was to provide a model for a widely marketable solid-state clone like the M-4.0t. According to Bob Carver, the earliest Silver Seven (the one we had reviewed) and the final prototype of the M-4.0t produced a null deeper than –60 dB when bridged together; since then, however, the Silver Seven has undergone certain minor modifications, whereas the M-4.0t has remained unchanged, so that the null today is almost surely not as deep. That, of course, does not make the M-4.0t a less "good" amplifier, since the null by itself is not a figure of merit; the important thing is that the Silver Seven, in both its earliest and current versions, and the M-4.0t are different executions of the same conceptual power-amp architecture and sound like fraternal, if not identical, twins.

Being among the very few who have heard the early *and* the current Silver Seven, as well as the M-4.0t and the latest-and-greatest Silver Seven-t (see below), we can unhesitatingly report that they all sound essentially like one and the same amplifier, although we had no opportunity to run a round-robin of ABX double-blind listening tests, which might conceivably have revealed differences in sonic freckles and eyelashes. Those who understand the principle of the Carver t-mod (as *Stereophile* appararently does not or would rather not) realize that there is no trick or magic to the creation of such soundalikes; the surprising thing would be if they sounded different because after the t-mod there is no physical mechanism whereby a difference could occur.

That said, we must state that we disagree with Bob Carver's choice of the Silver Seven's transfer function as the model for his new generation of amplifiers. The Silver Seven probably—indeed, almost certainly—represents the ultimate *tube* sound, but the ultimate tube sound is not the ultimate *sound* (i.e., absence of amplifier sound), at least not in our opinion. Our theoretical block-diagram ideal is a block with near-infinite input impedance, near-zero output impedance, and wideband gain with near-zero distortion. The transfer function of the Silver Seven, or of any other audiophile tube amplifier, differs from that ideal in several respects. The output impedance is typically well over an ohm; there is a light sprinkling of benign second-harmonic distortion (say, –55 to –60 dB) throughout the audio spectrum; and the distortion at the extremes of the spectrum tends to rise somewhat abruptly with level. The higher-impedance drive can be expected to result in frequency-response ripples at the amplifier/speaker interface, some of them in excess of 1 dB; the tweeter range will tend to be slightly rolled off; the Q will rise at the woofer resonance; the entire sonic presentation will have a marginally softer, rounder, warmer quality than with near-zero source impedance. We feel that this kind of sonic formatting should take place at the discretion of the record producer, whenever he considers it desirable; however, those who like to have it permanently built into their power amplifier are entitled to their preferences—*chacun à son goût*, as Prince Orlofsky says. We, too, are seduced by it from time to time.

Once you accept its very slight leaning toward the vacuum-tube personality—still far short of the C-J type of sound, for example—the M-4.0t is a super amplifier, and what makes it so is primarily the Carver magnetic-field power supply. We never cease to marvel at this design, one of the very few genuine inventions in amplifiers, which multiplies voltage/current output per cubic inch—and per dollar—by a factor of 4 or 5 or better, in comparison with conventional power supplies. In this admittedly extreme case, the Silver Seven is 22 times the price and 13 times the weight of the M-4.0t, yet the latter is very nearly as powerful, especially when the load is in the normal range of 4 to 8 ohms. (The output transformer of the Silver Seven gives it a tremendous advantage, of course, in the range of 1 to 2 ohms.) So much clean power in a small, lightweight, low-cost package will make anyone question the value of whatever alternative choices the marketplace offers. We must also point out the complete absence in the M-4.0t of the slight commutator noise that used to be a marginal flaw of earlier versions of the power supply. The amplifier is dead quiet, mechanically as well as through the speakers.

On the test bench, the M-4.0t exceeded by a considerable margin its basic 375/375-watt specification. Into 8 ohms, with both channels driven, clipping occurs in the upper 400's at most frequencies except the very highest. At 1 kHz you can nudge 500 watts per channel. With only one channel driven, these numbers go way up. Into 4 ohms, with both channels driven, the numbers go up but do not double; the power supply has its limits (and most houses have 20-ampere circuit breakers). The small-signal bandwidth of the amplifier is 0.78 Hz to 81 kHz (–3 dB points); noise measures –110 dB relative to 375 watts into 8 ohms (shorted inputs, A-weighted)—right on specs. The output impedance is 1.2 ohms, probably the most revealing number of them all.

We found absolutely nothing to fault in the sound of the M-4.0t, as long as we judged it as a tube amplifier. We heard nothing unpleasant or unmusical, ever, regardless of the speaker load; we never ran out of power; in fact, there was never even a minor problem of any kind. With a few speakers, such as the first-generation Carver "Amazing" (not the one reviewed in this issue), the amplifier's small deviations from utter neutrality actually proved to be synergistic. The Carver speaker had a wee bit more snap and sparkle with the Carver amplifier; obviously one zigged where the other zagged. Bottom line, though—what other clean, musical, almost 500/500-watt stereo amplifier is available out there for $799? (All right, for $839, which is the price of the new and no different TFM-42.)

Carver "Silver Seven-t"

Carver Corporation, P.O. Box 1237, Lynnwood, WA 98046. Silver Seven-t Magnetic Field Power Amplifier (monophonic), $1000.00 ($2000.00 the pair). Tested samples on loan from manufacturer.

There are two ways to look at this new departure in Carver electronics. One is to say that it is basically nothing more than a double M-4.0t. It uses the same circuit boards, slightly modified to fit; it has the same power supply, only double strength; it has the same output configuration, but with twice the number of output transistors per side. It could be argued that a mono-bridged M-4.0t is the same amplifier, except for the upward shift in the impedance-matching characteristics of the bridged configuration. All that is perfectly true—and a good thing to remember should some exquisitely subjective know-nothing assert in print that the two amplifiers sound totally different. But the other way to look at the Silver Seven-t is to recognize that it is Carver's first all-out bid for the high-end religionist's dollar and as such far more threatening to certain vested interests.

The difference is in the packaging. The M-4.0t looks like any other Carver amplifier. The Silver Seven-t echoes the sexy tube look of the big Silver Seven, with an old-fashioned round analog meter dominating the distinctive slanty-blocky mono chassis. The meter measures output level in dB (0 dB = 575 watts) and has four different scales for those who like to play Mr. Spock. The metalwork of the chassis is very nice; the $2000 seduction of the high-end customer is very convincingly orchestrated. We could name half a dozen very high-end amplifier manufacturers who may not be sleeping well as a result.

The Silver Seven-t is the most powerful amplifier we have ever reviewed. It delivers even more clean power into the load than the early Silver Seven, although Bob Carver claims that the present version of the latter, which we have not measured, puts out a few more volts than the t-mod but not more current. The Silver Seven-t is capable of over 50 amperes peak current into low-impedance reactive loads. To mess with such voltage and current levels for more than a few seconds without a specially equipped lab bench is a bit hairy, so we must be somewhat general in reporting measurement figures. Clipping level into 8 ohms is in the vicinity of 600 watts at most frequencies (well over that at 1 kHz); into 4 ohms we are talking about the 900-watt neighborhood (over 1000 watts at 1 kHz); into 2 ohms there is still an increase in power, with readings in excess of 1100 watts at all but the highest frequencies; into 1 ohm the clipping level drops back to the paltry 600's. With a clean signal just short of clipping, a load of 0.61 ohm will not trip the current limiter switch but 0.57 ohm will. Whew! THD at levels approaching clipping into either 8 or 4 ohms is in the 0.1% to 0.15% bracket at 1 kHz, rising to 0.5% or so at the frequency extremes, a very typical tube-amp profile (mostly 2nd harmonic, of course). Gain is 29.0 dB; signal-to-noise is comparable to that of the M-4.0t, as is the small-signal bandwidth of 0.8 Hz to 80 kHz (–3 dB points). The output impedance is a tubey 1.1 ohms.

The surprising thing is that none of the muscle of the Silver Seven-t is wasted when playing orchestral or organ CD's through big speakers of ordinary efficiency. We were quite disarmed by the happy musicality that goes with such immense reserves of clean power, tube-like character or no. What a nice sound—not even a small premonition of strain, ever!—despite the not quite tightly damped bass and somewhat polite highs. Go ahead, Bob, be a tube freak. ◊

Philips Gives Some Special Fillips to Its High-End Audio *and* Video Line

Does an "ultimate" CD player need to cost $4000? And do you need an ultimate CD player when you have an ultrasophisticated laser-disc video player? Philips has some answers, and we have some questions.

Let there be no doubt about it—Philips has the ability to do anything it wants to do. When a Philips product is engineered and packaged in a certain way, that is exactly the way the company wanted it. There is enough talent on the gigantic Philips payroll to bring you a Wadia or a Theta or a Krell if that sort of thing were the targeted end result. (In fact, the CD itself would probably not exist today if it had not been for the vision and stubbornness of Jan D. Timmer, boss of the Philips consumer electronics complex, who originally fought for the new technology against strong opposition and now, needless to say, walks on water in Eindhoven.)

We are pointing this out because we are receiving confusing signals from Philips regarding their current involvement in the ultrahigh-end digital market. They have obviously decided to go after the Wadia/Theta/Krell type of customer to top off their bread-and-butter Magnavox and mid-audiophile Philips-brand sales, yet they also seem to have decided that the 1987 level of technology will be sufficient for the foreseeable future, with 16-bit 4-times oversampling D/A conversion and all the rest. Even their new single-bit "Bit Stream" conversion technology appears to be aimed at some sort of technical parity with the older system (at lower cost) rather than a distinct improvement. We, on the other hand, after some initial exposure to the latest 18-bit 8-times oversampling architectures as well as to the Japanese MASH single-bit conversion system—not to mention discrete analog output circuitry—are beginning to feel that Philips is becoming perhaps a little too complacent. We are not saying that we have already heard clearly better-sounding CD playback than the best Philips has to offer, but it is our impression that they are not ahead of the game at this point and may fall behind at any moment, if they have not done so already. The ball is in their court.

An in-depth article on CD playback technology will be published in our next issue (No. 15), along with a good many CD player reviews; you can expect more conclusive statements on the subject there, after we have consolidated our test results. Comparisons of the various DAC architectures, old and new, will be one of the features of the article.

As for the high-end video scene, we refer you to Issue No. 12, page 10, for a summary of our philosophy. The laser-disc player review here is only the beginning.

Philips LHH1000

Philips Consumer Electronics Company, One Philips Drive, P.O. Box 14810, Knoxville, TN 37914-1810. LHH1000 two-chassis CD playback system (LHH1001 compact disc transport and LHH1002 digital-to-analog converter), with LHH1003/RC universal remote control, $4000.00. Tested samples on loan from manufacturer.

The question that arises almost as soon as the LHH1000 system is unpacked and hooked up is—what have we got here that we did not have with the combination of the Philips CD960 and DAC960, at less than half the price? The answer is—heavy-duty, professional-quality construction, obviously "industrial-strength" in all details; a "smart" remote control that can learn the codes of other remote controls; and, of course, the psychological intangibles that a lavishly packaged, cosmetically distinctive, "limited-edition" audio component represents to a certain type of buyer. Performance? Outstanding but no better than that of the CD960 and DAC960, or even of the CD960 alone. Subjective sound quality? Just as good—and that means very, very good—but no better.

The circuit boards are gorgeous, but the DAC is still the same TDA1541A S1 "Golden Crown" chip as in the CD960, DAC960, and even the CD880—all of them use the top-grade Philips chip. The analog output circuitry of the LHH1002 converter unit is still not discrete but uses quite inexpensive integrated op amps. In fact, we discerned no special, ultrahigh-end engineering mentality in the electronics of the LHH1000 system (not that we believe that such a mentality is meaningful in all cases). Basically, we are dealing with the Cadillac syndrome here—the working parts are standard Chevrolet/Oldsmobile/Pontiac/Buick, but there are extra little refinements, and the package is more luxurious.

Philips claims that the LHH1000 realizes "a startling 15.75 bits of resolution from the 16 bits available in the compact disc format." We measured 15.5 bits of resolution in the better channel and 15.3 bits in the other. We are convinced, however, that the claim is sincere because in the similarly configured DAC960 (see Issue No. 12, p. 35), we did measure exactly 15.75 bits in the *better* channel. So 15.75 seems to be the best-case scenario, not the production average. This is one area where the latest 18-bit architectures *might* offer a tiny improvement. Otherwise we found the output of the LHH1000 to be very clean, although not totally free from minuscule anomalies. The LHH1002 converter unit appeared to have a few millivolts of erratically fluctuating DC offset coming out of it—no big deal. The twin-tone (11 kHz + 12 kHz) intermodulation distortion test showed a very low-level 36 kHz product, something we cannot remember seeing elsewhere but again of no particular significance. All in all, the laboratory bench paints a very satisfactory picture but nothing to take our breath away.

In terms of sound quality, we discern no difference between the Philips LHH1000 and their lower-echelon CD players using the S1-grade DAC chip (CD880, CD960, DAC960). Unfortunately, we did not have for comparison a good player with a Precision Audio D1 analog board in it, which might have sounded ever so slightly sweeter, if indeed our original perception of it was accurate (see Issue No. 12, pp. 35-36). We still believe that the top-of-the-line Philips CD sound has not been *clearly* surpassed by any other manufacturer's product, an opinion that could change very rapidly because said Philips sound is certainly not *superior* to the best offered by a number of others—and we have yet to test some of the top contenders.

Of course, when it comes to visual and tactile satisfaction, the LHH1000 is vastly superior to all the excellent-sounding sheet-metal-and-plastic boxes that are the great boon of our digital age. A die-cast aluminum alloy chassis, a computer-like LCD readout on your remote control, or even just a push button with a really solid feel can make a life-style statement. So, if $4000 is small change to you, go ahead and buy the LHH1000. You will not be sorry. There is no reason to eat your heart out, however, if you decide against the expenditure.

Philips CDV488

Philips Consumer Electronics Company, One Philips Drive, P.O. Box 14810, Knoxville, TN 37914-1810. CDV488 combination CD/videodisc player, with RC488CDV universal remote control, $1300.00. Tested sample on loan from manufacturer.

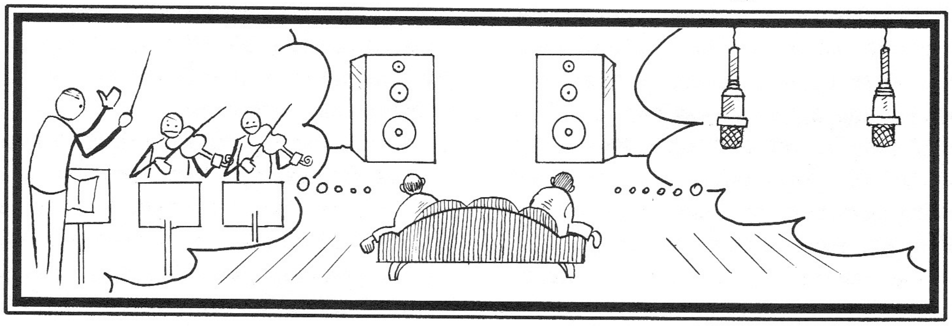
As we stated in our first review of a high-end video product (Issue No. 12, pp. 10-11), we are convinced that the marriage of high-quality audio and video is the wave of the future in home entertainment systems and that the future has already begun. This remarkable new Philips component represents just such a marriage, this time on one chassis, and we are rather excited about it. We are aware, of course, that laser videodiscs are not exactly the prime medium for movie rentals and such, but they are clearly the best prerecorded video medium as far as color fidelity and sharpness of image are concerned, and with a digital sound track they are the dream medium for opera in the home. Furthermore, as we shall see, the videodisc player part of the CDV488 is practically a free gift to the purchaser in view of the unit's capabilities purely as a CD player.

The fact is that the CD playback performance characteristics of the CDV488 are virtually identical to those of the three times costlier LHH1000. We say virtually because some of our measurements of the CDV488 resulted in exactly the same figures and others in ever so slightly *better* figures. For example, the resolution in *both* channels was 15.5 bits, not just in the better channel, and the output was totally clean on all tests, without even the tiniest anomalies. No die-cast aluminum alloy chassis, to be sure; no proliferation of power transformers; no separation of digital and analog all the way to the AC wall outlet; but the DAC is the same S1-grade Golden Crown chip; and the remote control is even more elaborate, LCD window and all, because it also has to incorporate all the videodisc functions. As for the sound of the CDV488, we found it indistinguishable from that of the other four upper-echelon Philips models, although a round-robin of ABX comparisons (which we had no intention to endure in this particular case) could conceivably have revealed minidifferences of no musical consequence, except perhaps to the self-flagellating tweako contingent. Overall, we think the CDV488 represents at least a $750 value as a CD player, so you are getting a state-of-the-art videodisc player for $550—not bad at all.

That videodisc player will, for openers, play optical discs of any size and format in existence, without adapters. The tray has the right-sized dimples to accommodate 3", 5", 8", or 12" discs, and the player automatically adapts to CAV (constant angular velocity) or CLV (constant linear velocity). There is an S-video output that feeds separated luminance and chrominance signals from the player's proprietary high-resolution processor into the S-video input of your TV receiver or monitor—if it has one. This helps to keep colors purer and reduces interference artifacts. The claimed 440 lines of horizontal resolution raised our eyebrows a bit; our video testing facilities are not quite ready yet, but others (Ed Foster, Len Feldman) have measured the video bandwidth of the CDV488 and have not reported that good a figure. Even so, the horizontal resolution is better than that of any VCR, including the S-VHS decks; we have never seen its equal. Philips has a demo disc in their 12" CD Video format, presenting a movie called *Flyers*, which was originally filmed in IMAX, a super-resolution system with a film frame 10 times the size of the standard 35-mm movie frame. This disc, played on the CDV488, gave us a dual thrill: the best color picture we have ever seen on our screen, with stupendous definition and color fidelity (not to mention fantastic digital sound) and stunt-flying sequences that we are still not quite willing to believe. As for special digital effects, the CDV488 offers perfect freeze-frame and slow-motion even on extended-play (CLV) laser discs; the jog/shuttle control on the remote unit allows frame-by-frame viewing as well as 1/2-speed to 10-times-speed forward or reverse. Strobe and mosaic effects are also available.

So—are you about to buy a CD-only player? Think. ◊

Records&Recording



There have been discussions with volunteers as well as mercenaries who would like to relieve your Editor of the chore of writing this column, but no one so far has gained his unqualified confidence. There is a basic difference between critical reviewing and personal opining that seems to elude a lot of would-be critics. The right talent will undoubtedly be found sooner or later; meanwhile you will just have to put up with the old one-man band.

All Kinds of Good Stuff: Catching Up on Our Backlog of Demo-Quality CD's

By Peter Aczel
Editor and Publisher

This is a temporary switch from vertical to horizontal coverage; so many CD's worth mentioning have come my way since the last go-around that I would rather discuss more than the usual number of them briefly than to focus on one subject or just a few selected items in depth.

Bainbridge

The company behind this audiophile label is Mobile Fidelity Productions of Nevada, and they seem to have some kind of tie-in with By The Numbers, another Nevada company that makes the Colossus digital recording system. The Bainbridge catalog is small, but the quality is high.

"Rhapsody in Gold & Blue." Works by Bach, Gershwin, Rogers, Bernstein, et al., arranged for percussion ensemble. Percussion 90, West Virginia University. Bainbridge Records BCD2104 (made in 1989).

The trouble with most of the demo-quality percussion recordings (such as, for example, *The Sheffield Drum Record*) is that it is boring to listen to more than three or four minutes of purely percussive instrumental sounds without melody or harmony, even if the rhythmic interest is considerable. Percussion 90 is a virtuoso ensemble that happily includes melodic/percussive instruments like the xylophone, marimba, celeste, etc., in addition to the usual drums and assorted bangables, so that an hour's worth of their percussing on this CD will probably hold your attention even if you are not a witch doctor. How about a short contrapunctus from *The Art of the Fugue* played on a dozen and a half percussion instruments? After all, J. S. left it in open score, an invitation to "realizations." I love it; they play it so convincingly. The recording is state-of-the-art, fully documenting the alleged strengths of the Colossus digital processor and of the MS coincident microphone technique. The spatial and directional clues are extraordinarily clear. All in all, hog heaven for the percussionado, not to mention the imaging/soundstaging freak.

CBS

One mainstream label that occasionally comes up with better-than-mainstream sound. What will happen now, after the Sony takeover, remains to be seen.

Samuel Barber: Concerto for Cello & Orchestra, Op. 22. Benjamin Britten: Symphony for Cello & Orchestra, Op. 68. Baltimore

Symphony Orchestra, David Zinman, conductor; Yo-Yo Ma, cello. CBS Records MK 44900 (made in 1989).

Both of these mid-20th-century works are of surefire appeal to those who are not necessarily on good terms with modern composers. Barber, the neo-Romantic, and Britten, the postmodern eclectic, are highly accessible and technically brilliant in all of their music, but in these large-scale opuses they rise to permanent repertory stature. I find the Barber especially beautiful, but the Britten is also riveting in a cooler vein. Yo-Yo Ma plays the cello parts with the utmost virtuosity and command of the idiom; as for the Baltimore Symphony Orchestra, they definitely sound like a major-league outfit here. The recording is a good example of the Schoeps omni sound, a little more closely miked than most, resulting in a drier and tighter quality than some may like, but in this kind of music I want analytical clarity, and that is exactly what I get. The massed strings sound a little aggressive now and then, but not so different from their front-row sound in the concert hall. I am sure the recording would have a much less straightforward, more heavily processed, committee-designed sound if this were more of a big-bucks glamour production.

Delos

So far I have reviewed only the big sound—symphony orchestra and organ—on this outstanding label, but what they call their "intimate sound" is of equally praiseworthy quality and shares the spotlight here.

Johannes Brahms: String Quintet No. 2 in G Major, Op. 111; Quintet for Clarinet and Strings in B Minor, Op. 115. Chmber Music Northwest: David Shifrin, clarinet; Ani and Ida Kafavian, violins; Walter Trampler and Steven Tenenbom, violas; Fred Sherry, cello. Delos DE 3066 (made in 1989).

The late B. H. Haggin, whose articles and books were among my earliest and strongest influences in matters of music criticism, bluntly listed these two quintets among the "Bad Works" of Brahms. ("The Best Works" were the 4th symphony, and the Haydn and Paganini variations.) He thought Brahms' chamber music was pretentious, labored, and often saccharine. Now that I am my own strongest influence, I respectfully disagree, at least as far as Op. 111 and Op. 115 are concerned. This is enchantingly beautiful music in the autumnal mood of the composer's later years. The idiom is inimitable, and the magic endures. The instrumentalists here are largely world-class; their performances are as good as any I am aware of; and the one and only John Eargle provides recorded sound that is about as vivid, natural, spatially palpable, and just plain gorgeous as the present state of the art permits.

Howard Hanson: Symphony No. 1 in E Minor ("Nordic"); Elegy in Memory of Serge Koussevitsky; Symphony No. 2 ("Romantic"). Seattle Symphony Orchestra, Gerard Schwarz, conductor. Delos D/CD 3073 (made in 1989).

To my knowledge, this is the only available recording of these works on CD, and certainly the only DDD version, so the panoramic John Eargle sound is merely a bonus here because one listenable performance of this music belongs in every serious music lover's collection. I find it very puzzling, indeed frustrating, that every little musical hiccup of Respighi, for example, is performed and recorded over and over again, whereas the only 17 years younger Hanson's masterpiece, the "Romantic" Symphony—no less accessible than the "Pines of Rome," with glorious melodies and harmonies, masterly symphonic structure, and brilliant orchestration—languishes in relative obscurity. The other pieces on this disc are also far from negligible, and Schwarz is good at this sort of thing. Highly—nay, enthusiastically—recommended.

David Popper (1843-1913): Romantic Cello Favorites. Janos Starker, cello; Shigeo Neriki, piano. Delos DE 3065 (made in 1989).

If you think it was Fritz Kreisler who invented the short, schmaltzy, hummable, virtuoso encore piece for string soloists, I have news for you. David Popper, the greatest cellist of his time and a prolific composer for his instrument, had already perfected the genre a full generation earlier. Janos Starker, possibly the greatest cellist of *our* time, plays 20 of Popper's superb bonbons on this CD, each of them more delightful than the one before. Yes, they are a little corny to the late-20th-century ear but disarmingly and endearingly so. Starker plays them as if nothing in the cello literature gave him greater pleasure; I have never heard better cello playing—and some of these pieces are monstrously difficult to play. The man's artistry is a perfect blend of scholarly musicianship and virtuosity. As for John Eargle's recording, the cello is very up-front and vivid but always sweet-sounding; the piano also sounds excellent but is mostly in the background, strictly as an accompaniment. I have played this disc so many times that, if it were an LP, it would already be full of ticks and pops. Delos has a world-class product in this instance

Robert Schumann: Overture, Scherzo and Finale, Op. 52; Konzertstück for Four Horns and Orchestra, Op. 86; Symphony No. 1 in B-flat Major, Op. 38 ("Spring"). Seattle Symphony Orchestra, Gerard Schwarz, conductor. Delos DE 3084 (made in 1989).

There are those who believe that Schumann's orchestration is thick, turgid, and muddy. And there are those who believe that Gerard Schwarz ain't no Toscanini. This CD will be an ear-opener to adherents of either belief. The "Spring" Symphony is as transparent as Mozart's "Jupiter" if the conductor sticks to Schumann's original, unaltered scoring and to the left-right deployment of first and second violins that Schumann took for granted. As for Maestro Schwarz, he is nothing short of inspired when he is proselytizing in behalf of Schumann as a symphonist of the first rank. My own prejudice about Schumann is more along the lines of the cruel quip that he "began as a genius and ended as a talent." I like the early works. The earliest work on this disc is the symphony and it is by far the most beautiful. The other two are merely interesting, except for some of the horn passages in the *Konzertstück*, which are quite dazzling. The recording is standard-issue Eargle/Seattle, the qualities of which need no further reiteration in this column.

Dmitri Shostakovich: Symphony No. 11 in G Minor, Op. 103 ("The Year 1905"). The Helsinki Philharmonic, James DePreist, conductor. Delos D/CD 3080 (made in 1988).

John Eargle's symphonic recordings remind me of my sainted Hungarian uncle's mot: "Women are like whiskeys—they're all good but some are better." This is one of John's better ones. The hall in Finland is smaller than his usual Seattle venue; both the orchestra and the conductor are very different but very good; the Colossus digital processor, which he does not always use, was in the recording loop—it is not the standard Eargle setup but it works out just right. The sound is only moderately live but far from dry; the orchestral choirs are very precisely located and have great presence. I would put the 11th on the junior varsity of the Shostakovich symphonies—it is basically anti-czarist program music with a movie-sound-track flavor—but DePreist conducts it with tremendous conviction, as if it were great music, and while it lasts I almost believe him. Good demo material, with lots of brass, drums, ostinatos in the low strings, etc.

Anton Arensky: Trio in D Minor, Op. 32. Peter Ilyich Tchaikovsky: Trio in A Minor, Op. 50. Andres Cardenes, violin; Jeffrey Solow, cello; Mona Golabek, piano. Delos DE 3056 (made in 1989).

This is another example of superb chamber music sound by John Eargle, recorded in a totally different hall and with different musicians than the Brahms but just as lovely tonally. The music is perhaps not as great but deeply affecting if you have the slightest feeling for the Russian Romantics, and it is performed here with a great deal of verve and emotional involvement by excellent instrumentalists. This is what stereo in the home is all about—the violin is on the left, the piano in the middle, the cello on the right, and you, the listener, are sitting in front of them in the same room. It is all very plausible, very close to the live experience. We have entered the golden age of audio.

Dorian

This new label has had a great start, but so far they are almost exclusively into keyboard (piano, harpsichord, organ) and voice (solo and choir) recordings. In those categories, Craig Dory makes the most beautiful-sounding digital recordings known to me, bar none. I can hardly wait for his first symphonic release, which at this point is only a gleam in his eye (and in the eye of A-and-R partner Brian Levine). There is, however, a luscious Dvorak chamber music CD in the pipeline, already excerpted on *Dorian Sampler Vol. II* (DOR-90002).

"Wachet Auf!" J. S. Bach: Cantata No. 56 (Ich will den Kreuzstab gerne tragen); Cantata No. 140 (Wachet auf, ruft uns die Stimme); Motet BWV Anh. 159 (Ich lasse dich nicht). The Bach Choir of Bethlehem & The Bach Festival Orchestra, Greg Funfgeld, conductor; Henriette Schellenberg, soprano; David Gordon, tenor; Daniel Lichti, bass. Dorian DOR-90127 (made in 1989).

For the sake of brevity, I refer you to my enthusiastic review of Dorian's first Bach cantata CD (Issue No. 12, pp. 45-46). This sequel is identical in every respect—performers (except for the soprano soloist), performance style, recording venue, recording team, sound quality—but of course the program is different. These cantatas are perhaps more popular, and No. 140 includes one of Bach's instantly recognized themes, the sublime and almost mesmerizing chorale *Zion hört die Wächter singen*, which also recurs as the first of the six Schübler chorales for organ. If you are comfortable with the Bethlehem style of performing Bach, grab this one.

Julius Reubke (1834-1858): Sonata for Organ in C Minor ("The 94th Psalm"); Sonata for Piano in B-flat Minor. Jean Guillou, at the Aeolian Skinner organ of Trinity Church, New York City, and on the American Steinway D concert grand. Dorian DOR-90106 (made in 1989 from masters recorded in 1987).

The doomed young genius who died in his early or middle twenties was almost a stock figure of the German-speaking world of the 19th century, or so it seems. The playwright Georg Büchner, the poet Moritz von Strachwitz, the composer Julius Reubke, and the philosopher Otto Weininger come to mind as quick examples. Reubke was a pupil of Franz Liszt and an admirer of Wagner, although the latter's finest works had not yet been composed when Reubke died. This CD represents the sum total of Reubke's surviving compositions and is a monument not only to the composer but also to Jean Guillou as one of the most astonishing keyboard artists of our time. The man turns out to be as great a virtuoso and interpreter on the piano as he is on the organ. (See my comments in Issue No. 13, pp. 53-54.) Both the organ and the piano sonata are grandiose, bombastic, declamatory, and then some, in the best Liszt/Wagner tradition, perhaps criticizable as the work of an overwrought young man, but I find them quite magnificent—possibly because Guillou performs them magnificently. The recorded sound is fabulous; the piano was recorded in the Troy Savings Bank Music Hall, a national treasure in acoustics, and is absolutely thrilling in dynamics and tonal quality; the Tinity Church organ sounds better than in real life from the floor of the church. Esoteric music that is not boring—and an audio blockbuster to boot—is not issued on CD every day; Dorian deserves the highest praise.

Robert Schumann: Carnaval, Op. 9; Kinderszenen, Op. 15; Drei Phantasiestücke, Op. 111; Gesänge der Frühe, Op. 133. Antonin Kubalek, piano. Dorian DOR-90116 (made in 1989).

This CD is remarkable just for offering 78 minutes (!) of superbly recorded piano music, but there is more to it than that. *Carnaval* and *Kinderszenen* are early Schumann and thus from his "genius" period (see above); only the best of Chopin is better in that particular vein. The later pieces are almost pathetically uninteresting by comparison, at least in my admittedly unscholarly opinion. Nor does Kubalek captivate me as a pianist, although he does rise above the general stodginess of his playing in a few of the more inspiring passages. On the other hand, Craig Dory's recording of the Hamburg Steinway D in the Troy Savings Bank Music Hall is absolutely state-of-the-art. This is the way a good piano sounds in a good hall when you are sitting up front. Play the best tracks only and groove over the sound.

"The English Lute Song." Julianne Baird, soprano; Ronn McFarlane, lute. Dorian DOR-90109 (made in 1988).

"Greensleeves: a Collection of English Lute Songs." Julianne Baird, soprano; Ronn McFarlane, lutes. Dorian DOR-90126 (made in 1989).

I was sorely remiss when I failed to review the first of these releases in the last issue; it had been out for a while. The second one with the same artists is quite recent. The two of them represent what I consider to be the most accurate recording of the soprano voice available at the present time. The newer recording may be even finer than the first by a gnat's eyelash—Craig Dory thinks so—but the first was good enough for me. It became my acid test for midrange reproduction in all of my listening evaluations. The recording of the voice is completely open, undistorted, uncolored, spatially accurate, and in perfect balance with the marvelous acoustics of the Troy Savings Bank Music Hall. The lute also sounds absolutely gorgeous. Ten seconds of this program material can tell me at least half of the things I need to know about a piece of equipment. (I have to smile when I recall the era of midrange testing with Amanda McBroom on Sheffield.) Julianne Baird's voice is not a big one but it has an enchantingly pure and lovely quality, and her singing is just about flawless both technically and musically—as a farfetched analogy, she is not unlike a female Aksel Schiøtz. The Elizabethan stuff she specializes in is not exactly what I like to hum in the shower, but it is historically important and often very beautiful music, which I prefer to listen to a few songs at a time. There is a total of 52 songs on these two CD's, enough to last me awhile.

"The Great Organ of Saint-Eustache, Paris: Inaugural Recording." Works by Bach, Mozart, Liszt, Widor, Guillou, and de Grigny. Jean Guillou, organ. Dorian DOR-90134 (made in 1989).

Have no fear; it is not my intention to rhapsodize about Jean Guillou's organ playing all over again, but here we have the new organ he helped to design for his own church, the one I anticipatingly referred to in Issue No. 13, and I must report that it is very impressive indeed, maybe even more so than the Zürich Tonhalle organ. There is something about authentic stone-church acoustics that makes the organ experience more complete than in any concert hall, and in this case the reverberant characteristics of Saint-Eustache do not even marginally obscure the subtleties of voice leading or tone colors. The new organ, of which this is the first recording ever, has 101 stops operating 8000 pipes in 147 ranks on 5 manuals—it is gigantic, but it is equally capable of the early-18th-century baroque sound and the thunder of late-19th-century French extravaganzas, not to mention Guillou's rather bizarre contemporary sonorities in his own compositions. As usual, Craig Dory's recording captures the big picture as well as the small nuances with total accuracy, making this CD the hottest current item for the organ fancier.

Harmonia Mundi

Robina G. Young, the producer for this label, and my friend Peter McGrath, their recording engineer, still have very strong analog/phono loyalties and have not yet gone DDD. I have a feeling, though, that reality will soon prevail over ideology even in their case—hey, look at Gorbachev.

George Frideric Handel: Water Musick. Philharmonia Baroque Orchestra, Nicholas McGegan, conductor. Harmonia Mundi HMU 907010 (made in 1988).

Harmonia Mundi has not sent me any recordings for ages, but this one is mentioned in the seminar transcript in this very issue—where Peter McGrath says that he is "very, very happy with the sound"—so I thought I would play it through "his" speakers, the Quad ESL-63 USA Monitors, and comment on it. This is an AAD, so Peter would probably tell me to listen to the LP version if I really wanted to know how it sounds, but I like the CD well enough. The sound is in the minimalist Schoeps omni vein, recorded rather close up, with a second- or third-row perspective. There is relatively little hall sound; the space is smallish (a school chapel); the strings have great presence but are quite unlovely in tone, not because of the recording but as a result of the performance style—vigorously "authentic," with fierce attacks and exaggerated inflections, emphatic double-dotting, etc. It is all very real, very lifelike, but not very beautiful. The woodwinds, too, leave a great deal to be desired in tone and in phrasing; the horns are better and have a nice resonance from the back of the orchestra. I wish someone had played for this group Jean Guillou's organ transcription of the "Hornpipe" (from "Organ Encores" on Dorian) to show them how a great musician phrases the same music. Peter McGrath's recording, however, proves that this is at least one very good way of doing this sort of thing; in fact, I hardly noticed the two A's in his AAD.

Reference Recordings

I was hoping to review RR's promised Ravel album by Nojima, but so far it has not been forthcoming. Anyway, it will probably deserve a longer review in a less crowded column. Meanwhile:

Kurt Weill: Threepenny Opera Suite. Edgard Varèse: Octandre. Paul Bowles: Music for a Farce. Bohuslav Martinu: La Revue de Cuisine. Chicago Pro Musica ensemble. Reference Recordings RR-29CD (made in 1989).

The Chicago Pro Musica group consists of members of the Chicago Symphony Orchestra, and they are superb. Nobody plays better than this; every one of them is a major-league virtuoso. The music they play here covers the period between the two World Wars, all of it highly accessible and mostly delightful. The Weill suite takes up about 40% of the whole program, and I have a stylistic quibble about it. The Chicagoans play it too well, too precisely. The style of *Die Dreigroschenoper* requires a sleazy manner of delivery, that of a bunch of beer-guzzling 1920's German nightclub musicians who *think* they know how to play ragtime or a tango ("Mack the Knife" was originally a sort of diabolical tango). Playing every note perfectly, in strict time, exactly as written, takes some of the tawdry authenticity out of the music. The other pieces fit in more readily with the symphonic caliber of playing. The recording by Keith O. Johnson is sensationally good; the instruments are miked quite close up, with a very precise "bite" but no harshness, and their localization in space is perfect. The sound of the trumpet, especially, is incredibly real. On top of it, the KOJ microphone hiss appears to be gone. Who could ask for more?

Telarc

Of all the labels known for consistently top-notch sound, this one still has by far the greatest depth in artists and repertory. You could say that Telarc has brought the audiophile boutique sound into the mainstream.

Hector Berlioz: La Marseillaise & Other Berlioz Favorites. Baltimore Symphony Orchestra & Chorus, David Zinman, conductor; Sylvia McNair, soprano; Richard Leech, tenor. Telarc CD-80164 (made in 1988).

There are other recordings of the ususal excerpts from *Roméo et Juliette, La damnation de Faust* , *Les Troyens,* etc., that I prefer, although these are by and large very respectable performances by a very respectable orchestra. The reason for owning this Berlioz collection is the rarely heard performance of the composer's 1830 arrangement of all six verses of "La Marseillaise" for soloists, double chorus, and full orchestra—a nine-minute blowup of what is surely the world's best national anthem (as all *Casablanca* fans know). It is an experience to which even Francophobes should expose themselves, at least once. *Formidable!* The recording is in the classic Jack Renner tradition: Schoeps omnis, 12th-row perspective, gorgeous tonal texture, not much in the way of spatial/directional information, outstanding dynamics, stupendous bass drum. Pour yourself a Pernod and check it out.

Benjamin Britten: War Requiem, Op. 66. Atalanta Symphony Orchestra & Chorus, Robert Shaw, conductor; Lorna Haywood, soprano; Anthony Rolfe Johnson, tenor; Benjamin Luxon, baritone; Atlanta Boy Choir. Telarc CD-80157/2CD (made in 1989).

No other audiophile-oriented label, with the possible exception of Denon, has works on this gigantic scale in its catalog, so it is hard to make sweeping comparisons. This is certainly an achievement in the same league with Telarc's 1985 Berlioz Requiem (also Shaw/Atlanta), made with the old but far from inferior Soundstream digital technology. The Britten recording sounds even better, of course; it is one of the first made with the dbx/CTI 18-bit analog-to-digital recording processor. In fact, it represents a whole new ball game for Jack Renner, who abandoned his usual Schoeps omnis here for a combination of Sennheiser omnis and cardioids to achieve a soundstage of tremendous width and depth, along with great clarity of all instrumental and vocal parts of this complex score. I cannot imagine a better workout for a great stereo system; this is transparent, low-distortion sound with unlimited dynamics, the way audiophiles have always wanted it. I have mixed feelings about the music itself; some of it is fascinating and of great originality, such as the instrumental/choral passages of the *Dies Irae*; the rather stark solo settings of Wilfred Owen antiwar poems, with which the Latin text is interspersed, I find somewhat boring on the other hand. (Obviously these poems meant more to Britten than to me; for the generation that produced Pound, Eliot, and Cummings, they strike me as a bit provincial.) The performance by the Shaw/Atlanta forces is very fine, insofar as my limited familiarity with the work permits me to judge.

Aaron Copland: Third Symphony; Music for the Theatre. Atlanta Symphony Orchestra, Yoel Levi, conductor. Telarc CD-80201 (made in 1989).

This is the latest product I know of from the "new" Jack Renner, the one with the dbx/CTI digital processor and the Sennheiser microphones (see above). In this particular recording I find the violins to be a bit on the bright side for my taste; otherwise the sound is more "panoramic" (*à la* Delos), with more crisply defined inner detail, than what I am used to from Telarc; the dynamic range is spectacular, as usual. Do I prefer their "new, improved" flavor to the old? I am not sure. It is a step in the direction of greater clarity, more information, but the edges need to be rounded off a little, at least in this kind of music. The Copland works are highly enjoyable in this performance; to me Copland is always exciting and fun but not very involving or moving.◊

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