VOL. 10 NO. 6

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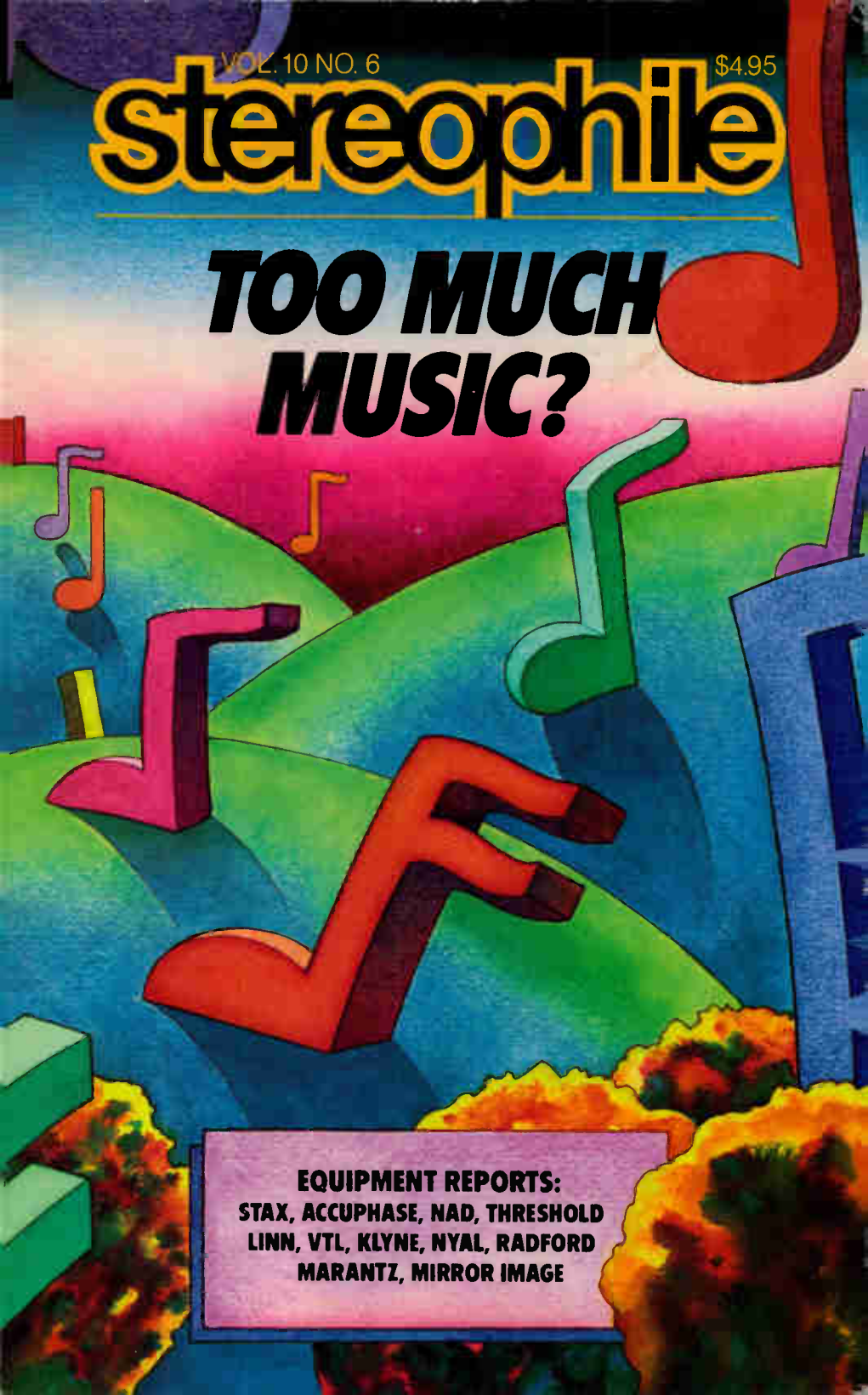

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Departures, Arrivals

It is my onerous duty to announce that Anthony H. Cordesman, one of our more senior, and prolific, writers, has resigned from the staff of *Stereophile*. He will no doubt continue with his writing at *Audio*, but more than that we do not know.

There have been two other departures from the staff in recent months: Allen Edelstein and Laurence Greenhill. Although both were valued contributors, the amount they wrote for us had dropped to the point where we felt it would be difficult for readers to evaluate their reviews in a proper context. Laurence Greenhill continues as a contributor to *Audio*.

Into every life, however, a little light must shine. Continuing as contributors—though to date without proper introductions—are Ken Kessler and Martin Colloms (like Christopher Breunig, who celebrates a year with *Stereophile* with this issue, both are former colleagues of International Editor John Atkinson at *HFN/RR*). And now added to our masthead is our first Musician in Residence—Lewis Lipnick. Kessler, who has been featured in *Stereophile* doing budget cartridges, is one of the UK's foremost authorities on popular music, particularly from the '50s and '60s. Martin Colloms has been reviewing components for *Hi-Fi Choice*, *Hi-Fi For Pleasure*, and *HFN/RR*, over the past 15 years; in addition, he evaluates products for numerous UK manufacturers, and even designs products for a few. Lew Lipnick, the principal contrabassoonist for the National Symphony in Washington, DC, has done music seminars for conrad johnson design, and appeared in some of their ads. He has now severed his professional relationship with that company so as to be able to review products for *Stereophile* without bias. Of course, he has written on live sound, concert halls, and CES in past issues of *Stereophile*; one of those articles garnered mention in *The New Yorker*.

Farewell to those leaving, but a hearty welcome to our new writers; long may you prosper in our midst.

Recording Rules for Orchestras

J. Gordon Holt

During my recent interview with the Sheffield Lab people in connection with their Moscow recording sessions (Vol.10 No.3), both Lincoln Mayorga and Doug Sax had some unkind things to say about the cost of recording an orchestra in the US. Their complaints are justified. It costs more to record in the US than anywhere else in the world, and these astronomical costs are detrimental both to symphonic music in the US and to the audiophile's pursuit of sonic perfection.

Orchestral music is the justification for high-end audio—its *raison d'etre*. The sound of a real, live orchestra letting loose is one of the most glorious sounds known to civilized man, and the hope of bringing that sound into one's home, as intact as possible, was what gave impetus to the high-fidelity movement from its very inception. Although it cannot be denied that pop and rock music sound better on good audio systems than on indifferent ones, merely "sounding better" is not what high-end audio is all about. We're talking about fidelity, which means faithfulness, which in turn implies an original sound to be faithful to. And anyone who knows anything about pop recording knows that fidelity is simply not a consideration for the typical 32-track studio.

The concept of fidelity has meaning only in connection with musical sounds that can exist without electronic assistance, which means the sound of unamplified, unprocessed, acoustical

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(mechanical) instruments. But as soon as one starts to apply standards of fidelity to sound reproduction, it becomes much more challenging; the goals are much more difficult to achieve. And the difficulty increases in direct relation to the number and variety of instruments one tries to reproduce. While it is possible with today's technology to reproduce a classical guitar with passable accuracy, the real sound of a symphony orchestra can barely even be *approximated*. There is too wide a variety of distinctive timbres to be convincingly portrayed by any one loudspeaker system, all of which, despite more than a century of refinement, remain as colored as the instruments they are trying to reproduce. Nonetheless, even the *im*perfect reproduction of a large orchestra in one's listening room can be almost as exciting as the sound of the real thing. An orchestra is the ultimate challenge for an audio system, as well as for a recording engineer.

Good orchestral recordings are hard to find because—as is well known to anyone who has tried—good places to hang microphones are also hard to find. In any given auditorium, there are usually at least two locations from which microphones will capture a decent recording, but the only way to find them is by trial and error. If the orchestra happens to be really worth recording—that is, if it is a professional group whose players are music-union people—you are prohibited from conducting any such trials unless you shell out at least $13,455 for the privilege. That's what it costs in America to record a 65-piece orchestra, whether for three hours or three minutes, and whether or not you have any intention of releasing the recording for public sale. It is also the reason why Sheffield Lab, and increasing numbers of other US record companies, do their symphonic recording overseas while American orchestras struggle to make ends meet. Why don't they just cut their prices for recording? They can't. It's against union rules.

How did this sorry state of affairs come to pass? Well, like most such institutionalized idiocies, it started modestly and reasonably many years ago. In 1942, live music in the US seemed in the process of being phased out. All over the land, radio stations, dance halls, and restaurants which had once relied on musicians were playing records instead. Unemployment among professional musicians was in the thousands and escalating, when an angry young man named James Caesar Petrillo resolved to put a stop to it. He proposed levying from the record companies (Columbia, Decca, and RCA Victor) a royalty on every record sold, for the support of unemployed musicians. The record firms were unimpressed, until Petrillo pulled out every musician in the US on a total recording ban which lasted two years. One by one, the record companies agreed to the royalty payments—of between ¼ cent and 5 cents per record—and the American Federation of Musicians earned a reputation as a union to be feared. In fact, after that, it was unstoppable. Every recording contract thereafter, and every revised set of work rules, was more draconian than the last and, as recordings came into ever-widening use, more inspired by paranoia.

Here, in essence, is what those rules have to say about recording in 1987:

- For a recording session, all musicians must be paid at least a standard recording rate of $69 per hour, regardless of their usual pay for performances.
- A recording session must consist of a minimum of three hours of time or a maximum of four. Each half hour of overtime must be paid for at a rate of $52 per half hour.
- If a recording session involves fewer than 65 players (but not less than 25), those members of the orchestra who are not needed must nonetheless be paid the standard minimum hourly recording rate for at least the first two hours of the session. Their presence is not required.
- For each half-hour of recording, no more than 15 minutes of material may be used for final release. This time can be averaged over two sessions.
- A symphonic work can be recorded during a live concert performance. The orchestra shall be notified in advance of which work will be recorded, and no other work on the program can be recorded. Payment for a three-hour session must be guaranteed as of the start of the first recording take.

There is no limit to the number of performances of that work which can be recorded at subsequent concerts, but when the recording is released, payment above the guaranteed amount shall be predicated on the basis of one

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session hour for each 10 minutes of released material.

- All of the above shall be in effect if there is a microphone present in the hall, whether or not a recording is made, and whether or not any such recording is intended for commercial release.
- Any exceptions to the above must be negotiated with the orchestra members and approved by *unanimous* vote.

On the surface, it looks like a good deal for musicians. But is it really? Let's look at some of its implications.

The hourly pay is about three times what European musicians get paid for recording. Perhaps a virtuoso musician is worth it in the US, but the fact is that most professional orchestra players are not virtuosi; they are only very competent players. The virtuosos tend to gravitate to the major orchestras, which pay the most and earn their major status by the virtuosity of their players.[1]

But we're talking here about a *uniform* recording payment rate. That means you will pay just as much to record 100 very competent musicians in Savannah or Memphis or Portland as you will to record 100 virtuoso musicians in Philadelphia or Boston or Minneapolis. Either way, that's $21,000 for a minimum session, from which you can use 45 minutes of material. Where, then, do you think a record company is going to record? Certainly not in Savannah or Memphis or Portland. The uniform-scale provision of the American Federation of Musicians, which is intended to reward all musicians equally for recording, ends up rewarding only those in a few top orchestras! The rest just don't get to make records.

In fact, it often costs *more* per minute of usable material to record a second-string orchestra, because virtuoso players are much more likely to get things right the first time through, whereas a second-stringer may require three or more takes to get certain passages down with the degree of perfection which everyone takes for granted from recordings. Rough edges which can be overlooked during an ephemeral live performance cannot be tolerated in a recording, where they will be heard over and over again with each playing. This is another reason why a record company will usually opt for the major orchestra, assuming it can find one that isn't already under contract to another record company.

So, regardless of what orchestra is recorded, the cost will work out to around $6900 per hour of playing for a 100-piece ensemble. And the three-hour session minimum multiplies that cost by a factor of three. That's a staggering $21,000 for 45 minutes of music, which works out to a whopping $460 a minute. No wonder Sheffield went to Europe!

Actually, 45 minutes is just about the exact amount of time needed to fill an LP, which usually runs to 22 minutes per side. But CD is another matter entirely.

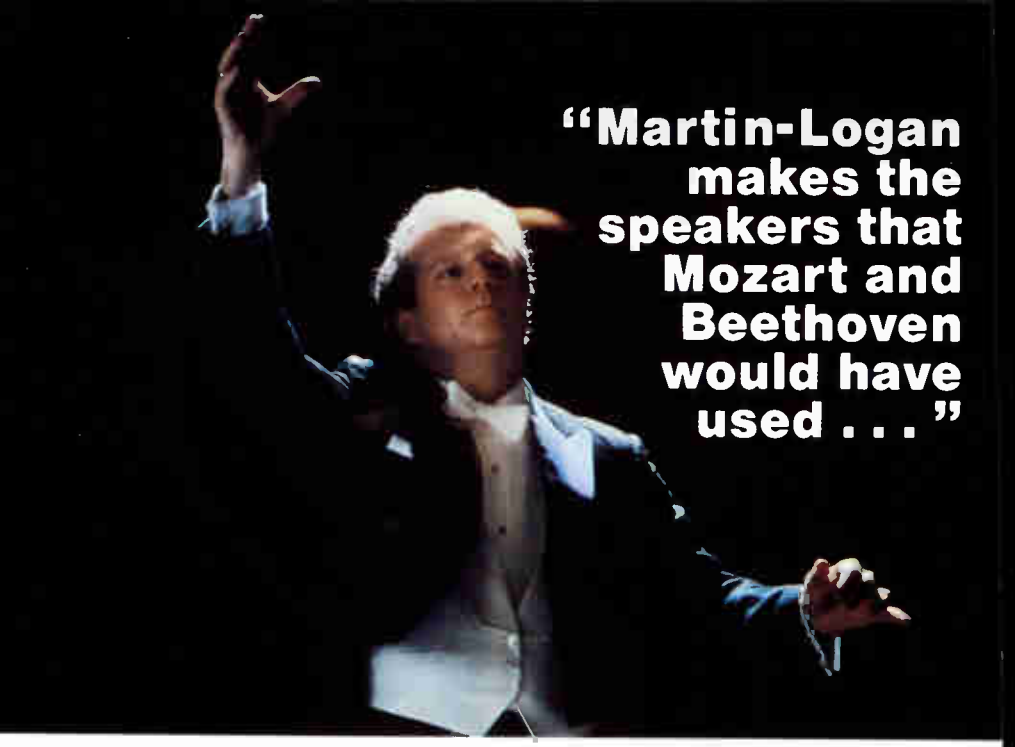
A CD buyer who finds he has bought substantially less than 60 minutes of music for his $13.95 feels gypped—particularly when he looks at the 70-minute releases coming routinely from RCA these days. This means that a CD producer *must* get his full 15 minutes of usable material from every hour of recording. The chances of this happening depend on the competence of the orchestra members and the producer's standards for perfection in performance, and those chances are rather slim. The alternative is to go into overtime, at time and a half for everybody, or $5000 for half an hour for 100 players. And most AF of M shop stewards are meticulous about their timekeeping, often standing in the wings with a stop watch, fully prepared to call an end to the session after the 59th second of the final minute. If just two more minutes for a short retake are all that would be needed to get a perfect performance, the producer is faced with an agonizing choice between artistic integrity and Big Bucks. Guess what usually triumphs?

But let's forget about the musicians for a moment and think about the music.

Every symphonic musician knows that he plays best on those special occasions when the orchestra and the audience are hyping each other—those rare times when the tension between them is almost electric. When this happens, the excitement and inspiration of that performance can transcend any shortcomings in the orchestra's execution. It hardly needs to be said that this almost never happens at a recording session.

Typically, a commercial recording is made

1 This may sound like a good hourly rate, but in actual fact, a typical American orchestra does so little commercial recording that it does not represent a major source of income.—LA



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*Mr. Slatkin is the conductor of the St. Louis Symphony.

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up of bits and snatches of different "takes," often played in a completely different order from the normal one. The result may be "note-perfect," but in no way can it be considered a "performance." So what do we have to show for 100 years of sometimes-inspired symphonic performances in America? Practically nothing, because the only symphonic recordings available are the result of recording sessions.

Symphony orchestra performing is the *only* artistic endeavor for which we have no historic record whatsoever! Most of the inspired performances of the greatest conductors of the past century are gone forever, leaving not a trace except for written concert reviews, which can only give a hint as to what the performances sounded like.[2] This is criminal!

Due to some AF of M rule relaxations in recent years, orchestra performances *can* now be routinely taped for broadcast or private archiving, without additional reimbursement to the players, and individual Union Locals are free to choose their own recording fees—as long as no recordings or broadcasts are disseminated beyond the geographical boundaries of that Local. But anything going outside that area automatically calls up the national Rules and the national scale for live-performance recordings. This means a civic orchestra could sell live-performance recordings in its own area, but without national distribution, no recording would be likely to do much better than pay for itself. Give it national distribution at national recording rates, and it would have to sell thousands *more* copies before *it* would break even. It's a can't-win situation.

It's too late now to do anything about those lost definitive performances, or the lost opportunities of struggling orchestras to supplement their income from recordings, but it isn't too late to change the situation. That is, if the AF of M has the courage to consider changing some of its archaic and self-destructive recording policies.

Before this can happen, it is necessary that the musicians' union stop its paranoid thinking about symphonic recordings. They are *not* a threat to orchestra musicians, but a potential boon. Unlike popular recordings, classical symphonic recordings are never used as a substitute for live musicians, because they are never played in places or situations where a live orchestra could ever be employed anyway, such as in a restaurant. And the appeal of symphonic concerts and symphonic recordings is so different that the latter cannot possibly be considered as being in competition with the former.

People do not attend orchestra concerts because they don't have the music on records. They attend because they like to go to orchestra concerts, and the love of symphonic music is probably the *least* important of the reasons for that liking. More important is the feeling that it is a special occasion, the desire to be seen in a socially acceptable environment by one's peers, a liking for the sound of a live orchestra (which, as I noted above, has still to be replicated in the home), and the appeal to one's herd instinct that comes from sharing an enthusiasm with a large crowd of like-minded people. Few music-lovers use symphonic recordings as a substitute for the real thing; the symphonic record collector who rarely attends concerts would not attend any more often if records did not exist, because the unique appeal of the concert hall is not changed in any way by the existence of recordings.[3] The recording can only supplement the live concert; it cannot replace it.

In other words, it would make good logical and economic sense to liberalize the AF of M's recording rules pertaining exclusively to symphony orchestras. Here are my suggestions for those new rules:

- The "standard" recording rate *must* be abolished for symphony orchestras and opera companies. The hourly pay rate for formal recording sessions should be determined through negotiations between the record company or producer and the representatives of the orchestra and its Local, but should never

2 A very few orchestras which have been regularly broadcast through the years, such as the Philadelphia and the New York Philharmonic, were, of course, routinely taped, but where most of those tapes are now is anybody's guess.

3 Whoa, JGH! This is untrue in my case. If records did not exist, I'd be going to concerts *much* more than I do. Besides, if records did not exist, neither would symphonic record collectors. Also, the unique appeal of the concert hall *has* been changed by the existence of records: orchestras play much more loudly than they did at the turn of the century, and most audiences have come to expect to hear the perfection they're used to from records. I'm not sure our orchestras' overall levels of proficiency have increased as much as our appreciation of hearing live music at all has decreased. —RL

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be less than what each musician is paid for a concert performance. This would make second-string groups economically competitive with the top-ranking orchestras for recordings.

- There should be no restrictions on how much session time per hour can be used for a final release. The 15-minute-per-hour restriction would unfairly handicap the best orchestras when they are being paid a higher rate than their lesser counterparts. If one hour of recording produces one hour of releasable material (very unlikely), all of this should be available for use.
- The management and musical director of all orchestras should be free to authorize archival recordings of all public performances, and their broadcast by noncommercial radio stations, without approval by or additional payment to the orchestra members, and should be encouraged to do such routine archiving.

Routine archiving would accomplish several things. First, it would allow the building, over time, of a library of real performances, for whatever purposes these might later be put to. Second, it would give the recording personnel the time to experiment with different purist mike placements, in the hope of eventually finding one that works in that hall. And third, it is insurance against performance piracy, which is easier to pull off than most orchestra managements realize. (Comparing a performance tape with one that is suspected of being an illicit recording of that performance will conclusively establish the truth.)

- If the decision is made to release a recording of a performance for national distribution, tion, the players should be paid *no additional money* for the recording, because it does not represent additional work on their part. Instead, a substantial royalty on each record sold should be paid by the record producer to the orchestra fund, with a previously contracted proportion of that money to be disbursed among the orchestra personnel. The royalty amount should be negotiated and contracted for before record production begins.

Royalties from records sold, rather than payment for the recording itself, would encourage a backer to invest in the possibility that a title of dubious sales potential will sell well enough to release. If it didn't, the investor is out only his production and promotion costs and the musicians have lost nothing. If the record makes money, everyone involved should continue to make money from it as long as it continues to sell. With the present arrangement, guaranteeing a three-hour session reimbursement at the outset, a record producer would be foolish to consider releasing anything that didn't look like a sure-fire best-seller.

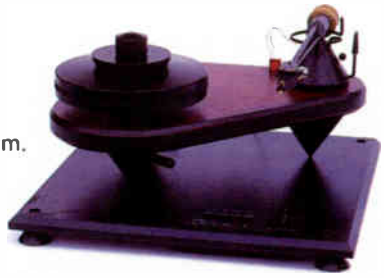
- Finally, the three-hour minimum for a formal recording session should be reduced to two, because without the 15-minute-per-hour restriction, a good orchestra could probably turn out enough clean material for one CD in that two-hour period. Anything less than a two-hour session would be impractical for those orchestra members who live a "commuting" distance from the concert hall, and would (rightfully) balk at driving an (unreimbursed) hour for an hour's work.

These revised recording rules would make many more American orchestras available for recording, and would provide a means for getting some variety into the rather mundane musical fare now available on disc. With the recording rate what it is today, only the major record companies can afford to record American orchestras, and the majors tend to opt for warhorse titles or glamorous performers with proven mass-market appeal. Bringing our second-string orchestras into the recording pool could encourage, once again, the recording of lesser-known but no less enjoyable works in the symphonic repertoire. (Who remembers Don Gillis, John Alden Carpenter, or Randall Thompson?)

As for supplemental income for starving orchestras, I won't pretend that one modestly successful recording of Chadwick's *Suite Symphonique* will save a civic orchestra from bankruptcy, but the slow, steady sale of ten such titles could make a significant difference in its financial health.

Then there's the dividend for us audiophiles. The opportunity for serious amateur recordists around the country to experiment at leisure (and without cost) with minimalist mike techniques will result in some superb concert recordings of the kind high-end audio is all about. And the opportunity to release these recordings for public sale could result in a bonanza for audiophiles.

Revised symphonic recording rules would benefit all concerned. It's past time they were considered. **S**



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LETTERS

We regret that time does not permit us to reply individually to letters, particularly those requesting advice about particular equipment purchases. Were we to do this, a significant service charge would have to be assessed—and we don't have time to do it anyway! Although all are read and noted, only those of general interest are selected for publication.

Endless, contradictory quibbling

Editor:

It is time someone told you Emperors about your new clothes. Or shall I say that medievai theologians debating how many angels could fit on the head of a pin didn't have anything on you guys? I refer to the endless, repetitive, and often contradictory quibbling about which piece of equipment meets which specification.

I am a professional musician, and records and tapes are cutting my throat. Did it every occur to you or your listeners to spend money attending live musical events? Did it ever occur to you that the average hall where you hear live music is *bigger* than a living room, and that no amount of money is going to enlarge the airspace therein? By definition, music is an art, and science and engineering are opposite in spirit. You and *The Absolute Sound* are just on ego trips to see who can spend the most money. Have you ever spent time trying to play an instrument, or to sing?

Let me end by saying that a good reason European orchestras can charge so much less than US orchestras is that in Europe they're tax-supported, and here they aren't. Believe me, check around and tell me how much orchestra personnel are getting here—it isn't much. Certainly it isn't enough to spend thousands on an ersatz means of copying music. The fact that you failed to note *why* American orchestras are "astronomically high"[1] in rates proves your ignorance about the workings of music in your native land. You are like the art critic who wrote a review about a piece of sculpture and ended up by giving its physical dimensions, color, weight, etc., but didn't even grasp its artistic intent or merit.

Hope you have fun with those angels. I've got to go play some live music—and it'll sound better than any record ever made.

John R. Snyder
Knoxville, TN

1 Vol.10 No.3, p.69

JA used to be a professional musician, and still plays as an amateur; choral singing used to be an important part of LA's life until Stereophile *grew to the point where it occupied 90% of his time (the other 10% being spent comparing Chardonnays and playing volleyball); RL used to play bassoon and at present is studying classical singing; AG used to play guitar more seriously than he does now; JGH, BS, and GG have spent more evenings than most recording live orchestral performances; DO experiences live music vicariously through his wife's semiprofessional singing; but you do have a point, Mr. Snyder: it* is *easy for hi-fi writers to lose touch with what it's all about, and recorded music still doesn't approach the real thing. I don't think, however, that that's a good reason for giving up!*

Regarding your comments on the high recording fees charged by American orchestras, JGH goes into this subject in some detail starting on p.5 of this issue. But I take exception with your comments about all *European orchestras being supported by taxes. The London orchestras (LPO, RPO, LSO, Philharmonia) are probably recorded more often than any others, yet only a small part of their incomes is derived from the taxpayer.* **—JA**

In defence of the Marantz 10B

Editor:

There is a saying that "the pen is mightier than the sword." However, in the case of Don Scott's review—25 years late—of the Marantz 10B tuner in Vol.10 No.4, I think that the pen *is* the sword. It is a sad fact that some (not all) of the writers for the various "underground" magazines have been lacking in technical expertise; as a result of this incompetence they have grossly misled the public (who are even less technically competent) on innumerable occasions. Don Scott's review is a case in point.

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High performance. Down to earth.

by Sidney S. Smith, the Director of Engineering for Marantz. He called it a "Quasi-Vestigial Sideband Filter," and it was also incorporated into the Sequerra tuner and, in a modified form, my "Charlie" tuner. I am not going to do some free engineering here for the uninformed, but one of the *primary* requirements of this filtering technique was the use of a 67kHz notch filter to eliminate the Storecast (SCA) signal. It is axiomatic that virtually *every* engineer of any merit *knows* the importance of having an SCA filter. To imply that the 10B did not is sheer blasphemy (in the true sense of the word) and a gross accusation of incompetence on the part of Sid Smith and Dick Sequerra.

The fact of the matter is that the 10B has four groups of filters: a 19kHz notch, a 67kHz notch, a 24kHz notch, and a 38kHz notch. The first two are part of the "vestigial" filter and the second two are the output filters. (DS is incorrect in identifying these last two as 19kHz and 38kHz filters.) These two filters also serve as part of the coefficients of the de-emphasis network. It is absolute stupidity to advise any 10B owner to go in and modify the filters, as everything would then be totally misaligned.

Wiser heads should prevail in this matter. Quite obviously, something was amiss regarding the so-called units under scrutiny for this "review." May I suggest the following possibilities: a) the units were out of alignment; b) the units were somehow defective (burned-out or shorted-out parts, etc.); c) the units had been tampered with. In addition, it would have been wiser to investigate the problem *before* accusing the designer of incompetence. The schematic—which *clearly* shows all the filter functions—could have been obtained. Either Marantz, or preferably Sid Smith, could have been contacted to ascertain the situation. Obviously, none of these things were attempted—shame on you, *Stereophile*! Lastly, archives of the test reports from 20-25 years ago could have been located (with maybe a little bit of effort) showing the performance of this legend. Indeed, when properly aligned—and we have three of these legends among our coterie of tube lovers here in Santa Barbara, including my own—there will be *no* SCA interference from a 10B tuner.

Now I am not going to claim that the 10B is absolutely the finest tuner ever made; indeed, a great many other tuners outperform the 10B on *individual* points. On an overall basis, however, there can be no doubt as to the *total* musicality of this legend. I and many others have listened to countless tuners down through the years (including my own), and we all *unanimously* keep returning to the 10B. In an overall sense, this legend has yet to be equalled.

James Bongiorno
Santa Barbara, CA

A properly aligned 10B has good IF bandpass and allows the undesired 67kHz SCA component to pass with little attenuation. Unusually stiff 67kHz filtering is needed, therefore, to avoid audible birdies (at least 45dB) and to prevent it from contributing to HF IM (80dB).

The single LC filter in the 10B doesn't cut it in this respect. Footnote 4 to my review should read, ". . . The 10B has four filter sections to remove 19, 24, 38, and 67kHz components. An additional section (the 24kHz) can be retuned by changing capacitor values and adjusting to double-notch the 67kHz component. I apologize for the lack of clarity and I thank Mr. Bongiorno for keeping me on my toes. **—DAS**

Solid-core cable

Editor:

Alvin Gold's article in Vol.10 No.4 regarding solid *vs* stranded speaker wire was very, very refreshing. However, in his point discussing low-level spuriae ("garbage") using the example of a 1000-strand cable with 3kHz signal, he multiplies the units "strands" with "Hertz" and ends up with a resultant with the units of "Hertz" again. Shouldn't the end result really be in units of "stranded-Hertz"? With so many of these little spuriae, it sure the hell would even "hert" a Monster's ears! **NVR**

Balboa, CA

You can find the Monster's response in Vol.10 No.5.

More solid-core

Editor:

Regarding Alvin Gold's article about solid-core cable: I used to use Monster Cable, but for the last 6 months have been using low-impedance, 16-gauge, solid-core Fire Alarm Cable, with roughly a 12-foot run per speaker. (An electrician friend gave me a 200-foot roll which I have distributed to all my fellow hi-fi freaks—I

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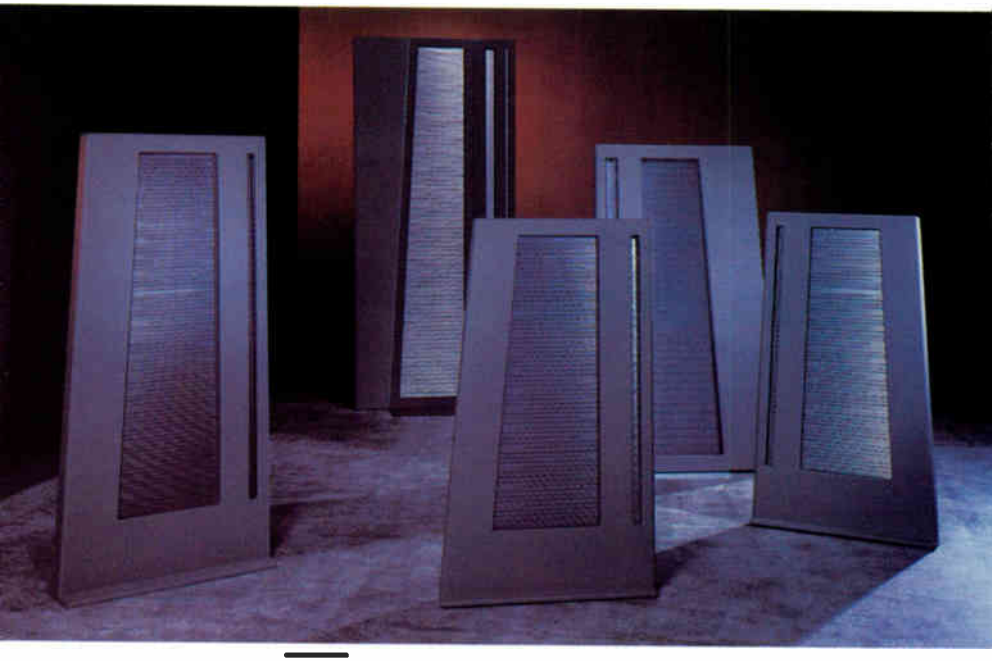

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still have over 100 feet left, if you want some.) I have been told that solid-core cable can store current, sending vicious spikes to the loudspeakers and possibly damaging both them and the amplifier or receiver.

I haven't damaged anything yet, neither my Nakamichi SR3A or my KEFs. I haven't figured out how to use solid-core between my Rega or my Shure CD player and the Nakamichi, but with it used as a speaker cable, my KEF R104/2s sound as if a curtain which was veiling the sound has been raised. There is less bass than Monster Cable, but the sound is clearer, with more information.

Kenneth McAulty
Framingham, MA

Solid-core again

Editor:

AG's article on small-gauge, solid-core loudspeaker cables in Vol.10 No.4 was fascinating, and, in my own case, fortuitously timely.

The reason for its timeliness is the fact that I have just purchased a new sound system in accordance with the word from *Stereophile*, namely, an Adcom preamplifier and amplifier, a pair of Magnepan MGIIc loudspeakers, and a Dual CS5000 turntable and a Sony CD-P520ES CD player. I have the loudspeakers located across the room from the rest of the equipment, connected to the amplifier via 30 feet of doubled wire (to accommodate the two leads) to each speaker. For the present I am using 16-gauge, stranded zip-cord instead of the usual 20- or smaller-gauge wire usually supplied for speakers.

My dealer is pressing me to buy very expensive, exotic loudspeaker cables. I have been holding off, however, because I am torn between *Stereophile*'s recommendations and my reluctance to spend the kind of money those exotic cables require, especially for such a long run.

Enter Alvin Gold's arresting article on solid-core cable. I want to start experimenting—now. While I shall communicate with DNM in England, what I'd like to do now is undertake some preliminary experiments with solid-core wire. Can you tell me what the American gauge equivalent of the 0.5mm² wire you suggested is? And what gauge equivalent would be needed for a 30-foot run for each lead?

John C. Guenther
Stuart, Florida

Solid-core cable revisited

Editor:

After reading Alvin Gold's article about solid-core wiring, I decided to give it a try. Some cheap ($0.04/ft.), 18-gauge, hardware-store wire, laid at least 6" apart on my floor, gave amazing results compared with my Straight Wire Music Ribbon 32s. My only hesitation was the body of the sound and the bass, which seemed a touch thin in my system (B&K 140 amplifier and Celestion SL6S speakers). Since 18 gauge is larger than the size specified by Gold (0.8mm²), I experimented with some good-quality Belden magnet wire. I first tried the Belden 18 gauge—the sound became smoother and fuller, sounding about the same as the Music Ribbon fullness. Then I tried the recommended 20 gauge. Lo and behold, the sound was cleaner, more open, and the bass just as good as the 18 gauge. Compared with the Music Ribbon, the sound is much more transparent and open, transients are sharper, the imaging and soundstaging is greater—there is no feeling at all of the speakers' presence. The music is now really alive in feel.

I do not know how this wire will work with a system that can go way down into the bass, but you definitely should follow up this idea of solid-core wire in a proper review. I plan to sandwich the Belden wires between duct tape and have my own ribbon cable at less than one hundredth the cost of my new (damn it) Music Ribbon 32s. It's not too pretty, but think of all those records one could buy with the difference in price. Next, I'll try the interconnects, but may not be able to overcome the insulation problem since those wires have to run a lot closer.

Paul Isaacs
Brooklyn, NY

Solid-core from Radio Shack

Editor:

Apropos Alvin Gold's article on thin solid-core cable, and with apologies to the Audio Cheapskate, I highly recommend Radio Shack indoor (white) 300-ohm twin lead both for interconnects and speaker cable. It costs $1.59 for 40 feet! (The thicker, brown, outdoor stuff sounds edgier.)

Along the way from tolerating my stereo toward relaxing with it, I spent several hundred dollars on variously concocted intercon-

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nects and cables. The worst were the twisted multi-gauge multiconductor type—lots of initially beguiling fake spaciousness and etched ersatz detail. The best was a speaker cable by Naim that looked like industrial-strength twin lead, with many strands. However, compared to thin TV twin lead, it was quite muddled in a woolly, rich-warm manner.

We're talking *major* improvement here, like getting more musical electronics: layers of glare removed, and simple clarity that lets you sink into the music and its interpretation, *sans* cringe factor.

I've got no axe to grind, but I have long felt remiss for not sharing this inexpensive serendipity with the cosmos. If you disagree, you're out only four cents a foot for the privilege. It may be relevant that my electronics are all tube: Dynatuner, Audible Illusions preamp, Quicksilver amps, 30 feet of twin lead to KEF 104ab speakers.

By the way, here's a cheap and easy solid-core experiment: connect preamp and amp chassis together, and use 10-gauge, aluminum TV ground-wire between phono receptacles. (It fits snugly without connectors.) Too much buzz for long-term listening, but a very coherent, non-strident musical image results. The last time I tried this, I liked 10-gauge aluminum better than 14-gauge copper. Really.

Now, having clearly certified myself as an audio visionary, permit me another minor iconoclasm. You can subtly improve on brass or aluminum cones under electronics by replacing them with hardwood knobs (three for a dollar at the hardware store). The best sound I've found are the 1¼-inch diameter made by Brainerd Mfg. in East Rochester, NY 14445. They look like little mushrooms and sound best upright. Their rounded tops approximate a single-point contact with electronics placed on top of them.

Finally, let me corroborate fully Mr. Duray-Bito's letter about VPI Bricks (Vol.10 No.4). I have four of them and they make nice doorstops.

Thank you for a highly informative publication, and a special nostalgic thanks to JGH, whom I've read off and on for 20-plus years.

Robert B. Laird
Rochester, NY

Having experimented with both Radio Shack 300-ohm cables, I feel the edgier sound of the outdoor stuff is due to the PVC sleeving surrounding the foam dielectric. The indoor has plain polyethylene insulation, considered to be one of the better-sounding dielectric materials. Regarding Mr. McAulty's letter, there is no way for solid-core cable to "store current, sending vicious spikes to the loudspeakers."

—JA

Solid-core & science

Editor:

A good equipment review will inform the reader of the nature of the problem that a designer had identified and solved. As a design tutorial, any review is useful to us all, regardless of our interest in the unit under test. Mr. Gold's article on solid-core cables was interesting, but he has made several inaccuracies which detract from the case for solid-core cable.

Mr. Gold presents two examples of electricity without pointing out that one is wrong. He writes, "one model of current flow has electrons weaving through the conductors, setting up electric and magnetic fields around them." This would require that electrons, which are matter, travel at the speed of light. The nature of current flow is, as he presents it in his second model, the electromagnetic effect, which *does* travel at the speed of light, and causes charged particles to move in response to the force exerted by the electric field. Electron movement is the *result* of an electrical field, not the cause of it.

In fact, the progress of an electron down a wire is extremely slow. Newton's Second Law says the electron's acceleration is proportional to the force applied to it. In a vacuum, the speed of an electron is a function of the strength of the electrical field. In a conductor, things are somewhat more complicated; judging from his concern with "random current flow," Mr. Gold may not be aware of this. The electrons lose some of their kinetic energy by bumping into the atoms of the conductor, causing the conductor's atoms to move faster (heat up). After all this banging about and rebounding in all directions and with the energy lost as heat, the electron's progress is slow. Consider that in a wire with an area of 1mm² carrying 20 amps(!), the net velocity of an electron is less than 1m in 10 minutes. This is a velocity of furlongs per fortnight, which moots Mr. Gold's point in his

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"startling conclusions" that we should avoid a too large conductor because "the excess metal creates conditions suitable for random current circulation."

Mr. Gold writes: "The key idea underlying solid-core cable is to constrain the behavior of the electrical current as far as possible." Actually, the voltage is constrained in the sense that the cable presents a nontrivial ohmic load, but the real purpose of the cable is to constrain the electromagnetic effect to an area where it is governed by the "skin effects." When AC travels on a wire, it does so largely at the conductor-insulator boundary. The wire acts as a "wave guide" for the electromagnetic effect. The higher the frequency of the signal, the less energy is present in the center of the wire. This tendency is called the skin effect. The degree to which the energy is only near the surface is reflected in a figure called the skin depth, which refers to how deeply energy has penetrated the conductor at a given frequency.

Dr. Malcolm Hawksford gives the following information for skin depth *vs* frequency: at 50Hz the depth is 9.35mm, at 1000Hz the depth is 2.09mm, and at 20kHz the depth is 0.47mm. Dr. Hawksford says that when the electric field's wavefront passes in toward the center of the conductor, its propagation direction becomes not quite straight down the wire. These non-straight paths arrive later, causing the wavefront to be time-dispersed. Since the skin effect varies with frequency, we have a frequency-dependent, time-dispersive smear. The solution is to configure a conductor so small in diameter that for *all* frequencies of interest the skin effect will predominate. Ergo, conductors of 0.5mm diameter.

The effect of the resistance due to the smaller size wire will be linear if the wire is ohmic, *ie*, its resistance doesn't change regardless of the applied current. From the list of skin depths, it may be seen that at bass frequencies one can have thick, single-core wire and still have a wire diameter much less than the skin-effect depth. In this regard, Mr. Gold's point #3 needs some modification. It is not important that all wires be thin, but that they be thin compared with the skin depth of the frequencies that they carry.

I question Mr. Gold's discussion of low-frequency damping. Any cable resistance will always worsen (decouple) the amplifier's control of the woofer. Servocontrol is a wonderful solution if properly implemented, but it is an odd solution for someone who mistrusts active stages. Mr. Gold, referring to loudspeaker drive units, suggests that "the real concern should be to DRIVE it, a much more fundamental requirement." A noble sentiment, but everything that is driven spits back current when the "drive" is over and the unit is coasting on inertia. Servocontrol will obviate many of the same problems which low-resistance cables minimize, but it is almost always better to avoid an undesirable effect than to correct for it later. It is possible to incorporate series resistance into a low-frequency alignment; however, this only accounts for behavior near resonance. Above resonance, the amp will always provide better control of the woofer when the series resistance is low.

I think Mr. Gold slightly overstates the case when he says, "There was nothing in conventional theory that would explain why" small solid-cores work so well. These effects are routinely accounted for in microwave and radio design. In radio telescopes, not only are "wires" hollow, thereby avoiding the effects of the center core, but they are also cooled to diminish the background thermal noise floor. Perhaps in the late 1980s, the earnest audio geek will have cryogenic cables, only to be superseded by superconducting cables in the 1990s. Then we may view contemporary cable prices with nostalgia.

Daniel Patrick Coyle
Grants, NM

Review errors #1

Editor:

I think JGH has let himself be misled by the documentary failures of the Recut and Chesky records he reviewed in Vol.10 No.4.

First, Kleiber's Beethoven Fifth was recorded (on tape) in the Concertgebouw in September 1953, initially in nine parts for potential issue on 78s, but published only as LPs in England (LXT 3022) and the US (as London LL-912). This performance has, of course, been reissued since, most recently here on Richmond 23232.

Very likely the Kleiber was balanced by Kenneth Wilkinson, who was often on the road for Decca in the late 1940s and early 1950s. And while I don't have the evidence, I am fairly sure he also balanced the Bartok and Ravel concer-

tos with Katchen, which were recorded in Kingsway Hall between the 9th and 13th of November 1965.

I *am* sure that he balanced Barbirolli's Sibelius Second from sessions on the 1st and 9th of October, 1962. And though Wilkie would be pleased with JGH's general comments about the sound, he would not be pleased to know that JGH feels there isn't enough middle in the stereo, since here, as in all his records, Wilkie was certainly using his trademark Neumann M-50s on a Decca "Tree" over Barbirolli's head, as well as his '50 outriggers and a few in-orchestra spots. Certainly the Tree-to-Outrigger-to-Spot balance changed from session to session; one thing *is* certain—the Tree was there to fill the middle of the mix—*always*.

Mike Gray
Alexandria, VA

Review errors #2

Editor:

I hope JGH pays more attention to detail in his equipment evaluations than he does in his record reviews (Vol.10 No.4). The orchestra led by Lorin Maazel in his spectacular recording of the Tchaikovsky Second/Rimsky-Korsakov "Antar" symphonies (Telarc 80131) is the *Pittsburgh* Symphony, not the Cleveland. Mr. Maazel has recently cemented a promising relationship with the PSO which hopefully will produce many more recordings of this caliber.

P. Holcomb
Pittsburgh, PA

The greening of the Monolith

Editor:

Stereophile's evaluations of the Martin-Logan Monoliths were very much on target. However, an important change has made the Monoliths into much more pleasant and musical speakers, passing *my* goosebump test with ease. As you may know, the power supplies have been "upgraded" and, lest you think this is minor, have transformed this speaker into one which has few, if any, of the problems you accurately pointed out.

For example, if you have an older pair of Logans, remove the old power supply and replace it with the newest power supply. (M-L supplies this change at modest cost, and it is worth it!) Second, forget the Martin-Logan recommendations *not* to biamp. Biamp these suckers. Third, rewire the woofers for dual voice-coil use, which changes the impedance to 4 ohms. And last, find an amplifier which will drive the rather difficult load the M-Ls present (made much easier with the new power supplies), and use a 6dB/octave slope at 100Hz. (The Ashly SC-22 does quite well combined with a McIntosh 2100.) Experiment with slopes and impedances for best results (16 ohms on the electrostats with 4 ohms on the woofers can sound excellent). If you have an old Mac lying around, try it. It can sound very smooth and powerful with the Monolith.

The chief key is to remind Martin-Logan owners to have that "old" power supply changed. They won't believe the difference.

Thanks for a great magazine. Keep them coming.

Carl E. Miller
Columbus, OH

It is, isn't it?

Editor:

All musical instruments have sounds that bloom and decay. The LP record is vastly superior to CD in this regard.

Charles J. Gaton
Bayport, NY

Yes, it is!

Editor:

I thoroughly enjoyed reading Martin Colloms's introductory remarks in his overall conclusion to his two-part review of CD players (Vol.10 No.4). He so clearly described what I do not hear when I listen to a CD front end that I was happy to find out that it is a shared experience. Up to this point, I had been baffled by why I had not read a "something missing" comment on the sound produced by a CD player. Consequently, I had assumed that the problem was caused by the equipment I was listening to and not inherent either to CD players or to the CD medium itself. Mr. Colloms seems to favor the opinion that the problem is with the players, *not* the discs.

I do not notice anything missing from the music when I listen to a popular song that only plays for two or three minutes. But if I listen to a concerto or symphony, then it becomes apparent as the piece progresses that something is intrinsically missing from the music. Mr. Colloms's statement, "Conscious thought often returns a null result: you can't hear any-

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thing specifically wrong," is so right on the mark that I only wish he had written it sooner. It would have saved me from some moments of aggravation in which I attempted to deduce what caused my feelings of uneasiness and irritation while listening to a CD front end for more than a few minutes at a time.

Until this problem with CD front ends is resolved, I will continue to use my turntable and heed William Shakespeare's advice (*The Merchant of Venice*, V, i):

"Here will we sit and let the sounds of music creep in our ears. Soft stillness and the night become the touches of sweet harmony."

Raymond Stanton
San Francisco, CA

Why no Infinity RS1Bs?

Editor:

Thank you for the "A Matter of Taste" series. I found it one of the most enjoyable reads in over three years of reading your magazine, and it raises a comment and a question.

The comment: Tellig may not know good sound, but he sure knows good Scotch!

The question: About a year ago there were numerous raves about the Infinity RS1Bs, yet none of the writers uses them in their systems. Why not?

Bob Snider
Austin, TX

Both AHC and JGH had the RS1Bs as reference systems for some time, but, by the time "A Matter of Taste" came out, both had moved on to other, preferable, systems. **—JA**

Listening is hard

Editor:

The recent letters and articles on blind A/B testing have caused me to think about the different ways that people listen to music. An analogy comes to mind with the different ways that people eat.

First is fast-food, where the food is sustenance only, eaten without enjoyment. This is analogous to when the music is heard, but not really listened to. An example of this is background music, where only something extraordinary may catch your attention.

Second is what I will call the "meat and potatoes" meal. A person becomes accustomed to the fare served in his home; while he may enjoy a meal, he doesn't dwell on the flavors involved. The analog is the mid-fi audio of the average American consumer. Usually a person is comfortable with his stereo system, and will accept its sound as the norm. He will enjoy the music, and only occasionally will the quality of the sound intrude. This is the prime market for CD. After all, frozen vegetables taste better than canned vegetables.

Next is the gourmet meal. The gourmet savors the food, and enjoys the meal as a complete experience. He may not dwell on a particular taste, but may notice things such as a delicate sauce or an interesting combination of herbs and spices. The analog here is the audiophile. The most important trait of the audiophile is that he listens to the music. Music is enjoyed as a holistic, gestalt experience. As with the gourmet, he may notice something in the sound such as a recording that is too bright, or that a particular performance is very smooth and mellow. Minor imperfections in the sound will be overlooked, but major aberrations will detract from the musical performance. Many of my musician friends form an interesting faction of this group: they don't care at all about the sound of their systems—only the music.

As opposed to the gourmet, the taste tester scrutinizes the taste of a single dish, dissecting the flavor into its components, which are evaluated individually. The experienced taster may be able to go further than just evaluating a dish; he may be able to suggest corrective action: Does this dish require less sugar, or a dash of lemon juice? Parenthetically, I wonder if there are tasters who claim to be able to tell if a dish was prepared in an aluminum pan or a glass pan? Or detect a difference introduced by an electric oven *vs* a gas oven? Is there the equivalent of analog and digital cuisine?

The analog of the taste tester is the component reviewer. The reviewer, like the audiophile, will listen to the music to get a general impression, and will note areas of interest to be examined closely. After a period of introduction to a new component, a formal review will be conducted. The important difference is that during the formal component review, the reviewer will listen to the *sound* rather than the music. The program source (no longer called music—and in fact may include material other than music) is carefully chosen to display a particular sonic characteristic. The reviewer

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will scrutinize the single characteristic of interest by playing the same source material over and over. Sometimes the source will last a few seconds, at most a few minutes. The standard to which the sound is compared is live sound; however, the reviewer will periodically replace the component under test with his reference to make sure that he is hearing what he thinks he is hearing. I might note that some reviewers keep many components on hand to use as tools in a review. A particular component may be prized for its analytical nature and ability to show flaws in other components, but may not be included in the reviewer's system for recreational listening.

As with any other skill, reviewing must be learned and improved with practice. The first step is being able to detect and identify something in the sound, such as excess brightness or poor imaging. The second step is being able to attribute what you hear to a physical property such as a resonance, or nonlinear frequency response. An experienced reviewer may be able to suggest a remedy, such as "the muddy bass is caused by polyester capacitors, which should be replaced by polypropylene units."

A variation of the individual component review is the component comparison, where two (but only two) components are compared against each other. The same procedures are used in evaluating the components, but rather than describing their sound referenced to live sound, their sounds are compared with each other. This is quite different from a blind A/B test. I must admit that I have never participated in a blind test, but I should think that it would be rather like being blindfolded, offered a sample of shrimp scampi from one dish, offered another sample of shrimp scampi from a second dish, and then being asked to identify the source of a third sample. I'm sure that I would fail dismally: Hey, the shrimp scampi are all the same—just like all amplifiers. The reason I would fail the test is not that the shrimp scampi samples were the same, but rather, I am not experienced in this sort of test. I'm sure that it's something that could be learned.

An audio blind A/B test with listeners not experienced in this kind of test, listening to unfamiliar music, and switching between many unfamiliar components, is doomed to failure. Let me take each problem in turn: first, the listener must be experienced. He must first learn to listen to the sound rather than the music. In some ways this is similar to the skill developed for a conventional review. There is, however, an important difference. In a conventional component evaluation, the reviewer can concentrate on a particular aspect of the sound. This is not the case with the blind test. In the first phase of a blind test the reviewer must take a mental snapshot of the whole sound, analyze that sound, and place a holistic image in his short-term memory. This must be repeated for each component in the test. During the phase of the test requiring that a component be identified, the listener must again make a holistic image, retrieve each of the reference images from short-term memory, and compare each reference in turn with the current sample. It goes without saying that the difficulty increases dramatically with the number of components in the test. I suspect that it may be impossible to reliably compare more than two components at one time. Certainly you could not expect the average person off the street, or even the average audiophile, to participate in such an activity with any sense of confidence in the results.

Discerning the difference between two very similar components should be very difficult. If the reviewer is intimately familiar with one of the components being compared, it eases things a bit because presumably he would have already established a holistic image in memory.

Source material to be used in the test must be chosen with great care. Imagine for a moment that you listened to a symphony orchestra through component A, then a rock group through component B. Then you were presented with a recording of a folk singer and asked to identify which component was used. I admit that this is absurd, but it does make the point. Source material chosen for the test must have two characteristics: first, it must be homogeneous. That is to say, it must not change between the time the reference image is made and the comparison image is made. Second, the source material must clearly reveal a particular characteristic. Let's say that two components were identical except for their bass characteristics, and only 10% of the source material contained adequate bass information. One could only hope to choose correctly on

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the 10% of the trials with the bass present. Even if the listener performed perfectly—10% correct choices—the 90% random choices would prove that there was no difference between the components.

Further, this implies that there must be a series of tests, one for each characteristic. This means you would be A/B comparing two components for only a single characteristic—for instance, bass extension. Assuming that choices were only allowed when the characteristic in question was present in the source, then all that could be inferred from random results is that there was no discernible difference between the two components for this single characteristic.

I believe that there could be a place for blind A/B testing in the audio arena, but that a lot of work needs to be done before we can rely on the results.

David W. Davenport
Raleigh, NC

But bias is bias

Editor:

I have been reading the items about "The Great Debate" which have been appearing regularly in *Stereophile*. Perhaps I'm misinformed, but I cannot seem to locate the "significant literature" (Vol.10 No.4, p.25) which has been systematically ignored by *Stereo Review*. Your references, most usually contained in a footnote, appear to refer to the January and May 1986 issues of *HFN/RR* and to the Leventhal paper in the *JAES* and the follow-up in *Stereophile*. I have reviewed all of these carefully and cannot imagine how you can conclude they have been ignored or even that they offer conflicting evidence.

Dr. Leventhal suggests that "small-to-moderate" differences may not be uncovered in a 16-trial test with "small-to-moderate" defined as being present 60% to 80% of the time, and urges caution in interpreting the results of small-N tests. He nowhere takes any issue with the test procedure or even really makes any statement that refutes the results. The *Stereo Review* tests were published after Mr. Leventhal's work was published, and to the best of my knowledge he has never commented on them publicly. The *SR* tests seem to meet any sample-size criteria handily in any case.

I also reviewed John Atkinson's Phase/Capacitor test in *HFN/RR*. Again, the results in each test JA ran are almost exactly those one would expect from chance. Even his conclusion stated that the tests didn't provide ironclad support for the notion that capacitors sound different. Likewise, a review of the otherwise well-designed Colloms amplifier tests shows about 25% of his sample was ignored during analysis. That is, when listeners refused to make identifications, the only conclusion a researcher could make is that they couldn't hear any difference and so should have been scored a "same." When scored in this fashion, the Colloms results are less than 50% scored correctly. While both tests are of interest, they do not seem to provide contrary evidence.

In the April issue, Harvey Rosenberg refers to a "recent series of articles in the *JAES*" that contradict the Masters/Clark methods. I have been a subscriber to the *JAES* for several years and could recall no such series. I even reviewed the last four years' issues, and the Leventhal article was the only place where the topic was even discussed that could have been considered critical in any fashion. So I called Harvey, who was unable to even remember exactly when the series was published (about two years ago, he thought), who the author(s) was, or even what the conclusions were. Sounds to me that a footnote would have been appropriate here, don't you think?

I have been an enthusiastic reader of both *Stereophile* and *Stereo Review* for a number of years. I think they are both excellent, although each tends to work opposite sides of the street. However, fair and fair. If *Stereo Review* is ignoring significant other work as you suggest, it would seem that *Stereophile* is at least equally guilty of referring to a body of evidence that doesn't exist. Bias is bias, no matter who is biased. Maybe evidence does exist. If so, where is it? What have I missed?

Tom Nousaine
Chicago, IL

It was the Leventhal work on the problems of testing with an inadequate number of test subjects which was referred to in Vol.10 No.4, and, in my opinion, the Stereo Review *tests did fall foul of the problems raised by Mr. Leventhal.*

Martin Colloms responded to the criticisms

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of his single-blind AES-meeting amplifier tests in the August 1986 issue of HFN/RR. *When analysed on a more rigorous basis, the results still appear to support the conclusion that the amplifiers could be distinguished by ear.*

Regarding my absolute phase/capacitor tests, the key word used by Mr. Nousaine is "almost" when qualifying the phrase "those one would expect from chance." In the case of testing for the audibility of absolute phase and the insertion of a series electrolytic capacitor in the signal path, the identification was small, but statistically it was *significant (for all types of music with absolute phase, but mainly for music with a strong bass content with the electrolytic capacitor). The correct conclusion to be drawn from these tests was that people* could *detect the change in signal polarity and the insertion of the capacitor (the latter with the qualification above), but under blind conditions, the magnitude of the audible difference was small. The result of my experiment which* did *give a random result—* ie, *the results* were *the same as would be expected from chance—involved the insertion of a high-quality polypropylene-dielectric capacitor in the signal path. Taken as a whole, it appeared that the listeners could not distinguish the capacitor from a straight wire.*

The one regret I have about those tests was that with one piece of music, sacred vocal pieces by Hildegard of Bingen, there was *apparently strong identification of both capacitors, presumably due to the 10% phenomenon mentioned above by Mr. Davenport; but the number of tests failed the Leventhal criteria. The reason at the time for abandoning this very analytical music for test purposes was that it was not loud enough to compete with the noise next door—unfortunately typical of hi-fi show circumstances. Such are the problems of blind tests! I am thinking of repeating the whole series of tests at* Stereophile*'s Manhattan Hi-Fi Show in October. Please let me know if you would be interested in taking part.* **—JA**

Anomalous hearing?

Editor:

In a recent letter, I suggested that single listeners trying to judge the comparative merits of equipment while listening in different acoustic environments and using unique configurations of equipment may not serve the rest of us out here in consumerland. This is especially true if we are interested in listening to classical music, rather than joining the endless quest for the ideal reproduction system.

One of the things that seems clear, even from a cursory inspection of the scientific literature, is that there are many puzzles in making sense of experiments that try to use human judgments of hearing. I have been wondering whether the golden-eared population may not be on the far end of the human distribution—three standard deviations out—in the acuity of their hearing, and that the differences in judgments may not reflect unusually anomalous hearing characteristics. Here we are talking about three people out of a thousand, a proportion that, disregarding young children, leaves room for about 5,000,000 "high-end" enthusiasts in the United States. Assuming about half these might be sufficiently interested in listening to music, that still leaves quite a marketplace for those with such anomalous hearing. The rest of the population may indeed be unable to profit by high-end judgments; indeed, there may be no real significant differences in how reproduced music sounds to them played through different amplifiers, CD players, or LP cartridges mounted in different arms and on different tables.

Everett Wallace
Los Angeles, CA

No Smoking

Editor:

An error in Vol.9 No.2 (p.13) could cause damage to an unwary listener's hi-fi. The picture of a passive "preamplifier" (which would be used by an untrained constructor) has the volume control wiper wired to the balance pot rather than to the outputs.

If the volume were fully down coupled with balance fully towards one channel, this would place a direct short across the output of the source (and I assume that a smoking CD player would not pass anyone's criteria for *transparency*!).

If it is any consolation, I had a discussion with John Roberts of Phoenix Systems a few years ago in which he spoke of a similar error which had been published by *The Audio Amateur.*

B. Kendall Berg
Chandler, AZ

An open letter from Bruce Brisson

Dear Audiophile:

A decade ago, audio enthusiasts were connecting high-performance equipment with cheap patch cords and zip wire. Today audio cable has come of age as a true audio component, highly technical in nature and as critical to the enjoyment of music as any other component in a quality home audio system.

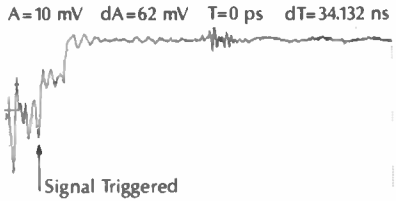
Appreciation and acceptance of superior audio cable did not happen by accident. The underground audio press, high-end dealers and audiophiles all realized they were hearing a whole new ballgame—a major step closer to the real thing! And since the early-1980s, MIT has been the acknowledged leader in developing high-performance audio cable, for the same reasons any product line emerges as #1—superior quality and performance, and customer enthusiasm and satisfaction.

This audible superiority of MIT cable derives from neither luck or magic. Rather it is based upon leading-edge computer design capability, rigorous testing and a unique understanding of the physics of cable performance. Scientific test procedures document the reasons why MIT cable products bring you closer to the live music experience. I would like to share these reasons with you, starting with the pivotal subject of "phase noise."

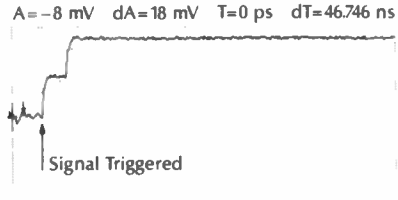
Phase noise is a degrading by-product of a larger cable problem known as "delay distortion." Instead of passing energy uniformly, non-neutral cable actually stores energy, subsequently releasing this energy—at the wrong time, and out-of-phase. Such cables have a large "settling time," which means that as a signal passes through these cables, substantial information is left behind.

This residue then piggy-backs onto the next signal, producing audible and out-of-phase sonic additions. This gremlin is phase noise, inevitable in any cable not correctly designed to fully pass all frequencies at exactly the same speed. Unlike other audio cable, MIT cable is designed to be truly phase coherent. Highs and lows travel through MIT cable at the same velocity, eliminating delay distortions and phase noise. Thus MIT cable is neutral—it does not alter the sound as do other cables.

A straight wire bypass is the generally accepted control when testing how other components change the sound (via substitution or insertion). And use of straight wire in audio cable is actually recommended in some quarters. So we decided to test MIT MI-330 head-to-head against straight wire, using 1 meter lengths (a complex tone consisting of a step pulse created by using a +.090v 40hz pulse delayed by 100ns, riding over a +.090v 40khz pulse*):



Straight wire:
measuring instantaneous phase noise



MI-330:
phase coherent and noise free

The test graphs demonstrate the phase noise superiority of MI-330. Using true 1.3ghz sampling and a 200ns time base for superior resolution, a phase noise residue of +.072v remains in straight wire *during what should be a quiet period (at dc).* For MI-330, this residual is only +.010v, yielding truer quiet passages. MI-330's defined waveform and quiet passages mean the music starts and stops only when it is supposed to, yielding a more holographic sound stage, intertransient silence and minimal distortion.

Cables with delay distortions may at first seem quicker, with more highs and ambience. Why? As discussed, these frequency dependent delays move energy from one location in a complex waveform to another. This changes the waveform or musical tone. Worse yet, moving energy around within the tone also results in noise, defocusing and removal of low-level detail.

Minimal phase noise is but one significant advantage of MIT cables. We will take up other key factors in future issues.

Good listening,

Bruce A. Brisson
President

*For a more technical discussion of delay distortion and phase noise, please write for data to Transparent Audio Marketing, P.O. Box 117, Route 202, Hollis, ME 04042

THE AUDIO CHEAPSKATE

Sam Tellig

Been to your favorite record store lately?

Every time I visit a Tower store in New York, I see more bins of silver discs, fewer bins of black. The other day, in Boston, at the downtown Harvard Co-op, I saw just a few bins of LPs. No classical LPs at all. The record department was 80% CD. According to *Billboard* magazine at the beginning of July, CD sales are now worth more in dollars than LP sales (both are below cassette sales).

The classical LP catalog is shrinking fast. Many items are no longer available. A number of reissues are coming out on CD and/or cassette only. Classical cutouts are becoming hard to find, and rumor is that Tower Records will close their Classical Annex cutout store in New York and replace it with a video store. Jazz cutouts, too, are becoming scarce. I think vinyl is fading faster than anyone would have believed a year ago.

What I'm saying, dear friends, is that you can't fight this even if you want to. Before long, your selection of all-analog LPs will be extremely limited. I wish people like David Chesky well. But Chesky Records and the like won't be issuing more than a handful of releases. If you want recordings, you'll have to buy them on CD (assuming DAT will be a dud).

How does the best CD compare with the best analog?

I still think analog is superior—in ways I'll attempt to describe in a moment. But CD sound quality is coming along rather nicely, thank you. Whether CD will entirely lose its characteristic coldness, sterility, stridency, lack of air, lack of inner detail, etc., is still an open question. I think it will, at least with the best recordings played on the best machines.

As LA or JA would tell you, I've been through sort of an emotional roller coaster with CD. Up on the medium one day—so convenient, no noise. Down on the medium the next—so lifeless, so sterile. But every time I think that CD will never do this or that, in comes a disc or a machine that proves me wrong.

For instance, I thought that CD was incapable of resolving very delicate low-level detail. Then I played a Philips recording of Beethoven's "Emperor" Piano Concerto with Claudio Arrau on an NEC CD 650E player, a cheapo unit that's been closed out recently at $299.95. Such detail! You could almost swear you were listening to a good moving-coil through a Klyne preamp. (Too bad the NEC machine is a wee bit strident. It's a bargain nonetheless.)

I was ready to state flat-out that CD was incapable of the same kind of dramatic soundstaging I have heard with cartridges like the Shinon Red or Koetsu Red Signature. Then along comes Mike Goldfield's Euphonic Technology mod of the Magnavox CDB 650 to prove me wrong. Soundstaging in spades, and dynamics, too.

I was telling LA that I found all CD players a little raunchy in the upper midrange. Then the Sonographe SD-1 proves that CD can sound very silky and smooth, if not particularly airy or detailed (more to come on that).

What I think is likely to happen in the next year or two is that machines will start to get it all together—exceptional performance in every respect. I hope so, because commercially it is all over for analog. And all over too, I suspect, for pricey turntables, arms, cartridges, and preamps, although that may take a while. Would you shell out $5000 for an Audio Research SP-11 if most of the discs you're buying are CD? (ARC is rumored to be working on a CD player.)

Meantime, what's an audiophile to do? Or, given the constituency of this column, what's a Cheapskate to do?

Well, one thing I would *not* do is spend big bucks on a player. I'd look to buy a decent-sounding player as cheaply as possible, as an interim step. If the player still works after a few years, I'd give it to the kids and buy something better—something that might stay state-of-the-art for more than a week, features a transport mechanism that isn't made of plastic, and a drawer that doesn't squeak. Mind you, I don't object to a plastic transport when it's in a good-sounding player I can buy for $200. But

I do object to all this plastic in a $1000 or $2000 machine.

If you buy an expensive player today, you can be absolutely certain that there will shortly be something better that you'll wish you had waited for. And ever try to sell one of yesterday's state-of-the-art CD players? The market is about as lively as that for secondhand Audi 5000s.

Here's what's worse about modified players. You can be almost assured that the mod will be continually improved; guys who modify players are tinkerers. So you'll probably want to keep getting your player "upgraded." Maybe you're best off not starting with the mods at all.

The Magnavox CDB 650/560

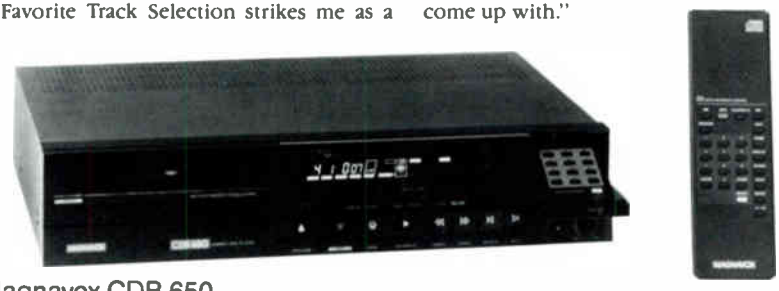
The Magnavox CDB 650 doesn't get discounted all that much. Maybe it's all those rave reviews. "State of the art," says the cover of *Audio* magazine. That's just great for sales. Everyone wants a Magnavox CDB 650, so all you usually get is $40 to $50 off. A CDB 650 is typically discounted to around $399. Meanwhile, the CDB 560, with remote control, but without Favorite Track Selection, typically sells for around $200. (The 560 also lacks a headphone jack and headphone volume control.)

Favorite Track Selection strikes me as a useless feature. But then I listen mainly to classical discs straight through. I do not want to omit the third movement of Brahms's Fourth Symphony forever, thank you. But FTS is useful with discs that have only one or two tracks: you can program your own favorite musical excerpts within or spanning tracks, using start and stop times.

My first exposure to a 16-bit Magnavox came several months ago, with the arrival of a CDB 650. I had expected to be blown away, but I was not. Perhaps it was the near-simultaneous arrival of the NEC 650E, but the Magnavox seemed a little lacking in resolution. Soundstaging was nice, though—wide and deep. And bass was much improved over the earlier 14-bit Magnavox models—deeper, tighter, more powerful. Not at all bad, considering the price. And for the price, I could forgive the flimsy plastic transport and drawer.

Later on, though, I got hold of a more recent Magnavox CDB 560 and found the resolution improved; these machines may have improved since they were first introduced. (Or perhaps it's normal sample-to-sample variation, and I got a particularly good-sounding 560.) The bottom line is this: the Magnavox CDB 560 stock machine is an astonishingly good deal for the money. The resolution could be better yet, and there is a slightly gritty quality to the treble. The improvement (over the initial 650 sample) came mainly in a sense of air—a very easy, open quality that I have not heard from CD before.

Len Feldman concluded his *Audio* magazine review of the CDB 650 by wondering whether small firms will start modifying the 16-bit Magnavox players. "Frankly," says Len, "I think if they do they may be wasting their time. I honestly can't see how they can improve on what the people in The Netherlands have come up with."



Magnavox CDB 650

Well, they have started kludging, of course, and have improved on the basic stock machine. The question is how much, and at what cost.

I managed to borrow a Euphonic Technology 650 over a weekend, and auditioned an Analog Design Group 460 and both a Mike Moffat 560 and 650. Of course, it's hard to say anything and be sure it will hold up, since these mods keep improving (as do, apparently, the stock Magnavox machines). The Euphonic Player improves on a stock machine principally in the areas of dynamics and soundstaging. The Analog Design Group player improves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most notably in the area of timbre—the player sounds so. . . tubelike. And the Moffat, so beloved of DO? Well, it has great dynamics and soundstaging, plus very good detail. The Moffat also has the best bass, by far, that I've heard from a CD player—due, no doubt, to its massive outboard power supply. (Moffat is said to be working on a less expensive mod, within the Magnavox chassis.)

Like all the other machines, the big Moffat still misses when compared to analog at its best. I don't get the same sense of acoustical environment that I get with the best analog—a feeling of the musicians', especially vocalists', palpable presence, and the separation of a vocalist from his/her acoustical environment. But CD has come along way, and you can enjoy most of the improvements in a stock machine.

Perhaps if Magnavox was to import the robustly constructed (made in Japan) CD960, the expensive custom mods would make more sense. But who knows—maybe the CD960 is so good, out of the box, that the mods would not be necessary.

The Sonographe SD-1

Sonographe (Conrad-Johnson) takes a lowly 14-bit Magnavox 2041, and turns this frog into something of a prince. Granted, its humble origins still show: the plastic chassis and the plastic transport mechanism, whose drawer (on my sample) opens and closes with the most annoying squeak. The basic Sonographe player lists for $695, plus $80 for a very awkward remote control (you have to use a pyramid-shaped box atop the player).

The Japanese would giggle at this thing, but I stopped laughing the moment I started listening. This is one of the smoothest-sounding CD players I have heard to date—bettered only by tubed players from California Audio Labs and others, which sell for much more money.

What has C-J done to make this player sound so good? Word is they have taken some of the FET technology used on their (excellent) Motif solid-state electronics and applied it to the analog output stage of this humble CD player. The real secret, I suspect, is that Messrs. Conrad and Johnson know what real music sounds like and have managed to kludge together something that produces a reasonable facsimile.

In the end, though, I think I would go with a stock 16-bit Magnavox—specifically with the remote-equipped CDB 560. The CDB 560 is an improvement over the Sonographe SD-1 in several areas. Resolution may not be one of them, but soundstaging is, along with bass response and a sense of air. The CDB 560 is also more extended in the highs; the Sonographe SD-1 sounds a little rolled-off, dark. A few months ago, I might have given the latter a rave review. The price is still reasonable, but the machine isn't a steal, like the CDB 560.

The CDB 560 is cheap and has remote control. It doesn't have Favorite Track Selection, the very existence of which I find annoying. True, the modified machines I heard all improve over the basic Magnavox. By contrast to the Moffat, for instance, a stock machine has a smaller, more compressed soundstage, less sense of air, less dramatic dynamics, less bass and sense of power, and finally a somewhat harder-sounding midrange and treble. (If you want a smooth midrange and treble, go with a tube player.) But we're talking about paying anywhere from three to ten times the price of a stock machine. That, as much as the performance of a stock machine, is why the Cheapskate can't recommend mods at the moment.

Sonographe SD-1

And now, as they say in the British hi-fi ads, "STOP PRESS!"

Sony CD-P505ESD

Just 48 hours before deadline I received the

Sony CD-P505 ESD

new $600 Sony CD-P505ESD, which uses the latest Philips 16-bit dual D/A converter, in conjunction with a Sony-designed 4x oversampling digital filter and multi-function VLSI chip.

A little background: Lots of people I've chatted with have been looking for a machine that works like a Sony but sounds like a Philips: Sony construction, convenience, ergonomics, and Philips sound—ideal combination.

I asked a modifier of Philips machines why he didn't do Frankensteinian brain transplants—Philips brain (DAC) into Sony body. Well, it's not so simple. The brain's wired into the rest of the nervous system. Those who mod Philips machines have been forced to take Philips's second-generation digital filter, which is supposedly good,[1] along with the Philips transport, which is not so good. Thus we have the spectacle of $2000-or-more CD players featuring the same flimsy transport found in $149.95 discount-house machines.

If you're Sony, though, you have resources. It's a snap to design your own digital filter and multi-function controller chip to operate with the Philips TDA1541 dual DAC IC. (Incidentally, the other new Sony ES machines—the CD-P605ESD and 705ESD, full review next month—use a 16-bit DAC said to be similar to the Philips chip used in the 505ESD, but not identical. All three new ES machines have 4x oversampling.)

Enough suspense! How does the Sony CD-P505ESD sound? Is it what many of us have been hoping for—a Sony that sounds like a Philips?

Yes! This, for me, is the machine that finally does it—gets it all together. Smooth sound—no digital edge. Ambience. Soundstaging. Dynamics. Fine resolution—something that still seems to elude Philips, based on the players I have heard. And all this straight off the shelf—no costly kludging.

The Sony has all the smoothness of the $695 Sonographe. It has the soundstaging and ambience of the 16-bit Philips machine, along with what seems to be slightly better resolution of low-level detail. I still think the NEC CD-650, now discontinued, has a slight edge, though. In more ways than one, unfortunately.

The CD-P505ESD is not perfect, but it is more or less right. It still misses that elusive "palpable presence," but the CD-P505ESD goes a long way toward recreating it. And, to be fair, the Sony player is far better than any $600 cartridge/tonearm/turntable/phonostage combination I can name. I can beat it, though, with a Shure V15 Type V MR/Rega RB300/AR ES-1/Superphon Revelation Basic Dual Mono combo (some mouthful), which would probably cost around $750 at actual street selling prices—superior soundstaging, ambience, resolution. . . that old "palpable presence" I keep raving about.

But the major labels, at least, ain't makin' new all-analog classical recordings any more. And jazz, too, is going digital.

So back to the Sony CD-P505ESD. Treble is unfatiguing. . . Hallelujah! Bass is extended and tight—almost as good as with some of the custom-modified Magnavoci. Sony seems to have done its homework with the power supply.

Lousy recordings (and there are many of those) still sound lousy, but at least they are sometimes listenable with the Sony, as they are not with many other machines. Good record-

1 It has been said that there is a wrong coefficient in the filter program, resulting in a loss of resolution at the -90dB level. Certainly, all the Philips 16-bit, 4x oversampling machines show about a -6dB level error at -90dB (see MC's review of the Mission PCM7000 in Vol.10 No.3), *ie*, the 16-bit digital words equivalent to a sinewave at -90dB are translated as a tone at -96dB. —JA

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ings (of which there are more and more) can sound great. But there's still room for improvement: even better resolution, and stronger dynamics. Most people, though, will think this is the best sound they've ever heard, and they'll be right . . . because they've never heard a Superphon preamp, Rega tonearm, etc., etc.

Operationally, there are no problems to report. No hitches or glitches, except that the drawer mechanism isn't as smooth as on some early Sonys. It's flimsier, but not as flimsy as the infamous Philips transport. (We have the weak dollar to thank for this—costs had to be cut somewhere.) The machine weighs in at just under 12 pounds. I use a VPI magic brick and a Mission Isoplat for support (perfect size).

Look, there's no point holding out any longer. Within a few months, there won't be many LPs left. Even the cutout bins are emptying out. If you haven't already done so, it's time to buy a CD player, and if you have $600 (I certainly wouldn't spend more, given the rapidly advancing technology), the Sony CD-P505ESD is it. If you don't have $600, I'd opt for one of the cheaper Magnavoci, or possibly a close-out NEC CD-650.

As for the kludges, *nyet*. Not any more. Just take the Sony off the shelf. Of the other Japanese players, the Denon is probably the brand most worthy of audition—I found that the DCD1500 and DCD1700 came very close to the new Sony machine in every respect; these, too, are 16-bit, 4x oversampling models. A good deal on a Denon might shift the balance that way. For me, though, at present (today, not necessarily tomorrow), it's the Sony CD-P505ESD.

A close-out surprise

Yesterday I received, by surprise, the finest phono preamp I have ever encountered. Better, yes, than the Klyne SK-5. Wouldn't you know—just when I felt I was content with the Quad 34. Damn! It's the $399 Superphon Revelation Basic Dual Mono Plus. The manufacturer refuses to make any more because: 1) his profit margin is too thin, and 2) dealers can't stand selling something so cheap.

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For further information, call or write: Sansui Electronics Corporation, PO Box 624, Lyndhurst, NJ 07071 (201) 460-9710.

VINTAGE by Sansui

a new and awful SILENCE

Bernard Holland[1] feels that serious music is losing its measure

The electric clocks in my house keep better time than the ones I wind, yet I scarcely look at them. It is the ticking, I think, that comforts me. I like to lean my ear against these various pendulums and, back and forth, gently rock my life away.

These ticks and tocks give me a meter to the passage of the day; they are a metaphor for silence. Silence, after all, is not an absence of noise but a subtle acknowledgment of this metronomic beat, the force that both brings new life and inscribes tomorrow's obituaries. There is luxury and terror in this act of resignation, this silent attention to the ticking of our lives.

Silence's most eloquent contradiction is music—not because music *breaks* silence with its sounds but because it interrupts its motion. All the arts do this: books freeze events between two covers, pictures pin them against a wall. But music goes viscerally to the source of our mortality. It stops time in its tracks and reinvents it. What a supernatural act it is to command a tempo and a rhythm, to set time in motion and bring it to a halt. In a life of temporal endlessness, the musician who makes time start and stop plays at being God. This is music's comfort and its triumph: that somewhere there exists an antidote for decay.

Music scarcely exists any more, having multiplied itself into silence. This probably makes no sense at all to you, but let me try to explain. To call music an interruption is also to say it is an event, something that can seize our attention only if it is preceded by uneventfulness, and then succeeded by it. When I first heard Bach's B-minor Mass some 35 years ago, that's how I experienced it—like a monolith rising out of an empty plain, a magisterial presence defined by the emptiness around it.

I fear I shall never have that sensation again. The plain is no longer empty. Developers have taken it over. On my FM radio the B-minor Mass is now but a commercial break away from the *Goldberg Variations* and the *St. Matthew Passion*. My days have become chains of such great events.

The technologies of the ear (the radio, the record player, the compact disc) both give and take away. How marvelous that Mozart's 27 piano concertos, Beethoven's nine sym-

1 Bernard Holland is a music critic for the *New York Times*.

phonies, and Bartok's six quartets are only fingertips away. And how horrible. It is a cruel trick that the wondrous accessibility of these great works has rendered them invisible.

We have, of course, only ourselves to blame. Science and the arts once met in a world of mutual congratulation. Stars moved to the music of the spheres; ancient musicians sang to the Pythagorean scales, serenely conscious of their geometric purity. But science is no longer as sure of its answers. Solutions retreat as we approach them. Thus we calculate our progress in degrees.

This calculation has made of us a society of measures—how tensile the steel, how quick the 100-meter dash, how slow the drip from the ketchup bottle. *How well* has given way to *how much, how many*.

So it should not surprise us that music has become quantitative too. One record on my shelf fills me with wisdom; three more records multiply it. One of my colleagues claims more than 50,000 items in his collection, and I know hardly any in my business with fewer than 10,000. Such a privilege—to have in one's home the capacity to hear Brahms's Fourth Symphony played 19 different ways!

In the South of my youth, where concerts were as rare as eclipses of the moon and Ernest Tubb ruled the airwaves, each new long-playing record was a discovery; the ecstasy was in that first moment, never to be relived. Listening machines and their paraphernalia advertise to us what they cannot fulfill—a reenactment of epiphanies. Each new Brahms Fourth promises such a rediscovery. Will Toscanini via RCA, or Bruno Walter according to CBS, bring us close enough to touch that first thrill again? Some avenues bring us nearer than others, but none near enough. We measure these nearnesses one against the other, and ask: who shall be first among Brahms Fourths?

Fishing in my pitiful collection of records the other day (my filing system is of the I Ching persuasion), I did find a recording of Brahms's Fourth. I put it back on the shelf, I'm not quite sure where. I am not worried. I know how it goes—the key of E minor, two beats to the measure, upbeat swoops down, upbeat swoops up, very beautiful in its austere way. This is how I relive (and therefore live) this music—in my imagination.

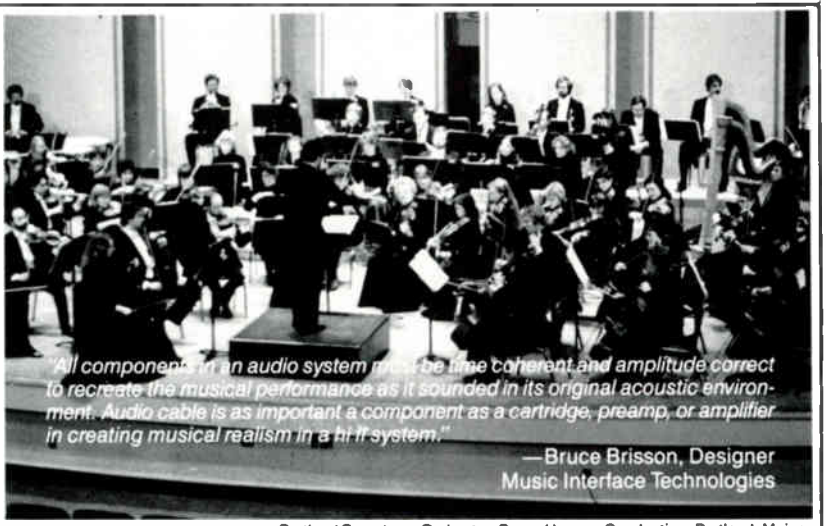
The imagination is our great healer. It is also the world's leading interpreter of the Brahms Fourth Symphony. The 19 performances that other critics have at their fingertips seem puny beside it. And my imagination does other things—it calms the wow in my turntable, smooths the distortion of my woofer.

The walls of records in my colleagues' homes, arranged, catalogued, cross-catalogued; the giant loudspeakers the size of refrigerators looming in their living rooms—all help explain to me why musical masterpieces no longer move me as they once did. The technological prowess of these new instruments of music is amazing. But they cost dearly: they usurp our capacity to dream. As a young man, I cherished the B-minor Mass, the *Quartet for the End of Time*, the *Symphony of Psalms* principally as voices speaking within me. On my shelf today, they are measured like real estate, by the frontage foot.

The ear plots its escape, but fails. Even the streets are not safe, and subway platforms ring with Bach's solo violin sonatas or the Spanish guitar. New England villages worthy of two gas pumps now add summer festivals to their inventories. One approaches their outskirts apprehensive, car windows rolled shut. In restaurants, Mozart serves as aural garnish for the fish of the day.

We are strangled by the very volume of our resources, dwarfed by them too. I think back to my friend with his claim to 50,000 records. How small he seems beside them, like a computer scientist facing an immensely potent machine that he does not quite know how to address. Modern science instructs us—shames us with the fact—that the universe has become very big and we very small. Beethoven, you will remember, promised us to take fate by the throat; one wonders if its size today might not exceed his grasp.

Music, indeed, seems to have become that popular science-fiction nightmare—the man-made creature that grows beyond expectations, seizes autonomy, and smothers its masters. The quartets, the masses, the songs and sonatas that once rose as isolated protests against the vastness of time have become something very different. They have swollen, then merged and melted into time's fabric. They have become a form of silence themselves. S



Portland Symphony Orchestra, Bruce Hangen Conducting, Portland, Maine.

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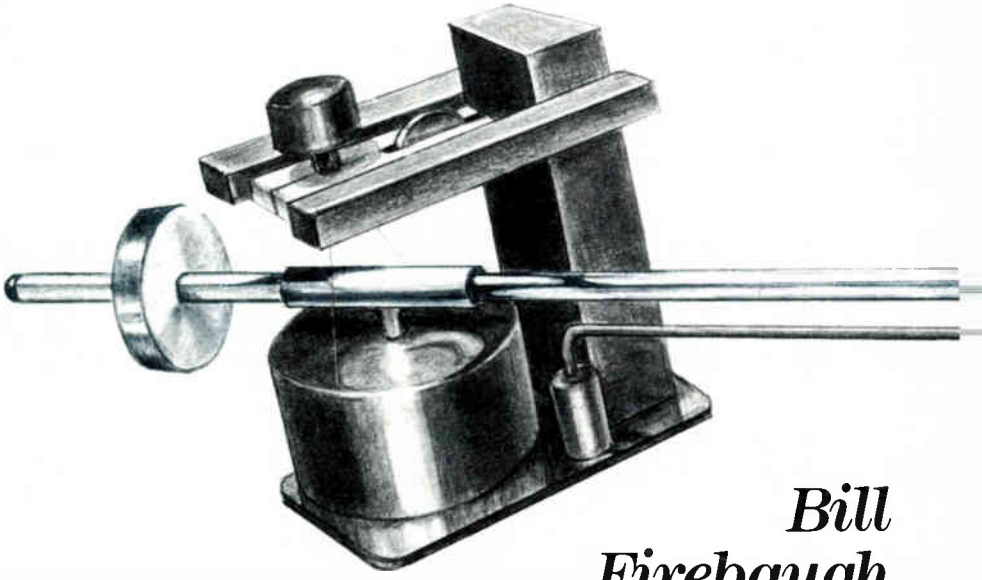


Bill Firebaugh talks with J. Gordon Holt

Bill Firebaugh's first product, the outrageous-looking Well-Tempered (tone) Arm, established him as one of audio's most innovative designers. At the 1985 Winter CES, he showed a prototype companion product—the Well-Tempered Turntable—and was producing production units by January 1987. He discusses here the WTT's unusual design features. (Readers should note that, since we have not yet tested the new turntable, this interview is not to be interpreted as an endorsement of the product.)

JGH: Bill, what prompted you to get into the turntable business?

BF: Well, after I had made my arm, and achieved what I felt to be the correct degree of stability in it, I thought it would be a rather simple matter to make a turntable. I thought, "There's nothing to a turntable. It's a platter, a bearing, a belt, a motor, and so forth." To me, it seemed to be the simplest thing in the world to make them all work together. So I started out using an aluminum platter, and I made a very nice spindle sleeve and ball-type bearing.

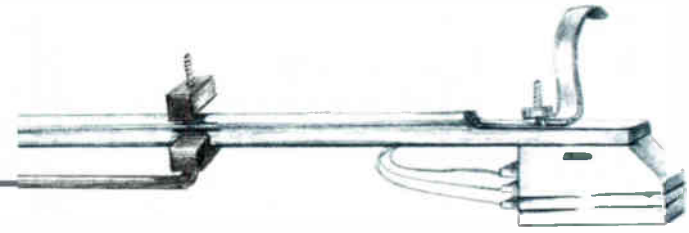
JGH: In other words, the usual thing.

BF: The usual thing. It took hardly any time at all to make a prototype, but it didn't work well at all. It didn't have that nice, sweet, musical sound, and I wondered, "What's going on here anyway?" After a period of consternation, I started experimenting with these sleeve-spindle, ball-type bearings, and I made maybe three dozen different varieties, trying different materials and different finishes and geometries.

JGH: Why did you assume right off that it was the spindle bearing that was at fault?

BF: Because if I turned the bearing by hand and listened to it, I could hear it go "scrape, scrape, scrape" as I rotated it. It didn't sound smooth. I could feel the roughness when I turned it. So I made lots of different versions of that conventional bearing arrangement, including ones with standpipes of lubrication. I came to understand that one of the really important issues is lubrication. If you have a spindle fitting into a sleeve and there's only a fraction of a thousandth of clearance between them, there's hardly any room for lubricant to get in there. And any small speck of stuff that gets in there really causes trouble.

So I made quite a few prototypes that had reservoirs of oil; I would drill a hole through the side of the sleeve and connect it with a reservoir of oil so it could get to the spindle



The Well-Tempered INNOVATOR

through the hole. And then I added more holes to get the oil to the spindle because it still sounded kind of dry.

Then it occurred to me that, if you have enough clearance around a normal spindle that it doesn't bind, and then you have a belt pulling it to one side, it's going to tend to walk around the top of the spindle hole until the belt tension pulls it back, then it's going to go "Clunk!" and do the same thing again. And you *know* that's not going to help the sound. I believe that's what makes belt tensions and all those things so critical on a normal type of bearing. But I was stuck. I couldn't seem to find a way around the problem.

JGH: How did you solve it?

BF: I just waited. And while I waited, I kept kind of fiddle-diddling, and eventually new ideas came.

I started thinking about what it would take to provide stable contact between the bearing and its housing without confining the spindle so it would jam up, and I thought of a round peg in a square hole. You pull the platter toward one corner of that square hole, the way belt tension will pull it, and the spindle rests firmly against the edges of the hole at two points 90° apart. And the bottom of the spindle, of course, leans up against the other two edges of the square hole. The spindle can't wobble, yet it isn't confined to a limiting space at either its top or bottom. So I made up a square spindle-well out of nylon and it worked: there was no play at all in the spindle. I thought I had discovered this radical way of using, for the first time, the old round-peg-in-a-square-hole idea.

But then, one Saturday night—actually it was a Sunday morning, I'm a night-owl type—I thought of a simpler way of doing the same thing. If the well is only contacting the spindle at four points, why not just do away with everything except those four points? I thought about using a big spindle sleeve, with an eighth of an inch clearance between the spindle and the sleeve, and then putting in four supports that come through the wall of the sleeve, so they only contact the spindle at a couple of small points.

I had some nylon setscrews on hand, so I threaded those and put them in from outside the spindle well at right angles to each other, so the top of the spindle was seated between two screws at the top and two at the bottom. It worked, so I knew I was headed in the right direction. But there still seemed to be too

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The Well-Tempered Turntable: note half-twist in belt

much friction. There was lubricant in there, but the bearing still wasn't as smooth and easy-turning as it should have been.

So I tried Teflon and polyethylene "screws" instead of nylon, but they still seemed to be binding too much. In addition, they were greatly accentuating the surface quality of the spindle. I wanted something nice and soft and smooth that the spindle could ride against, so I tried using a sort of a setscrew with a rubber point on it. I took a metal setscrew and drilled a hole in the end, and I cut off a piece of an O-ring of the same size and stuffed it into the hole. And that was it! Boy, that was so silky and smooth! It made an immediate difference that you could hear.

I eventually wound up using metal setscrews with the end ground off flat and a 3/16" flat disc of nitrile rubber bonded to it. Nitrile rubber is the stuff oil seals for car engines are made of. Working against polished steel, those seals will last for a hundred thousand miles.

JGH: But wouldn't rubber have higher friction than Teflon against the spindle?

BF: By itself, yes. But there's maybe a shotglass full of silicone lube—Dow-Corning 200 silicone fluid—in the spindle well. With silicone fluid, this nitrile rubber against metal is extremely slippery. And furthermore, there's a fair degree of damping in the bearing, simply by virtue of the fact that you have five of these pads riding against the spindle. And it's not only damping the rotation, it's damping any vibrations that might be excited within the shaft itself.

JGH: You use one of them at the bottom, to support the platter?

BF: Yes. And the bottom one is off-center! If I centered it, it just squeezed all its lubricant out and ran dry. Putting it off-center means the part of the spindle surface that isn't on top of the rubber disc at any given moment picks up a fresh coating of oil. The lubricant is constantly being drawn into the surface between the disc and the flattened end of the spindle.

JGH: I noticed your turntable uses the drive belt flipped, so that it runs upside-down around the motor pulley. What's that suppose to accomplish?

BF: It seems to have a significant effect on flutter. I spent a lot of time studying flutter, both on a 'scope and on strip-chart recordings. I

was trying different belts, and I suddenly noticed one of them was giving lower flutter than it had previously. And then I found it was because I had put it on with that half twist in it. It was a lucky accident.

I'm not sure why it reduces flutter, but I think the reason is, as the belt comes off the platter, instead of losing contact the same distance from the motor pulley, the top of the belt separates before the bottom of it.

JGH: You're attributing the flutter to vibration of the belt?

BF: That's probably it. See, the frequency at which the belt will vibrate is related to the length of it that's in free air—not contacting either the platter or the pulley. Both of those will prevent it from vibrating. With the usual belt threading, the top and bottom of the belt leave those surfaces at the same time, so the lengths of free belt are the same at the top and bottom of the belt. The whole free length of belt will vibrate at the same frequency, giving a fairly high-Q resonance. But if we put a half twist in the belt, its top and bottom edges leave the platter and pulley at slightly different places, so the belt resonance is more distributed in frequency. It has a lower Q, so vibration at any single frequency is much reduced.

I don't know if I could assign a magnitude to the improvement, but you can sure see it when you're making flutter tests. And you can hear it. A friend of mine who has a home-built, very massive turntable like a Micro Seiki, uses a very stiff belt with it, made out of rubber-impregnated Kevlar. He tried flipping it over and he told me he very definitely notices a difference. He always uses it twisted now.

It was while I was experimenting with belts that I discovered something amazing. The belt goes slower than the pulley. And it's not because the belt is slipping, it's more like it's crawling. Here's what happens, I believe. On one side of the pulley the belt is tight, because it's having to overcome platter pivot friction, while on the other side it's loose, because it's just taking up the slack. And in between, around the pulley, those tension differences have to be adjusting to one another. So there's a sort of a wave of suddenly-changing tension crawling backward all the time through the pulley-contact area.

JGH: Wouldn't there have to be *some* slippage through this area?

BF: Sure, but see, the slippage is in one direction at one end of the belt wraparound, and it's in the *opposite* direction at the other. You'd expect them to cancel, except that the area where the belt is crawling is continually feeding part of the pulley-contact surface backward toward the tensioned part of the belt.

It's hard to visualize. And it took me a long time to figure it out. But what it does is, if you try and calculate the platter speed from the pulley speed and the two circumferences, it doesn't work out right. The platter doesn't run at the predicted speed; it will always be slow. I'll tell you, this drove me nuts until I found out what was going on. I wasn't the first person to discover this by any means. I was explaining my speed problems to my belt manufacturer, and he knew all about it. He confirmed that this is a known phenomenon in belt-drive systems, especially ones with relatively compliant belts. The pulley goes faster than the belt. But I had to learn it the hard way. I ended up having to correct the size of the pulley empirically to get the right speed.

JGH: Why does your turntable weigh 50 pounds?

BF: It weighs about 42 pounds. The base material is Medite, which is a very dense, high-quality fibrous board. It's resin based, like chipboard, but denser. And perhaps you know that I was using sand for a while to damp it further. In the first prototypes.

JGH: So I understood. But you aren't any more?

BF: Oh no, it was a menace! Sand is marvelously inert; it's really great stuff for damping turntable bases. And tonearm tubes—I pack the tube of the WTA with it—but it was no good in the turntable. It made the base very inert, but when I brought my prototype WTT to Las Vegas in January 1985, it was so dry there that my demo records built up a fierce electrostatic charge, and some of those grains of sand got stuck to them and ruined them. They all ended up with scratches. So that was the end of the sand.

I wound up with a Medite sandwich—three plates of Medite fastened together with sound-absorbent stuff. It looks like double-backed adhesive tape, but it's very sound-deadening. It's made by 3M. They've got a huge catalog of double-backed stuff with all sorts of different properties, made out of lots of different materials with different densities and thicknesses and compliances and so forth. You wouldn't believe the variety of stuff they make!

JGH: What about acoustical isolation?

BF: I don't attempt to isolate the platter from the base, but the whole turntable is suspended on four fairly stiff feet, whose compliance is such that the mass of the table provides isolation from the underlying surface from about 15Hz upwards.

JGH: Are the feet spring-loaded?

BF: No. I tried using springs, but I never could get the result that I wanted. It was always too jiggly and unstable, and intuitively, I know that if you touch something, and it jiggles up and down for several seconds, that's going to give you instability problems when you're playing a record.

JGH: Did you try damped springs?

BF: Yes, but the damping provisions were either ineffectual or prohibitively costly. My whole experience with springs soured me completely on them. By solving problem A they just introduced problems B, C, D, E, and F. So I gave up on springs.

JGH: Your 'table's suspension feels very stiff. Is that the way it's supposed to be?

BF: Yes. And that's the kind of compliance you need to isolate a mass of 42 lbs at frequencies above 15Hz. While 15Hz may seem kind of high, the type of stability it gives you makes the player very immune to such things as footballs and so forth. In my home, I have the record player sitting on a dining-room buffet in which there are many drawers and doors which I use to store audio junk in—old cartridges, arms, record-cleaning brushes, alignment protractors, all sorts of the things you accumulate through the years. I can open and close those drawers and doors without in any way disturbing the playing of a record.

JGH: You said you started out with "the usual" aluminum platter. Is that what you ended up with?

BF: Oh, no. Like everything else I started out with, it went through a lot of experimentation. The aluminum platter rang like a bell. It was very poorly damped. No, the final version uses an acrylic platter—Plexiglas. It's highly internally damped. It weighs about 3 lbs, and there's a thin rubber mat on it.

JGH: That's pretty light. Maybe that's why belt vibrations were affecting the sound.

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BF: Maybe, but twisting the belt helped that very heavy platter my friend had on his home-brew 'table.

JGH: What about the motor suspension? That looked pretty unusual too.

BF: I went around the world on motors, and tried lots of different schemes for mounting the motor on the same assembly as the platter and then isolating it with springs. You know, that's the usual technique. And I just couldn't get it to work. I couldn't get the motor vibrations down to a level that I wanted, so I got kind of torqued off one night because I was at the end of my rope, and I took some lead—scuba divers' weights—and . . .

JGH: Which, I suppose, you just happened to have lying around!

BF: No, I *do* scuba diving. But I did happen to have a Coleman stove and a big pot, and I melted some lead and poured up a brick of it. I clamped the motor to that lead brick and said, "Okay, motor, if you want to vibrate, fine, you do that, but you're gonna have to vibrate this lead brick along with you." And it just completely eliminated all the motor vibrations.

JGH: Just by adding mass!

BF: Just by adding a *lot* of mass.

JGH: How big is this "brick"?

BF: It's four and a half inches square. I've forgotten how thick it is; an inch and a quarter, maybe.

JGH: How is it attached to the motor?

BF: It's clamped against the bottom of it. There's a metal plate at the top, with the spindle sticking through it, and then there are four long screws that pass through the plate and through the lead brick at the bottom. The motor's captured between those two.

JGH: How is the motor fastened to the turntable unit?

BF: It isn't. It's a completely separate assembly. But it doesn't stick out like an outrigger; it sets into a compartment at the left-hand side of the base. The motor stands up on four stiff rubber feet attached to the underside of the brick. You can't have the feet too compliant, because the whole motor will tend to rotate.

JGH: Was that your finished turntable?

BF: Not quite. The object of all this was to try to get the flutter and wow down to low levels, and I eventually wound up fluid-damping the motor. See, no motor manufacturer I contacted would guarantee me any rotational stability spec. You buy a 300-rpm motor, they guarantee it goes around 300 times per minute and that's all. But its speed during each revolution? That's completely uncontrolled. The manufacturers assume inertia will take care of that, but it doesn't completely.

JGH: Because the motor normally would have a tendency to cog. It *is* a hysteresis synchronous type, isn't it?

BF: Yes, it is. So I asked myself, "What would happen if I just put a certain kind of grease in there?" I happened to have some saltwater wheelbearing grease, which is quite heavy and doesn't get runny when it gets hot. And I squirted that in there, and gosh, that worked great. I mean, it made a difference you could really hear.

JGH: Doesn't it creep up the sides when it's spinning?

BF: Well, I had to experiment a lot with the viscosity of that grease, but, no, it doesn't. It's too thick.

JGH: Then it wouldn't leak out if someone shipped the turntable back for repair.

BF: No. The motor grease isn't thin enough to flow. The platter lubricant does, but in that case, the bearing just comes right out and you could just pour that out. It's a completely nontoxic silicone fluid.

JGH: And now that we've found out how to clean it off things, spillage is no problem. I suppose you know that lighter fluid will do it, too.

BF: Yeah. Gasoline is one of the better solvents for silicone fluid. The Dow-Corning brochure for their silicone grease lists all the solvents for it. Rubbing alcohol is one of them. It isn't the most effective solvent, but it's very safe, and that's what I always recommend. Lighter fluid of course is gasoline, and that can be dangerous.

JGH: Yeah, but if you use it very sparingly, like on the end of a Q-tip, it would hardly be hazardous. With lighter fluid, you just squeeze it out of the dispenser onto the Q-tip. The worst you can do is set the Q-tip alight, and it would be hard to do that unintentionally.

BF: Trichloroethane works really well too, and is commonly available. It's used in the electronic industry for cleaning circuit boards. And it's the same kind of stuff Kodak sells as film cleaner for slides and home movies. Camera stores sell it.

Actually, the turntable is much easier to prepare for shipping than the tonearm, because

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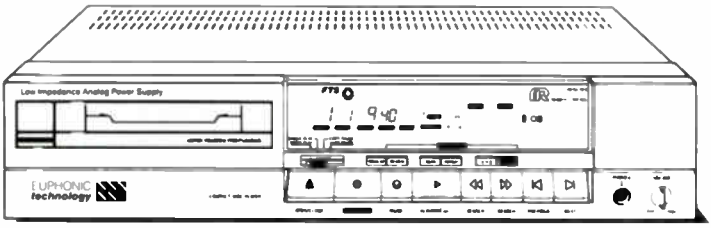

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all you have to do is pour the fluid out of the platter spindle into a Ziploc bag, which you hang on to. Then ship the 'table back.

JGH: What *about* shipping your tonearm? A number of people have asked me how they could prevent that sticky goo from getting all over everything when they ship a viscous-damped arm.

BF: Well, of course, you don't ship the viscous damping goo. To ship my arm, first you cut the strings, then you remove the damping paddle, then . . .

JGH: Whoa, now! You cut the pivot strings?

BF: Yeah, just cut 'em. They're very easy to restring. At least by anybody with some manual dexterity. Then you pry the damping cup off . . .

JGH: *Pry* the cup off?

BF: Yeah. It's held on with double-backed tape. The secret to separating something held on with that tape is a slow, steady pressure. You put a prying instrument under it and apply slow, steady pressure, and it separates. But if you try to do it fast, boy, that stuff sticks like crazy!

JGH: So you dump the goo into a Ziploc bag . . .

BF: Yes. Scrape off as much as you can from the paddle and well, let them stand overnight over the bag to drain off the rest, then wrap them in grease paper and pack them with the rest of the arm for shipping. Keep the goo until the arm comes back.

JGH: Bill, going back in time a little bit, whatever gave you the idea originally that you could design a tonearm that would be accepted by the high-end audiophile community?

BF: I didn't plan to sell it. Originally, I just thought I'd try making an arm for myself. Sort of as an intellectual exercise. At the time, I had not the slightest idea of what was going on in a tonearm. But in 1977 a guy named Poul Ladegaard from the Bruel & Kjaer Company in Denmark wrote what I think is a landmark article entitled "The Audible Effects of Mechanical Resonances in Turntables." B&K had this fabulous instrumentation system using Fast Fourier Transform analysis and parallel processing and all that kind of stuff. Actually, I think the main reason they published the paper was to show off the capability of their test instruments, but they were obviously also interested in doing basic research about audio.

I read that paper, and I had the feeling I thoroughly understood what the fundamental issue in a tonearm and a turntable is—stability. It started me wondering about the stability of the arm I was using then.

JGH: Which was?

BF: I had one of the original AR 'tables, which I had bought back in the dark ages. So I hooked up a high-gain wideband amplifier to my 'scope, and tried to figure out a way of measuring how stable the arm was.

B&K had used impulse testing, but I had to figure out what to use to induce impulses to the arm. The technique I eventually came up with is to straighten out a paper clip and use it to tap on the arm at various places.

JGH: That's a technique I've been using for years, except that I listen to what happens instead of watching it on a 'scope! Put the stylus in a stationary groove and tap on the arm. A poor one will go CLICK; a good one will go CLUNK.

BF: I used it because it's very easy to generate a pulse that way, and it's very easy to analyze the output and to understand what you've got. I was tapping on those things and looking at waveforms and getting what I thought were pretty good results, but the sound . . .Well, that needed work.

It was very clear to me that the AR arm had a lot of friction in its pivots. You could see it in the erratic tracking-force measurements you got, and the way the stylus deflected from side to side when tracking an off-center disc. So I thought, "What kind of pivot might give the least amount of friction possible?" Then I remembered what I had read about Henry Cavendish, the 18th-century English physicist, and his experiments into the nature of gravity.

Physicists have known for hundreds of years that the lowest-friction "pivot" you could get is a single long cord fastened to the ceiling. Cavendish used this arrangement to test for gravitational attraction between a suspended bar and stationary objects of known mass, and his results enabled him to determine, for the first time, the mass of the Earth. And Gauss used this type of device when he was investigating very weak magnetic phenomena. I just used the same idea to make a frictionless pivot for my AR tonearm.

JGH: You used a single thread?

BF: I started out that way, using the AR arm,

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but it was a complete disaster, of course. It flopped all over the place. It wasn't too bad in the vertical plane because the stylus is resting on the record, so it can't go very far up or down. But torsional movement is completely uncontrolled. It was obvious that I would have to use two cords to prevent the arm from twisting. So I drilled a hole in the arm and fastened a rod across it, and attached the cords to the ends of the rod. That was an improvement, but that thing was still tremendously unstable. That was when I started thinking about viscous damping.

Maybe I've told you that when I was working on this tonearm, I made 81 different very distinct prototypes of it. And one of the early ones had a little tab sticking down in a bowl of STP, the oil additive. I used that because it had fairly high viscosity and I could buy a can of it locally for 50 cents. That did make a noticeable improvement in the sound, but it still had a long way to go. By using these impulse techniques, I tried different amounts of damping. I couldn't change the STP's viscosity, but I could change the damping coefficient by using different-sized paddles.

The more I got into it, the more problems I found. According to my pulse tests, I should have been getting pretty good sound, but I wasn't. Even designs that looked as if they were behaving themselves *sounded* bad. I had to start questioning my instrumentation.

Then I made a rather important discovery. I had been overlooking an important characteristic of magnetic cartridges: They're rate sensors. That is, you get so many mV of output per cm per second of stylus motion.

JGH: Yeah, they're velocity-responsive.

BF: They're not amplitude-sensitive. So you can fool yourself into thinking you have a stable system by observing the output from a cartridge, because all the worst instabilities are at low frequencies, where stylus velocities are low. What you have to do is to transform that velocity signal into an amplitude signal. And that's done electronically by what's called an integrator.

JGH: Which is essentially a device that introduces a 6dB/octave treble rolloff across the whole audio band. An equalizer.

BF: Yeah. You need something that will double the voltage output each time the frequency is halved. So I fed the cartridge through an integrator and gave the arm a tap, and man, that thing jumped all over creation. It was tremendously unstable. And, of course, that causes scrubbing.

JGH: From the vertical tracking angle of the stylus.

BF: Laterally, too. It just happens at a different place.[1] And scrubbing is the fundamental evil when an arm is unstable, because it doesn't generate orderly harmonics. It generates nasty little sidebands which aren't harmonically related to the signal at all. The unstable motions of the arm may be subsonic in frequency, so you can't hear them directly, but they powerfully affect the rest of the sound.

The problem with using an oscilloscope was that, while it showed me that I had problems with resonances, it didn't tell me much about *where* those resonances were. So I bought a computer with a Fast Fourier Transform program, and all of a sudden I could *see* all those sidebands.

JGH: What you see is essentially all the release of the stored energy.

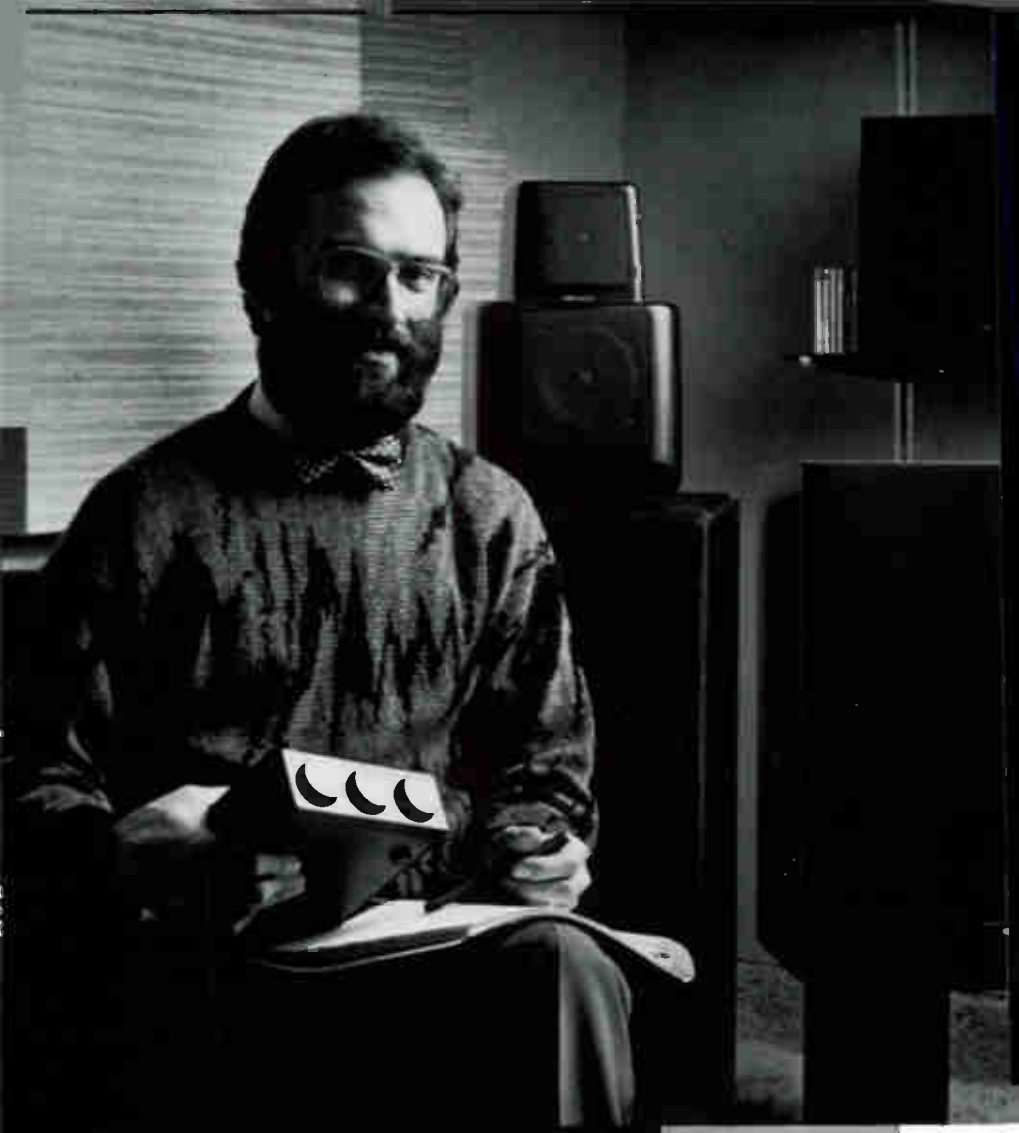
BF: Yes. An impulse contains all frequencies, and its duration is so short in comparison with the time it takes for the system to release the energy from the pulse that it stores, that all you read is the system's response. That's how the FFT works. It allows you to suppress the input pulse so all you read is what happens *after* the pulse. It's almost like having an output signal with no input.

Once I saw how unstable the arm was, even with that modicum of damping, I knew that there wasn't enough of it. I knew I would have to use a fluid with better viscosity characteristics. The thing about tonearm damping is that you want a lot of friction when the arm is moving fast and very little when it is moving slowly. And when it is hardly moving at all, you want as little friction as possible. That's the trouble with mechanical damping—pure friction. You get the same amount of friction at all motional speeds. The proper damping fluid cleared up most of the arm's misbehavior.

JGH: You haven't mentioned the sand.

---

1 Scrubbing occurs when LF motions of the headshell move the armature pivot away from its normal point relative to the average groove path, effectively changing the length of the armature and thus varying the speed with which its tip scans groove modulations. For a given amplitude of headshell vibration, vertical scrubbing produces much more flutter than does horizontal scrubbing, because the armature's at-rest position is at a 15-degree angle to the groove path.



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BF: Even after I got the damping coefficient correct, as evidenced by these impulses, piano and female voices still sounded colored. My pulse tests indicated that it was the arm tube itself that was taking off and resonating. It, too, needed to be damped.

I use a thin-walled stainless-steel tube for the arm. I tried filling the arm with various things, including Dacron yarn. I tried expanding foams and liquids. Liquid made it much worse. One morning while I was having breakfast, I noticed the salt shaker and thought, "Hey, salt! How about salt?" And man, I zoomed out to the garage and took one of the arms apart and filled that thing with salt, and when I listened to it, I knew I was on to something.

I also knew I couldn't *use* salt, because it's highly corrosive and in a humid climate it would absorb water and turn to liquid. That was when I thought of sand.

I bought some of the "play" variety that's processed for use in kiddie sand boxes. It's very clean, sterile, beautiful stuff, dust-free and all that. It really deadened the arm, and that's what I now use to fill the arm. And when we applied for the patent on the arm, we threw that in too, and sure enough, I got the patent on sand-filled tonearms.

JGH: Is the sand loose in there or is it held by some kind of binder?

BF: No, it's in there loose, but it's packed pretty tight. It's kept in with a cotton plug.

JGH: You haven't mentioned those incredible cartridge-connector clips that you use. I don't believe any other tonearm manufacturer has ever thought to look around for a connecting clip that would adapt to different-sized cartridge pins.

BF: I had a heck of a time with that. All the available cartridge clips seemed to be too darned stiff, and they had to be adjusted to fit different cartridges. I thought they would be uniform, but they varied all over the place. And adjusting those conventional clips to fit is a terrible job!

JGH: So tell me about it! I've been complaining about this in *Stereophile* for years.

BF: I started looking through all the surplus stores within a wide radius of my house, in their Pins and Connectors departments. I came across some things made by Amp. They're nifty little gold-plated things, designed to accept pins from integrated circuits, or to attach to wire-wrap pins on a circuit board. I tried some of these, and they were perfect. This particular clip fits snugly on a wide range of pin sizes without any adjustment.

Those things cost 20 cents each, though! I ordered 1000 of them, and when I went to pick them up, those guys at the store were laughing. Two hundred bucks these cost me, and I get a little plastic bag with maybe a tablespoon of them in there!

JGH: Bill, what's your background in all this? Were you schooled as a mechanical engineer?

BF: Well, I'm an engineer for Ford Aerospace, and have been for about 25 years. I have a degree in physics, and I work as an engineer in optics and electronics. I'm a laboratory-type engineer; I'm not the type that wants to be in administration. I like interesting engineering work.

JGH: And you do all of this audio designing in your spare time?

BF: Yes. I'm hanging on, but barely. My life will soon be undergoing a tremendous upheaval though, because I have to turn out 1000 turntables PDQ. And that's going to eat up a lot of my time.

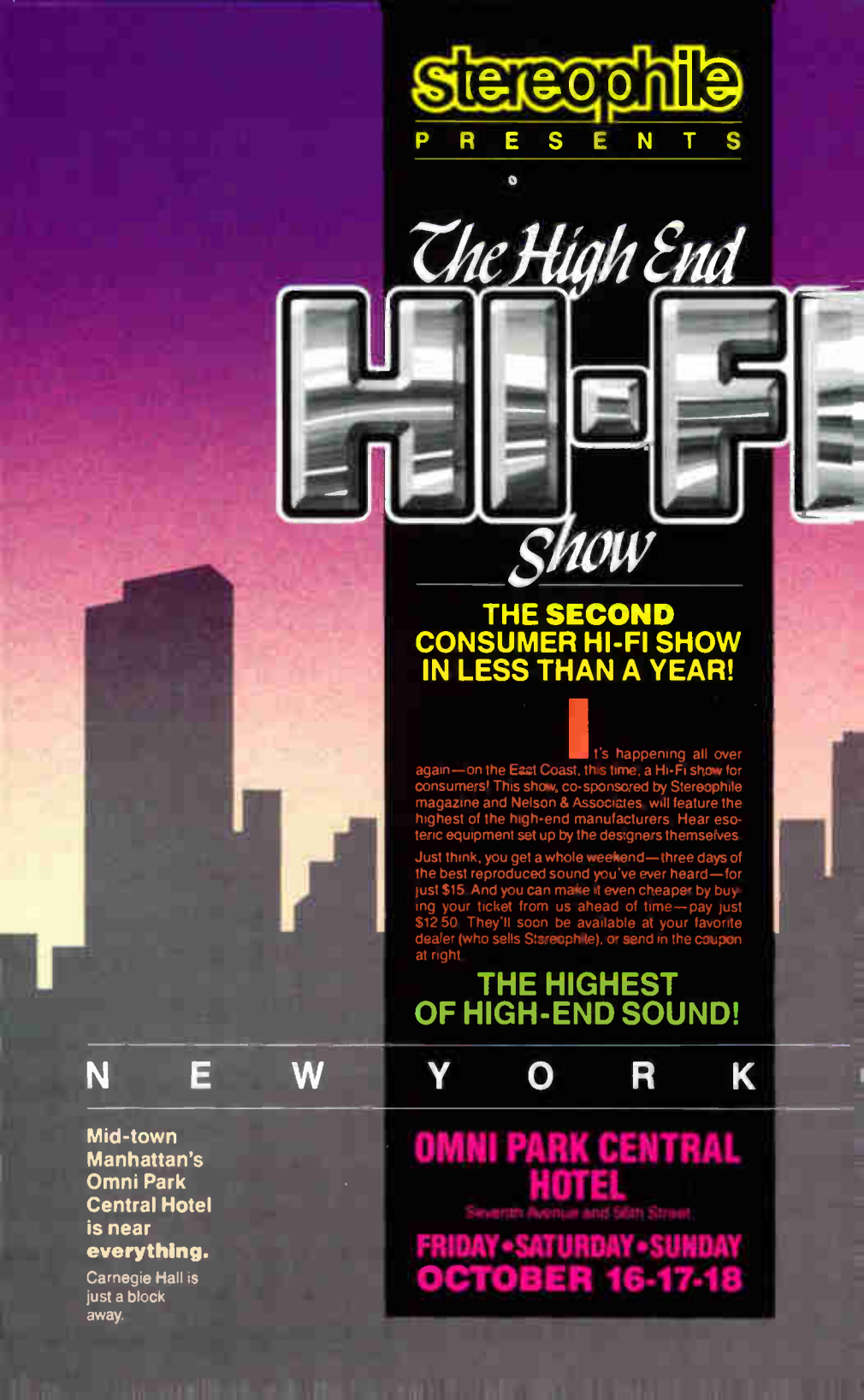
JGH: Somewhere along the line you're going to have to make a momentous decision: to curtail your audio activities or quit your job with Ford Aerospace.

BF: I am. When I turn 55 in just a few years, I'll probably retire from Ford Aerospace. But right now, I'm going to have to hire some helpers, and believe me I'm not exactly looking forward to that, because production is nothing but a headache. However, there may be a rather nice financial payoff, which can tend to diminish one's reluctance.

But the thing is, right now the quality of my life has taken a step backward. I don't have much chance to go scuba diving or to listen to music any more, and I enjoy both.

When I'm 55, I will have 15 years before I'm 70. I figure that's the amount of time left during which I'll have some degree of steam. And during that time there are a lot of things I want to do. Basically I want to enjoy myself.

Yes, you're right; I've got some fundamental decisions to make. How am I going to spend those 15 years? You know, the object of being down here is to enjoy yourself. If you're not enjoying yourself there's definitely something wrong. **S**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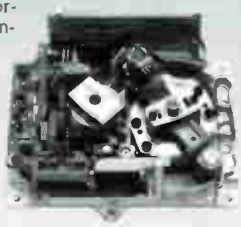

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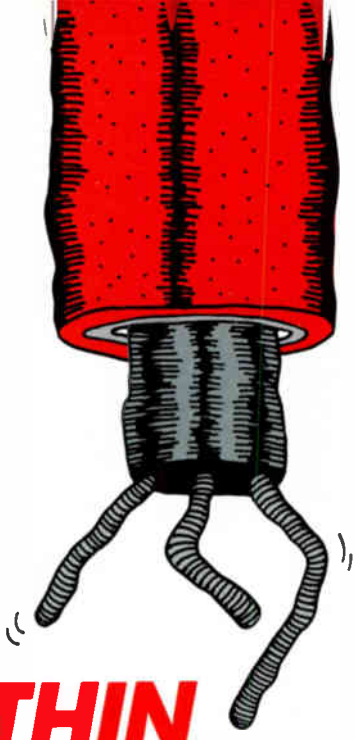

Illustration by Michael Motley

THIN SOLID-CORE CABLE: BOON OR BUST?

Dick Olsher

In his "Pure Gold" column in Vol.10 No.4, Alvin Gold fervently promoted the cause of thin solid-core cables as the ideal cable design to sonically stick it in the ear to "stranded designs." He went on to say: ". . .the fact that a cable is thin will outweight whether it has PTFE insulation, uses linear-crystal conductors, or ones made from oxygen-free lark's vomit." In other words, "thin" is the key parameter in successful cable design. According to Alvin and the rest of the solid-core enthusiasts, "thin" is in while "fat" is out. I suppose that such a stand would appeal to the diet-conscious audiophile, but as far as I can ascertain, there is no experimental evidence to support such claims (in cables, anyway). As I see it, many of Alvin's assertions are misleading. For example, use of thin speaker cables will, in many applications, alter the tonal balance of a loudspeaker and adversely affect its time-domain behavior.

In Akira Kurosawa's cinematic masterpiece *Rashomon*, the four protagonists relate differing versions of what occurred; the truth, it turns out, has been filtered through their emotional biases and needs. Similarly, while the basic factors underlying cable design (materials, mechanical integrity, and geometry) have all been promoted individually, recent research and experimentation point the finger toward a synthesis of *all* factors as being essential in evolving the state of the art in cable design.

Why thin cables sound good at all

To be fair, it should be emphasized that DNM's thin cables, distributed in the US by Music Hall, are sonically very good, although clearly removed from the top echelon of commercial alternatives. Both DNM's interconnect and speaker cable fared well in MC's recent comparative cable review in the June 1987 issue of *HFN/RR*, being rated as excellent value. Besides the attraction of low cost, what else do single-strand, small-diameter cables have going for them? The answer to this and other critical questions in cable design lies in a proper understanding of the skin effect.

A simplistic view is to regard the skin effect as AC resistance to current flow that becomes significant at very high frequencies. This is a fair macroscopic view of the situation, but it glosses over some important physical details. On a microscopic scale, the electromagnetic wave nature of the transmitted signal plays an important role. With increasing frequency, a current wave moves into a conductor less and

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less, the depth of penetration being only about 0.5mm at 20kHz. (Incidentally, this offers an explanation of why a metal is opaque to light waves: light consists of electromagnetic waves of extremely high frequency, thus there will be virtually zero penetration.) As the current penetrates the conductor, the relative phase of the current changes. With the current density at a maximum at the surface and decreasing inward, the phase is continuously retarded. It is possible that at a certain depth the relative phase of the current is reversed, so that the current in this region moves in the opposite direction to that on the surface. Such circulating currents can cause additional ohmic heating of the conductor, decrease the net current flowing in the cable, and modify low-level microvolt signal information.

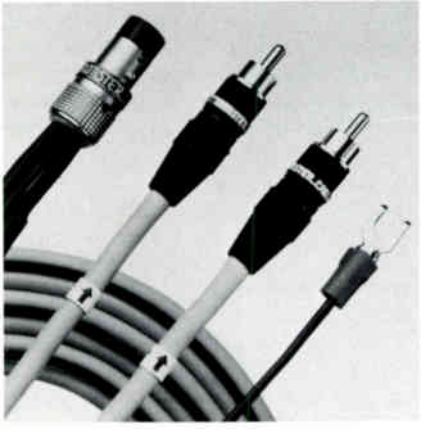
This argues that small-diameter strands are important in minimizing eddy-type currents in a conductor, but in no way does it follow that multiple strands are excluded. In fact, having a number of individually insulated strands for each leg in a proper twist is critical for optimal transmission of audio signals. Mechanically, of course, there's less to screw up with a single strand, and it is possible to make Litz-type cable sound pretty bad, but ultimately, multiple strands are necessary for the best sound. Again, it follows from the skin effect that since the current density is nonuniform across the conductor, the inductance of the wire decreases slowly with increasing frequency—the speed of propagation of the highs is faster than that of the lows.

This is a key point. When modeling signal transfer in audio cables, it is not only necessary to take into account the variable AC resistance of the cable, but its variable inductance as well. The shunt capacitance of the cable (but not the capacitive reactance) is fixed by the geometry of the cable and the dielectric properties of the insulation used. Therefore, an electrical model of a single leg of cable would look like a variable resistor in series with a variable inductor with a shunt capacitor. Bruce Brisson of MIT (the cable company, *not* the school) has done this sort of modeling on a computer, his cable designs being based on the results. He has found that a large number of different-diameter wires are necessary to control the impedance of the cable while minimizing phase delays due to differential propagation speeds. Bruce can now produce test data to back up his theoretical arguments. Using very fast scopes with true 1.3GHz sampling rates and time-domain resolutions on the order of 1ns, waveform distortions and intertransient "noise" of cables can be readily measured. Needless to say, his cables measure very well on his test set-up. Unfortunately, Bruce does not use the best materials available. . .and so it goes.

What's Bad About Thin Cables

Because a small-diameter strand is required before a cable can qualify as a full-fledged thin cable, one serious handicap of such interlinks is a high series resistance. The DNM cables with a 0.4mm strand size have about a 0.9 ohm loop resistance per meter length. This is quite high in the realm of interconnects, but really inconsequential because of the low currents carried. However, as a speaker cable this has serious ramifications for two areas of loudspeaker performance: bass alignment and crossover performance.

First, bass alignment: The amplifier's output impedance, the cable's series resistance, and the DC resistance of crossover coils in series with the woofer voice coil, all affect the electrical damping of the woofer and hence the electrical Q of the woofer. All of these parasitic resistances serve to increase the woofer's total Q. However, what is important is the speaker's overall Q, and a speaker can be designed taking into account the effects of all of these resistances—as long as they are reasonably well known. Generally a designer will assume



A gaggle of Monsters

less than 0.1 ohms output impedance for the amp (a good figure for modern amps). The DC resistance of crossover coils can be measured and taken into account. Because speaker cable is somewhat of a mystery, however, being a variable introduced by the consumer, speaker designers assume a reasonable 0.2 ohms or so.

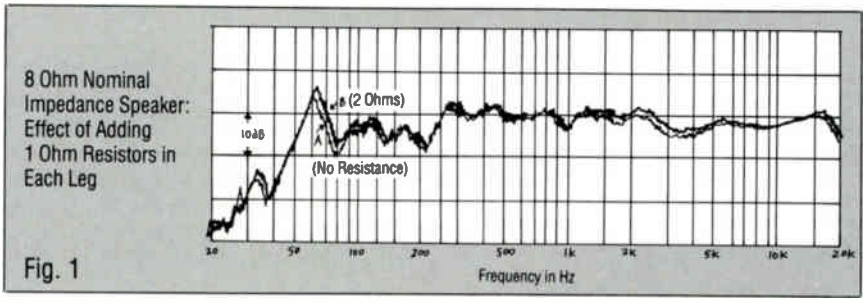


Fig. 1

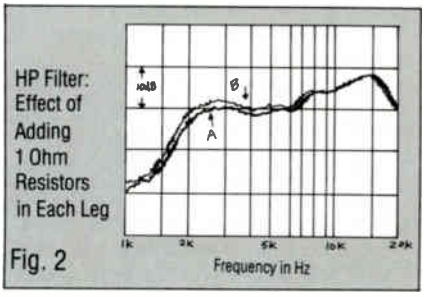


Fig. 2

Along comes the thin cable, with at least 2 ohms of loop resistance in the circuit, and with some speakers, the impact on bass quality and quantity should be very audible. (Although not so much, I think, on MC's preferred Celestion SL600 system.) But take a look at figs.1A and B. These show the effects of an additional 2 ohm loop resistance on bass response. Levels were carefully equalized to compensate for the added resistance. The loudspeaker being measured is an 8-ohm nominal-impedance bass-reflex design, and the measurements are in-room sweeps using a ⅓-octave signal source. The 2dB increase in the 60Hz bass peak with the additional resistance is evident.

The other area affected by thin speaker cables is crossover performance. A crossover is designed for particular source and load impedances. The load (*ie*, driver) is well known to the designer, but the source impedance is less certain. One usually assumes that the load-to-source impedance ratio is large, and designs the crossover with those kind of terminations. With a high-resistance cable, however, one major assumption underpinning the crossover design is rendered invalid. The crossover points shift slightly, resulting in improper blending of the drivers around the crossover frequency. This effect is evident in figs.1A and B, where the range from 1 to 3kHz is elevated 1-2dB by the additional 2 ohm loop resistance. And this is with an 8 ohm speaker—the effects will be magnified with a 4 ohm load! Even a simple 6dB/octave high-pass filter is not immune. Figs.2A and B show the before and after effect with a 4 ohm Audax dome tweeter. The added resistance gives a 2dB noticeable shift in level around the knee in the filter.

Finally, it is well-known that, with the exception of the very rare loudspeaker which is purely resistive in nature, placing a series resistance between the amplifier will modify the speaker's frequency response. Due to Ohm's Law, the speaker's modulus of impedance will be reflected in its amplitude response.

The proof is in the listening, and if the DNMs, for example, *did* sound better than the competition, the solid-core cable aficionados could rest their case. As things stand now, however, cables such as the Siltechs, the van den Hul Silver/Gold, and the MIT-330 all clearly surpass the DNM—but for a hell of a lot more money. While not demonstrating a blueprint for the ultimate cable, therefore, perhaps the thin cable does show the way to design the most cable for the money. S

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George M. Graves finds that another digital myth bites the dust

As far as I'm concerned, digital sound falls far short of the promised "perfect sound forever." The first CD players were just this side of unlistenable, and though more recent incarnations are much improved, they still have a long way to go. The myth about CDs being perfect turned out to be just that—a myth. The problem is, engineers design the things, and audiophiles listen to them. We audiophiles long ago learned the basic truth about hi-fi: the engineers can't measure what we hear (and thus tend to discount what we say), and what they *can* measure often seems to mean little in terms of audio quality.

Do engineers even really know what they're talking about? I have read many articles explaining the CD from a technical standpoint, all of which purport to understand intrinsically the mechanisms at work in both the record and playback phases of digital audio. The engineers tell us about compromises made in the name of practicality, and then tell us that, because these compromises occur in the digital domain, they have no effect on the perceived sound. (These are the same people, remember, who told us we could not hear the effects of the brick-wall low-pass filters used in the early players!) CD engineers tells us that two CD decks played through the same D/A converter and analog filtering section will sound identical. I'm here to tell you, gentle readers, that it jest ain't so!

I was recently sent, by the nice people at Luxman of America, the new Luxman LV-109 integrated amplifier. This huge, 150Wpc amp sports, among other things, a built-in D/A section which can be switched between two direct serial digital sources, and which will automatically choose the proper sampling rate (depending on whether a CD player or a Digital Audio Tape machine is connected) from either 32, 44.1, or 48kHz. Along with this amplifier came the new Luxman CD player, which has a direct digital output in addition to the normal stereo audio outputs. Since I also had on hand the Mission PCM7000 (which also has a direct digital output), the Luxman

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amplifier enabled me to compare the two players to see if, indeed, they would sound the same.

When I connected the two players to the amp, I had to use different cables for each—an obvious no-no with analog signals, but it shouldn't matter at all with digital data transfer. I had wanted to use a pair of Esoteric Audio cables that I had obtained at the Winter CES in Las Vegas, one for each machine (in the direct digital mode, only one cable is necessary because the bit-streams representing right and left audio are sent in multiplexed form down the same cable). Unfortunately, I found that the shallow RCA jack on the back of the Mission machine would not hold the massive RCA plug on the Esoteric Audio cable. I had to substitute, therefore, a handmade video cable with cheap, tin-plated RCAs to connect the PCM7000. In order to keep things from getting too complicated, I used only the D/A section in the Luxman amp for this experiment, patching the Luxman's "tape record output" jacks into the CD inputs of my reference preamp. This arrangement eliminated the unknown qualities of the Luxman's high-level preamp stage and of its power amp sections.

Two identical CDs were placed into the two players and started together; comparisons were made by switching from one digital input to the other on the Luxman. Level matching was not necessary due to the fact that only the bit-stream from each CD player was used; since both were played through the same analog section, the audio output levels would be identical.

My knowledge of digital audio, plus what I had read on the subject, led me to expect that all of the differences noted between players were due to the differences in the design of the analog sections of those players. Therefore, I would expect the two players in my experiment to sound identical when played through the same D/A and analog circuitry.

I was wrong.

The Luxman and the Mission players sounded as different from each other when played through the Luxman digital decoder as when played through their own analog circuits! My initial thought was, "No, this can't be. I've made a mistake in my lash-up somewhere." But a quick check showed that everything was working properly, except for the two dissimilar cables connecting the decks to the direct digital inputs on the Luxman amp . . .

Wait a minute—cables may have an audible effect on an *analog* signal, which is directly altered by changes in capacitance, inductance, and resistance, but a bit-stream, at this data rate, is only going to be integrated differently (the squarewaves that comprise the bit-stream will be altered in shape to a differing degree) by the two different cables. And the circuitry that looks at this incoming bit-stream is only looking for transitions from a logical "1" to a logical "0", or vice versa. It doesn't care what the squarewaves are shaped like. It doesn't have to: it is fundamental to digital processing that the actual shape of the bit-stream waveform doesn't matter!

In spite of my self-assurance on this point, I changed the homemade video cable on the Mission player to Esoteric Audio cable. (The EA cable didn't want to stay put on the back of the Mission, but I finally got it to stick with a Scotch-tape tether.) Now both cables were the same. They sounded closer, but there were still significant differences between the two units. Lest you think that I was imagining things—the same thought crossed my mind—I repeated the procedure for several of my audiophile friends. Without my telling them what they were listening to, or for, all agreed that the Mission (identified only as CD player #2) definitely sounded better than the Luxman (CD player #1) when auditioned via the Luxman amplifier's D/A section.

What conclusions can be drawn from this experiment? One could make a case that differences in error-correction performance would explain the audible differences, but how can the cable difference be explained? By any science that I know of, cables cannot affect a digital bit-stream as long as they have sufficient bandwidth to pass the signal so that the D/A circuitry can identify the transitions between "ones" and "zeroes." Both the EA and the homebrew cables are more than adequate to that task (in fact, both pass a very good LaserVision picture, thank you). You tell me.

It would appear that there is a vast difference between fact and theory when it comes to digitizing sound. And if this basic assumption about how digitized audio behaves is wrong, then what other portions of digital theory are wrong as well?

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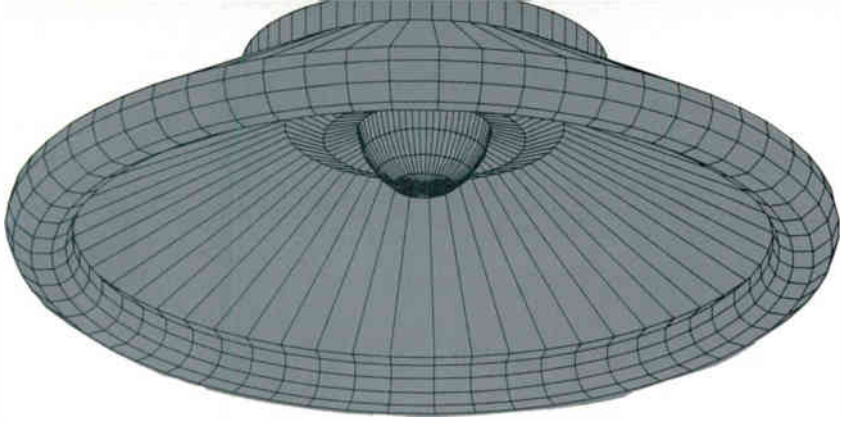

a european DAHLIA

Martin Colloms offers a different tuning of Dick Olsher's DIY loudspeaker

In Vol.9 No.1, Dick Olsher offered a neat two-way loudspeaker design which exploited the virtues of the Son Audax 8" bass/mid unit, a TPX-coned unit which I also favor. As Dick could not avail himself of measurement facilities to determine the flatness and balance, his crossover was more or less to "book." Having tried his network, I found the resultant sound rather forward and lightweight compared with neutral, commercial, BBC-balanced references, while the bass, in my opinion, was not fully integrated.

I have an alternative crossover (see figs.1 and 2) for those who may wish to try it with the same drivers. This results in a neutral tonal balance, corrects for some mild "sizzle" in the upper range of the tweeter—the tried-and-tested 1" soft-dome Audax HD12X9D25—and also provides more bass extension (achieved at the expense of overall sensitivity—90dB for Dick's version and 87dB for mine).

The new balance finds the original bass alignment underdamped, since it is maximally flat with a vent resonance close to 46Hz. Retuning to 35Hz extends the bass and maintains the correct dryness—a good feature of the original. The new port, estimated at 2" diameter by 4.5" deep, can be closely approximated to advantage: using the original, install a full-length port liner of soft polyurethane foam to reduce the internal bore to approximately 2.5". A 1"-deep plug of open-cell polyester foam (like Declon foam grille material) may be used in the port, if a still-drier bass is required in some room arrangements.

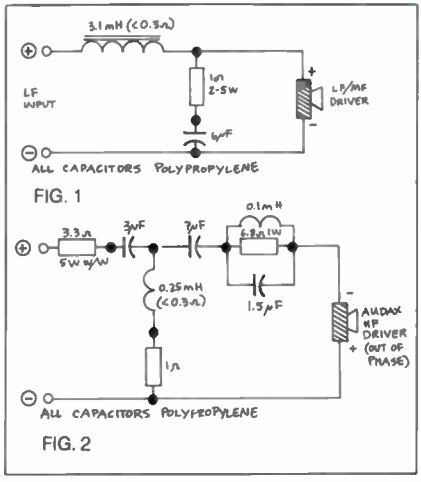

FIG. 1

FIG. 2

While trying out the alternative version,

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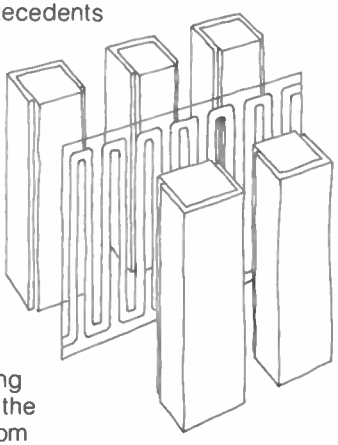

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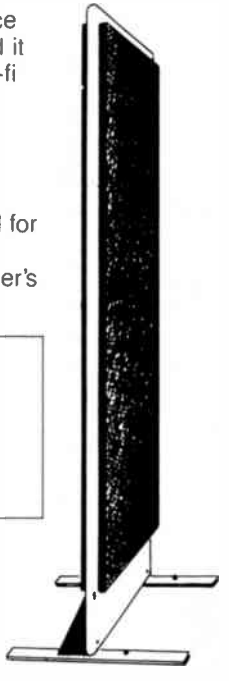

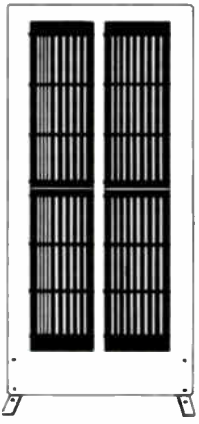

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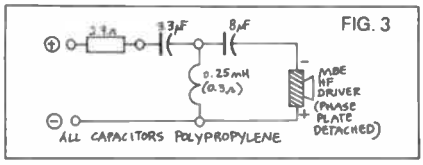

FIG. 3

felt or foam absorption should be removed from around the tweeter.

In Vol.10 No.4, Dick updated the Dahlia to the Dahlia Debra.[1] I concur with his suggestion of using the larger-magnet TSN bass-unit, which will improve the sensitivity and avoid the need for the open-cell plug in the port. Furthermore, the MB Electronics titanium-dome tweeter is a fine upgrade for which I can suggest a further improvement: With a strong blade, *carefully* pry off the front grille/protective plate. This improves the 15-20kHz range considerably.

My alternative HF filter works fine with the deletion of the final three-element network and the damping resistor, which are unnecessary with the Ti dome (fig.3). (Do not press on the dome—it will crease or dimple.) **S**

1 Back issues of *Stereophile* are available for $5.00 including postage.

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PURE GOLD

Alvin Gold

If the words on this page seem to meander slowly from side to side, put it down to the author's exhaustion, having just completed a project which involved reviewing more than 50 cassette decks while the kitchen was being knocked down and rebuilt, and with a highly pregnant wife, four zillion brats and an overdraft (you're not a real journalist without one). At the end of the week after next, according to my computerized diary (which has started intermittently jamming the computer since I changed from a PC clone to an AT clone—it's Sidekick V!.54, any ideas?), I am due to have a complete day off. "Whether I need it or not," it says. Zzzz. . .

Good morning there. Ah yes, the review project. . . I don't intend to go into all the gory details here. In any case I mustn't pre-empt the book (and, hopefully, the film of the book, the serialization and syndication rights—No Reasonable Offer Refused), but I cannot resist telling you that one of the best-sounding of the lot outside the predictable top Nakamichis is a little delectation from Sony called the WM-D6C—also known as the Walkman Professional. I had actually tried this deck once before, but hadn't really come to grips with it. Now, I discover, this slightly outsize and almost geriatric battery-powered recording Dolby-B/C portable is on a par, measurement-wise, with almost any mains-driven machinery, including some at very much higher prices, and that sonically it just about betters them all. It sounds sweet, lucid, clear, and altogether easy on the ear, works as to the manner born with Type IV metal tapes, and doesn't even cost all that much. It urinates (in the best possible taste, of course) all over any of the current CD Discpeople on all counts up to and including practicality on the move, and excluding background hiss—though Discpeople often sound noisy too.

The other rather shocking realization at the end of the project was to discover how poor most of the decks sounded when Dolby was used, especially Dolby-C. Some decks managed to incorporate Dolby processing without the sound becoming grossly flattened and synthetic, but most didn't, which points to low-grade chips and/or low-grade installations. To add insult to injury, Dolby has been responsible historically, through no fault of its own, for almost total stagnation in the development of cassette-deck technology over the past decade or more. On the one hand, all the old, and now largely discredited, measurement guys were feeding the makers with the "fact" that hi-fi meant 60dB signal/noise; on the other there is Dolby offering 10dB on a chip (20dB with Dolby-C, which predictably sounds worse), and as most machines are already good for 50dB, they didn't need to try any more.

Instead they could concentrate on the really important matters, like adding extras like track-search facilities and electronic counters. It's not that I have anything against such features, it's just that there's no such thing as a free lunch. It reminds me of a throwaway line by one Norman Willis, who heads the British Trades Union Congress. He was being interviewed on breakfast television on the day he was expecting a rather hard time from union extremists. Conversationally, the interviewer—200 miles away in a warm London studio—remarked on the cloudiness of the weather he could see on the video feed. Norman, a sad-eyed Friar Tuck lookalike of great concealed depth and even greater, unconcealed girth, looked out the window behind him and intoned, almost to himself, "Yes, it does look rather ironic up here."

Now I'm left with the rather difficult task of switching to a subject that has no conceivable relation to the last, beyond the rather tenuous one that it'll rattle a few vested interests. But you'll have to be patient; I'm going to tell this *my* way.

It started innocuously enough with a scheduled meeting of *Hi-Fi Answers* main contributors held just a day or so after I cleared the last of the cassette-deck copy from my desk. We met at the home of Jimmy Hughes (regarded by some as the UK's Enid Lumley) because his place is fairly central, and Jimmy regaled us with tales of his experiments with . . . no, this must wait, though I must tell you that even he couldn't keep an entirely straight

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face while telling the story. About three hours and a not conspicuously alcoholic meal later, with most of the day's business out of the way, JMH demonstrated to us some of what he had been telling us about. I say "us" advisedly so that you know I have witnesses, including the extremely levelheaded Editor and Assistant Editor of *Hi-Fi Answers*, Keith Howard and John Bamford. They are not suggestible types, which is a good thing in the circumstances. Neither am I.

First, though, I'd better tell you a little about the system. At the present time, the hardware consists of a choice between a pair of active PMS Linn Isobariks and Magnepan MG-IIIs (heavily modified), three Krell KSA-50s, two Linns—one with a Breuer and one, the active one, fitted with a Mission Mechanic—and Kiseki Agate Ruby or Goldbug Ms. Brier cartridges. For compact disc, he also has a Cambridge Audio CD-1 player. Almost none of this was in use. When we were there, most of it was on a landing outside the listening room, and what he was actually using was a solid-core wired system with a DNM pre- and 30Wpc power amp (I have one of these too, and will be reporting on it soon), with, as input, a middle-market Denon CD player. On the other end, a pair of £200 ($350) 8" two-ways, the A&R Arcam Twos, extensively modified.

To give you a flavor of the modifications, a 3.3-ohm resistor had been removed from the tweeter circuit and replaced with a long loop of thin wire to restore some of the resistance "naturally" without all the problems caused by the physical construction of resistors, plus some mechanical (foam) filtering added around the tweeter. The Linn/Mechanlc combination was also in use, but with a cheap Grado moving-magnet, modified in ways I cannot begin to describe, but stem from the ideas of one Peter Belt (see later): "I'm not sure about this cartridge," JMH confided. "That is, I'm not sure if it's better than the best $800 moving-coils or not."

More interesting than the equipment was the way it was used, which in large measure defies rational explanation. The room concerned has a number of problems, one of the most important being that it "booms" quite badly when excited. There are also some quite severe flutter echoes. In the US, and to some extent in this country, this kind of problem would have been treated using wave traps and acoustic screens, perhaps with some carpet or other sound-absorbing material on the walls to suppress the flutter echo. Not JMH. In the five years or so he has lived in that house, the acoustic problems have been convincingly tackled by successive improvements in the performance of the system. And sure enough, as each layer of glare and artificiality has been stripped away, the contributions of the room have receded to the point where they are now, for all intents and purposes, inaudible. Yet the loudspeakers, designed for free-air use, were placed hard into the corners of the room, and incidentally wired in reverse absolute phase. Never mind why.

Lest this all seem like a peculiar species of affectation, I must tell you that the system truly works. If it didn't I wouldn't be writing about it. I have more expensive equipment at home, and I figure I have a reasonable amount of expertise in setting up systems so that they play tunes. I am certain that I can set up a system with a more sophisticated sound, that goes farther up and down the frequency band, that maybe sounds sweeter. But I must concede with some regret that, at the time of writing, I cannot get my system to work as *well* as his. We listened to both CDs and records, in the main with software of recent vintage—often just the kind of stuff at which audiophile magazines like this turn up their noses. For all practical purposes, *both* formats worked so well as to transcend their stereotypes.

The system had an almost elemental quality, as though the soul of the music was stripped bare. It had all the qualities we struggle to achieve with expensive gear, but fail because it is too complicated for its own good. There was no question of artificiality. There *was* soundstaging—we were sitting in it—but most of all we were immersed in interpretation and performance in a way that parallels live music-making. The system doesn't *sound* as good as the real thing; that is impossible with the present state of the art, and on reflection it's not really necessary and arguably not even desirable. It simply had the *communicativeness* of real music-making, allowing the same kind of communication because it offered truly high levels of resolution and a quite remarkable lack both of "glare" and of the superficial qualities that have always characterized high

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fidelity. You've probably had the same experience yourself on occasion from a pocket transistor radio, most likely in your youth when you hadn't educated yourself out of being able to listen. I guarantee you have never heard anything so musically valid from a roomful of Audio Research and Infinity IRS.

But I've left out certain details: a whole series of bolt-on goodies that come from a guy named Peter Belt. The goodies are foil-like objects to stick on loudspeaker stands, loudspeaker diaphragms (!), and CDs, crocodile clips and washers that attach to wires, peculiar little plastic edifices, special resistors that go in the mains line or amplifier PSU, and much, much more.[1] The best way to approach the goodies is via a subject that has been discussed at some length in this column before, namely the electromagnetic field effects that underpin the thin, solid-core cable idea. The central theme here is sort of related; rather than electromagnetics, the ideas here come from an even more neglected area—electrostatics. From here on, however, I get lost.

Imagine. There we are, sitting quite comfortably listening to JJ Cale's *Grasshopper*—or was it the new VPO/Maazel Mahler 3?—when JMH picks up a device consisting of a battery holder fitted with two 1.5V batteries with a plastic arm at each end and a couple of color-coded electrets near the battery terminals, and "treats" a disc by using it rather like a record-cleaning brush. The direction of application is critical, we're told. He plays the disc, takes it off, reverses the direction of treatment, plays the disc again, and the sound just falls to pieces. I mean to pieces. We're not talking subtleties here, and I *don't* believe in magic any more than I believe in TV evangelists.

The disc jewel boxes are treated with stuck-on electrets, though only a few of JMH's 2000 CDs (and none of his 8-10,000 LPs) have been so treated. There are further electrets and other peculiar plastic constructions on the other equipment, on the floor, and even on the room light. The room light? JMH readily demonstrates that lightbulbs have a small but definite deleterious effect on the sound of a system in the same room. When the bulb is treated, by a device that looks like a crocodile clip (the treatment is said to be effectively permanent), the system sounds better.

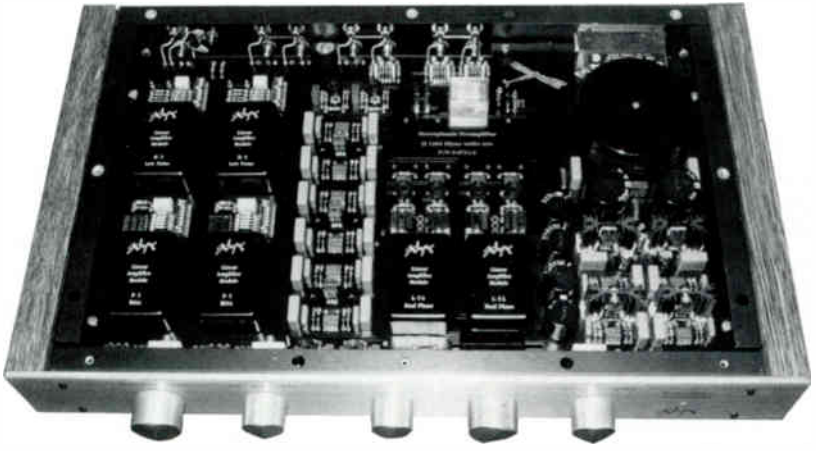
Here's something even stranger. Well, I mean, what would *you* make of a compact disc in its jewel box being removed from a shelf and replaced upside down—it's about 12" from any of the gear, by the way—whereupon the sound of the system *noticeably* deteriorates? Had anyone suggested such a thing last week, I'd have told him where to get off, which not coincidentally is exactly the nature of the reaction that Peter Belt has had so far from trade, public, and press alike.

I remain skeptical about the nature of some of the solutions myself. I've yet to hear even a convincing articulation of the problem. The trouble is, I've heard it with my own ears. I cannot ignore the way the qualities of the system I described earlier were replaced by muddle and confusion when some of the "treatments" were undone. I'll be back to this topic, which, I suggest, will run and run. First, I have a bit of research to do. . . . **S**

[1] Peter Belt was for many years the manufacturer of an excellent-sounding British electrostatic headphone, and his accessories are based, in the main, on the work he did with permanently charged electrostatic materials (electrets). We have collected a set of Peter's palliative measures in the *Stereophile* office, and will report further on their audible effects when I can find a writer willing to suspend his disbelief.
—JA

KLYNE SK-5A PREAMPLIFIER

J. Gordon Holt



Klyne SK-5A preamplifier

Solid-state stereo preamplifier. Inputs: Phono, Line, Aux, Tape. Controls: Source, Mode (Tape Monitor, Tape Source, Mono), Balance, Volume, Phase (Invert, Mute, Normal). Gain: Phono 22-34dB; Line 14-20dB. Bandwidth: MC phono 1Hz-150kHz; MM phono and Line in 0.5Hz-250kHz. Output impedance: 300 ohms. Dimensions: 19" W by 2½" H by 12½" D. Weight: 12 lbs. Price: $3250. Approximate number of dealers: undisclosed. Manufacturer: Klyne Audio Arts, 828 7th Ave. NE, Olympia, WA 98501. Tel: (206) 943-5420.

Klyne Audio Arts has an almost Zen-like approach to the design of its products. Like the best Japanese designs, Klyne's preamps are aesthetically pleasing in appearance, do exactly what they're supposed to, and their controls are not only where you would expect them to be, but have an almost sensually smooth action. Internal construction, too, is a work of art—the kind of design which, transferred to a tapestry, would grace the wall of any listening room. You have to see the insides of a Klyne preamp to appreciate how attractive-looking an audio component can be. But physical beauty is only one aspect of Stan Klyne's designs; of all the electronics manufacturers I know of, Klyne Audio Arts also makes products more adjustable than any others.

Most preamps have what are called "features," which allow them to do many interesting things like tape cross-copying, ultrasonic filtering, stereo channel reversing, and feeding a mono source coming into one channel to both outputs. Some feature-laden preamps also have lots of multicolored indicator lamps which eliminate the need for looking to see what switches are set where, if you can remember what the different colors mean. The Klyne SK-5A, on the other hand, has few features and only one light—which tells you when it's turned on. The preamp can select between two high-level sources and phono. It has a monitor facility for one tape recorder. It has a mono/stereo switch—an unusual, and welcome, accommodation for those of us who still listen occasionally to primitive recordings because some are superbly

performed. And it has a ganged volume control and a separate balance control (bless you, Stan), and a polarity switch with an intermediate -20dB mute position. Instead of additional features, the SK-5A has numerous adjustments primarily for tailoring the sound of the phono preamp, allowing you to do a few things with greater precision than I, for one, care to exercise.

Stan Klyne's preamplifiers have *always* been characterized by the inclusion of more adjustments than you could shake the proverbial stick at. Not every adjustment in the SK-5A is guaranteed to have a dramatic effect on the sound, but for those who can hear their effects, the adjustments are all there. As far as I'm concerned, this offers the dubious attraction of an open-ended series of listening experiments to warm the heart of any neurotic-compulsive. I do concede that Klyne, in their owner's manual, offers guidance for settings to match some of the popular cartridges—but I fear that many perfectionists will use the flexibility of these adjustments to endlessly diddle the sound of their phono system.

It reminds me of the SA-2 head amp from Counterpoint, which had a front-panel adjustment to be set for "the best sound." This allowed the user to make the device sound like a solid-state head amp or a tube head amp or an anything-you-want-in-between head amp. And what did the control do? It adjusted the amount of harmonic distortion the thing added to the signal, but *that* was never mentioned in either the instructions or the descriptive literature.

The SK-5A has adjustments for resistive and capacitive loading for cartridges; the desired loads are selected by small DIP switches inside the unit. You can choose the usual 47k-ohm MM-cartridge load, or any one of eleven sensibly incremented MC values ranging from 15 to 1000 ohms. For capacitors, the choice is five values, from 650 to 110pF. If you feel you need a value not represented by those switches, there are special receptacles on the circuit board that allow you to plug in resistors and/or capacitors of your own choice, without the need for soldering. (This accords you the bonus privilege of agonizing over which kind of capacitor sounds best.)

Other switches allow you to adjust the gain of the phono preamp stage from 22 to 34db, and provide for high-frequency rolloff with -3dB points from 50kHz (!!) to 16kHz to tame the fiery high end of those MC cartridges which, while popular with perfectionists, have (as far as I'm concerned) absolutely nothing going for them but extreme quickness. The only rolloff that had an audible effect in my system was the lowest one (16kHz), but it was disadvantageous, yielding a slightly closed-in effect at the top end.

Overall preamp gain is also adjustable, as with my ARC SP-11, but here it's done with resistors inserted into sockets on the circuit board. (The SP-11 accomplishes this with a significantly more convenient front-panel control.)

It's not that I disapprove of the ritualization of record playing; it's just that I don't sympathize with it. (The entirety of my own preplay prep consists of a slow wipe with a damp velvet pad and the attachment of the SOTA Star Sapphire's vacuum-seal cap.) But every field of endeavor has its personalities that love to tweak to their heart's content, and there are a lot of audiophiles who relish any ritual which, like foreplay, heightens the tension and excitement of actually listening to recorded music by postponing the consummation as long as possible. Of course, my own enthusiasm for all these adjustments is muted by the fact that I needed to use only the adjustable phono gain provision. My Ortofon MC-2000 was chosen specifically because it has a flat high-frequency response (since none of the other preamplifiers I've used offer the facilities of the Klyne), and must be operated with its own step-up transformer, which eliminates the relevance of cartridge loading. So, I shall concentrate on describing the sound of the SK-5A, and let you decide whether you could live with the uncertainty of having to set all those adjustments by ear.[1]

1 If you truly seek a final solution, there is a way. First, use only records of known and impeccable ancestry for listening. That means Sheffields, Reference Recordings, Telarcs, Wilson Audios, and the like.

Second, feel The Force. Trust your instincts. If you feel that one switch setting, or combination thereof, sounds better on a statistical majority (more than 51%) of those recordings, use that setting and be done with it.

Third, if you can't decide between two settings, be honest with yourself and admit that the moment you stop thinking about it, the difference won't matter anyway. Toss a coin, accept its decision, and get on with the business of listening to music, which is a lot more fun and much less harrowing.

The SK-5A also has two sets of outputs in antiphase. What does this buy you? Well, just by reversing the polarity of one pair of speaker wires, you can operate any stereo power amplifier with out-of-phase signals, to mirror-image the demands on its power supply. (This arrangement, christened OOPs in 1976 by Trevor Lees—Opposite channel Out of PhaSe—simulates the behavior of a two-phase AC line circuit, and has been shown to improve both soundstage breadth and overall detail of an amplifier.) You can also drive any amp capable of bridged operation, directly from the preamp, without having to switch in an additional input-inverter stage. And if you get the impression you have *everything* hooked up in reverse polarity, the front-panel Reverse switch inverts *all* signal sources.

For testing the SK-5A, I used what has been my reference system for the past six months or so: an Ortofon MC-2000 cartridge (with and without its own step-up transformer), the Well-Tempered Arm, the SOTA Star Sapphire turntable, Sony's 650 and 703 CD player/converter combo, an Audio Research SP-11 preamp, a pair of the mono Threshold SA-1 power amplifiers, and a pair of Sound Lab A-full-range electrostatic speakers.

Sonically, the SK-5A was an unusual combination of incredible detail and slightly laid-back perspective. Bypass tests revealed that the high-level section was contributing most of the preamp's "flavor," characterized by a very slight crispening of the extreme high end and an upper midrange that sounded (but did not measure) as though it was broadly dished downward by perhaps 1dB. The high-end crispness, far from what could be described as edgy, may well have been nothing more than the subjective reaction to the very slightly withdrawn upper midrange. Low-frequency performance through the high-level section was essentially flawless, replicating exactly the range and heft of the input signal.

Bypassing an actual phono source using my inverse RIAA network, as I do on every phono preamp, showed that the RIAA equalization was right on the nose; I could hear no difference between input and output signals. In actual use with phono cartridges, the sound of the preamp (or the sound of your cartridge, if you prefer to put it that way) will vary depending on the resistive loading and high-frequency rolloff (if any) you choose. What slight colorations the SK-5A has appear to originate in the high-level stages.

The only reason for testing the SK-5A with an actual cartridge source was to determine how the unit handles the spurious material unique to disc reproduction: the ultrasonic pulses stemming from slight (or less than slight) groove mistracking. In this area, the SK-5A acquitted itself superbly, adding no perceptible roughening or exaggeration of the pulses. The preamp's front end is, by the way, extraordinarily quiet. With my Ortofon cartridge, however, with an output (0.05 mV) by far the lowest of any MC, hum was barely audible at normal, moderately high listening levels, but the hiss level was, quite simply, intolerably high. Though no more obtrusive than 7.5ips tape hiss, there was just too much, being clearly audible through every moderate- to low-level musical passage. This was not a problem of inadequate gain, for, with every gain adjustment set to maximum, there was plenty of gain to drive the power amps to full output. This simply confirms my past experience: there isn't a preamp around that can successfully amplify the tiny output of the Ortofon. I sincerely doubt that background hiss would ever be a problem with a conventional MC cartridge having more than 0.1mV of output. (I have always maintained that the Ortofon MC-2000 *must* be used with its own T-2000 step-up transformer, and have yet to find a preamp or head amp that would prompt me to qualify that admonition.)

Low-frequency performance from discs was outstanding, although a little on the lean side. Only once before have I heard such deep, detailed bass from my Ortofon/Well-Tempered Arm combination, and that from a preamp with, in my system, such intractable hum problems—even with average-output MM cartridges—as to be unusable.

Soundstaging too was excellent, with quite remarkable depth, but not the best rendition of perspective that I have heard. Imaging was stable and very specific, but the real surprise was the SK-5A's soundstage breadth, which, considering the power supply common to both channels, was greater than I felt it had any right to be! Klyne claims to provide lots of power-supply isolation between channels—but so do a lot of other manufacturers whose

products demonstrate definite lateral compression of soundstage. From the SK-5A, recordings which normally image slightly beyond the outer placement limits of the loudspeakers were definitely and solidly located farther beyond those limits than I had ever before heard them. Yet there was no false broadening of the images between the speakers, as would be the case were the widened soundstage merely the side-effect of an intra-channel phasing anomaly. I would be interested to know the techniques that Klyne uses to obtain this kind of performance, which I would have expected only from completely separate mono preamps.

Inner detailing was amazing! This preamp separates and delineates the voices in complex material as well as, if not better than, any preamp I have auditioned. Complex orchestral *tutti* were so well resolved that it was often possible to follow the lines of the individual instruments in each choir, and the envelope of each sound seemed etched against its background. This preamp almost redefines the concept of clarity, but does so without giving the impression of clinical coldness which I had felt to be the only major shortcoming of the previous-model Klyne preamp, the SK-5.

I felt compelled to compare the SK-5A with Audio Research's SP-11, my current reference preamp, and one of the two best available preamps by expert consensus, in spite of the $1750 price difference between them. The SK-5A and SP-11 are not similar. The SP-11's colorations—a slight thinness in the deep bass and warmth in the midbass region on phono—were not shared at all by the Klyne. By comparison, the Klyne sounded almost thin in the midbass, but seemed to have no effective low-end limit. It had far more LF impact than the ARC, and noticeably better detail and pitch delineation. (Interestingly, there was less audible difference between their LF performance on bypass tests using an inverse-RIAA equalizer into their respective preamp stages. I can't explain this.)

By comparison with the Klyne's extraordinary detailing, the SP-11 almost sounded muddy. That's an exaggeration, but the difference was not very subtle. In other areas, I felt the ARC had the competitive edge, although in ways that are a little difficult to describe. To say, simply, that the SP-11 sounded "more musical" to me is a cop-out, but I'll say it anyway. I just found the SP-11 to be more seductive and emotionally involving than the Klyne. The SP-11's high-level section does better on bypass tests than the Klyne's, but its phono section is a bit weak at the extreme low end, and doesn't quite have the Klyne's definition. In all other areas, though, the SP-11 is, for all intents and purposes, a straight wire with gain: I cannot attribute my preference for it to a mere taste for "euphonic coloration." The ARC is sweeter and—here we go again!—more musical at the extreme top, more "alive" and immediate-sounding than the Klyne, and somehow makes instruments sound a bit more like they do in the flesh. Having said that, I shall now waffle.

My present power-amp/loudspeaker system has been carefully chosen to produce what is, in my opinion at least,[2] the most musically natural sound it is possible to get from what I believe to be the most sonically neutral signal sources,[3] taking into account my preference for a rather warm sound over a cooler but more etched sound. My system is rather less than forward in perspective to begin with, so it is understandable that it might not be very flattering to any electronics component (like a preamp) which is itself a little recessed. Consider also that my speakers do not have the slightest tendency to soften or round off extreme highs, and you will see why I am not prepared to give an unequivocal figure of merit to the SK-5A *or* the SP-11. My point is that the Klyne SK-5A will almost certainly produce more musical naturalness from other systems (brighter in the upper mids and softer at the extreme top) than it did from mine, without any loss of the preamp's remarkable definition and soundstaging performance. (On the other hand, it meshed so poorly with my system when I substituted the Mirror Image 1.1 amplifier for the Thresholds that I didn't care to listen to that combination for more than a few minutes. The sound was thin, dead, and uninvolving.)

There is no doubt in my mind that the SK-

2 JA: Feel free to footnote a disagreement if you see fit.

3 I wouldn't presume to, JGH. My only disagreement with your choice of components is that it is less than forgiving of the recording faults typical of modern rock recording. But, with nearly all present-day artists, who would care anyway? —JA

5A is, in many respects, a world-class preamplifier, whose attractiveness is little diminished by the fact that it costs about $1750 less than the acknowledged world leader, the SP-11. The degree to which the Klyne approaches perfection in your system will depend entirely on the peculiarities of that system and on what you are looking for in reproduced sound.

Regardless, then, of any personal reservations on my part, the Klyne SK-5A deserves a strong recommendation. And gets it. **S**

THRESHOLD FET 10 PRE-TROL (CONTROL PREAMPLIFIER/RIAA AMPLIFIER)

J. Gordon Holt

Threshold FET 10 preamp

Dedicated line-level and phono preamplifier units. FET 10/H: Line-level control center. Gain: 20dB max. Frequency response: 1.5Hz-150kHz +0, -3dB. Max output: 20V. THD: 0.01% at 1V out. Output impedance: 100 ohms. Inputs: Analog Disc (Phono to you, fella!), Auxiliary, FM/AM Tuner, CD Player, Video Audio, 2 Tape Monitors. Outputs: 2 Tape Record, Main to power amp. Controls: Source, Mode (Stereo, Reverse, Invert, Mono), Signal Distribution (Monitor Tape 1 or 2, Copy 1 to 2, Copy 2 to 1), Volume, Balance. FET 10/P: Phono preamplifier/RIAA equalizer. Gain: 40 or 60dB at 1kHz. RIAA accuracy: 0.025%. Frequency response: 1.5Hz-150kHz +0, -3dB. Max output: 20V. THD: 0.01% at 1V out. Output impedance 420 ohms. No controls. Dimensions (each unit): 19" W by 2" H by 10" D, overall. Approximate number of dealers: 45. Prices: $1800 (FET 10/H), $1100 (FET 10/P), $2900 if bought together. Manufacturer: Threshold Corporation, 1945 Industrial Dr., Auburn, CA 95603. Tel: (916) 888-0600.

What, you may well ask, is a "pre-trol?" Well, Threshold Corp. calls its FET-10 a preamplifier, but it isn't, really. In fact, it isn't an It at all; it's a Them. Only half of Them is a preamp, and you can buy each half separately. If that sounds a little confusing, maybe it's because some of the old, familiar language of audio is starting to lose its relevance.

Once upon a time, in the dark ages before magnetic pickup cartridges, that part of an audio system that controlled everything—input switching, volume, bass and treble—was

called the control section. When the control section became physically separated from the power amp, so you could put one on a shelf and the other out of the way, the control section became a control unit. Then came the revolutionary General Electric variable-reluctance magnetic pickup, along with the need for pre-amplification and a new concept in home hi-fi: playback equalization. Ever since then, every control unit has included a preamp section, and has been known as a preamplifier or preamp unit. Until now.

The enthusiastic public acceptance of CD, and the waning of LP's popularity, is making a preamp stage superfluous for increasing numbers of audiophiles, who balk at the idea of paying $3000 or more for a full-featured preamplifier in which perhaps half of the cost may be tied up in a no-holds-barred phono preamp stage they will never use. They have continued to pay the price, grudgingly, because they had no choice. Now they are getting that choice.

I don't know who did it first, but *Audio*'s 1986 Equipment Directory lists five "preamplifiers" which aren't—from Cello, dB Systems, Motif, Threshold, and Strelioff—but are instead dedicated high-level control units, including a line-level output buffer amp but lacking a phono preamp section. The Mod Squad's Line Drive unit (reviewed by AHC in Vol.10 No.3) even omits the output buffer amp. Undoubtedly, more such will follow, increasingly underscoring the need for a return to the old way of talking about things that control audio signals. Since some of those things no longer contain preamps, but all are signal controllers, the term "control unit" is going to have to come back into use. The alternative is to call them PWPs—preamps without preamps, which is just silly enough that it will probably catch on (sorry I suggested it!). Then, with one unit unambiguously called the "controller," its companion piece, for the reactionary LP fringe, would, naturally, be called a "preamp." Nice and tidy! Except . . .What do we then call a preamp plus a controller plus a preamp, such as Threshold's FET-10? Logically, a "control preamp," which is clumsy at best. Hence my coining of the term "pre-trol," which is only a little less clumsy. If anyone has a better suggestion, I'd be happy to hear it.

Another term suffering increasing ambiguity these days is "audio." *Stereophile* readers automatically think of audio as music reproduced from records. But video sources also have accompanying sound (or audio), and a lot of that sound is more or less musical. So, how does one make a verbal distinction between audio audio and video audio?

In a combined music and video system, a plug or receptacle marked "audio" could be for a CD, a phono, or a video source. Threshold makes the necessary distinction by marking one input to their FET-10/H "Video Audio." This is awkwardly polysyllabic, and takes up too much label space besides. What we need is a single short, snappy word to describe this. Like "vaudio." Okay, so it's ugly sounding, but certainly no more so than eigentone, giga-Hertz, or *litzendrabt*—any one of which sounds like exotic profanity. The word "vaudio" says exactly what we want it to say—succinctly, trisyllabically, and in only six letters. (It's also a high scorer in Scrabble.) All it takes is getting used to.

Which brings us back to the Threshold FET-10. It consists of two matching units: a controller which handles only high-level signals, and a dedicated, control-less preamplifier for phono cartridges. They are identical in shape and size, designed to stack neatly one atop the other, and use identical outboard 15V power-supply boxes with enough wire in between (five feet) to allow the supplies to be placed where they won't radiate hum into the preamp unit (assuming you use it). If you don't own an old-fashioned electro-mechanical LP player, the controller is all you will ever need in order to listen happily to CDs, cassettes, open-reel tapes, FM broadcasts, video sound (vaudio), or an auxiliary. If you insist on clinging to the past, just add the preamp unit to the controller. If you don't, you pay only for what you need.

As you can see from the controller's input lineup above, it provides just about every control and all the input connections a serious audiophile could want. The only things missing are Mute, Phase Invert, and Bypass settings, whose utility for routine use is arguable. The preamp unit, on the other hand, looks like your basic Black Box (in brushed aluminum), with one pair of inputs, one pair of outputs, and nothing on the front panel but Threshold's logo medallion. The preamp, however, is

much more sophisticated than it looks.

Having just rhapsodized over (and questioned) the plethora of cartridge-tailoring adjustments in the Klyne SK-5A preamplifier as though they were something unique, I now find that the FET-10 is almost as versatile as the Klyne. The Threshold also has switchable resistive and capacitive loading for the cartridge, as well as adjustable preamp gain. Available by slide switches are resistive loads of 22, 47, 100, and 1000 ohms (all for MC cartridges), and 47k ohms for MMs. Available capacitive loads are 50, 100, 150, 250, and 1000pF (1nF). Gain is selectable at 40 or 60dB by moving small plug-in jumpers. All that's missing, relative to the Klyne, is the ability to plug in resistors or capacitors that aren't provided via switch selection.

Since the Klyne report is slated to appear in the same issue as this review, I refer you to that for a discussion of how I feel about cartridge matching. But it's been a while since I explained my feelings about bypass testing of preamps.

It is much easier to assess the accuracy of a preamplifier than it is to evaluate a power amp. As I mention in my report on the Mirror Image 1.1S power amp (also in this issue), the problem with a power amplifier is that you can't listen to it without using a loudspeaker. And since you can't listen to a loudspeaker without an amplifier, it is almost impossible to tell what either sounds like. You can make an educated guess, based on what a lot of loudspeakers sound like on a lot of amplifiers, but you can never be absolutely sure. Not so with a preamp. Being a voltage amplifier, a preamplifier is supposed to be matched into a load high enough to have no effect at all on its sound. That is, typically, a load at least 10 times its output or source impedance. And since—except for volume and balance adjustments—a high-level controller like the FET-10 is intended to put out an exact replica of what goes into it, its success is easily gauged by comparing, by ear, the input with the output signals. That, the basis of the bypass tests which I have used for the last several years, is the reason I am able to report that a preamp's high-level section is either absolutely accurate, or deviates in this or that way from absolute perfection.

Bypass testing a phono preamp stage is a little more difficult though, partly because it is *supposed* to change the sound by impressing upon the signal the standard playback-equalization characteristic defined by the Record Industry Association of America, and partly because real-world phono cartridges emit certain spuriae peculiar to mechanical groove-trackers (and not easily replicated by another signal source). The first problem is easily circumvented by the use of an inverse-RIAA network, which gives the input response a characteristic complementary to the playback EQ. The second requirement—the addition of small mistracking pulses to a cartridge's sound—while electronically simulatable in theory, has so far evaded such replication. For that reason, all my preamp tests include some listening with a real cartridge. The extent to which the preamp stages exaggerate or add roughness to the cartridge's slight mistracking pulses continues to be an important difference between competing preamp designs. (High-level stages have a similar effect on the garbage content in CD sound, but the basis of this problem has been diminishing steadily as players continue to improve.)

For bypassing, I derive my test signals from CDs and from original and 1:1-copied 15ips open-reel tapes. A special switch box enables me to adjust the bypassed signal to a comfortable listening level, while the preamp's volume control is then used to match its level to that. Other components used for my FET-10 tests included the Ortofon MC-2000 cartridge with its own transformer, the Well-Tempered Arm, SOTA Star Sapphire turntable, Threshold SA-1 power amplifiers, the Sony ES CD combo, and Sound Lab A-3 loudspeakers, with acoustical treatment of the room by ASC Tube Traps.[1]

First, the high-level ('10/H) controller. The most easily assessible of the two, this came through with flying colors. Set for unity gain, a device like this should sound like a straight wire without gain. It did. Even under unpressured conditions of leisurely listening, I was unable to tell whether the FET-10/H was in or out of the circuit. I know some of you are going to say, "Well, AHC would have heard a difference!", or some such, but while I am as

1 I consider these standing-wave absorbers to be absolutely essential for decent LF reproduction in any room smaller than an auditorium. They are the *only* means for solving LF problems in a room not initially constructed with standing-wave traps in the walls. Any audio perfectionist who isn't using them is wasting a large percentage of his component-purchase outlays.

aware as anyone of my responsibility to be a nit-picker, I will not deceive you or myself by fabricating nonexistent differences just so I can demonstrate how fantastically well I hear. (I do not accuse AHC of having done that; I'm just saying that I won't.) I defy anyone, on a fair test, to reliably distinguish the FET-10's high-level unit from a straight-wire bypass.

Next, I turned my attention to the FET-10/P. Upon removal of the cover, this simple-looking sealed can becomes, like the Klyne SK-5A, a worm farm. The adjustments thus revealed allow you to optimize the matching of just about any phono cartridge in existence. Unlike Klyne, though, Threshold does not challenge you to do this by ear, but suggests that you follow the cartridge manufacturer's loading recommendations. I was happy to do so, but there's no reason *you* have to.

Also like the Klyne SK-5A (and, in fact, like every other preamp I have tried), the Threshold is not directly usable with the Ortofon MC-2000 cartridge. As usual, the problem isn't lack of gain, but excessive hiss. I have yet to find a substitute for Ortofon's own T-2000 step-up transformer for use with this cartridge.

Bypass tests on the FET-10/P preamp unit were no more rewarding than those on the control unit. The input and output were so alike that I could never be certain which I was listening to, without checking the setting on the bypass switch. *Nothing* seemed to change from one setting to the other, not spectral balance, texture, detail, or spatial characteristics. With a real phono cartridge instead of the bypass signal, there was not a trace of added texturing at the high end, where differences between preamplifiers most often show up.

That is not to say, however, that the FET-10/P sounds exactly like the front end of either the Klyne SK-5A or the Audio Research SP-11. It doesn't, even though those other preamps also did outstandingly well on bypass tests. But how can two preamps which sound perfect on a bypass test sound different from one another? Because individually subliminal imperfections may become "liminal" when the errors are in opposite directions and compared with one another. That, apparently, is what happens here.

By comparison with the SP-11, the phono stages of both the Klyne and the Threshold have noticeably better low end. Deep bass from discs is deeper, obviously more extended, and has much better heft and impact. However, where the Klyne's bass is tightly controlled almost to the point of being dried up, the Threshold's bass is somewhat rounder, giving bass instruments (basses, cellos, organ pedals) more the kind of bloom I hear at live concerts. (The SP-11's deep-bass deficiency, interestingly, is more noticeable from a cartridge than it is from the inverse-RIAA bypass source I use for evaluating a preamp's phono stage. I cannot explain why.)

Extreme highs from the Klyne and Threshold are virtually identical: crisp without being edgy, open without being tizzy. By comparison, the SP-11's extreme highs are very slightly soft, with an airy delicacy which may be less accurate than what I hear from the other two preamps but is, at least with electrostatic speakers, a little closer to the sound of live musical highs.

In terms of detail, the Threshold is perhaps a shade better than the SP-11 but not quite the equal of the Klyne. Since I do not value detail above all else, I did not find the differences between the three to be worth much weight, but others are welcome to differ with that. Soundstaging and depth from the Threshold were identical to that of the Klyne; both were excellent, although neither's depth rendition was quite in the class of the SP-11. I should mention, however, that there are those who feel that the SP-11 exaggerates depth. So be it—a little exaggeration does no harm at all, and sounds nice.

So, what did I think about the FET-10's sound? I loved it. It is exceedingly, persuasively musical, in the best sense of that word, and a delight to listen to. The fact that I find the SP-11 (with my associated equipment) just a shade more seductive, despite its very slight imperfections, does not in the slightest diminish my regard for the Threshold. Having, at the time of writing, listened to the FET-10 for about 30 hours, I still hear nothing about it that would bar it from a top Class-A recommendation. The Threshold FET-10 pre-trol is possibly the best such in the world; it is almost certainly the best buy in a pre-trol that you will find.

But is such a device cost-effective in the first place? A reasonable question: When any CD player with a volume control can be connected directly to a power amp, is it possible

to justify $1800 for another volume control and source selector? It isn't, if CDs are all you will ever listen to. But if you want to listen to other signal sources, or do any serious tape recording, you will need a balance control and a way of juggling program sources. That means some active circuitry, with its potential for fouling the sound. If $1800 is what it costs to avoid the fouling, that's the way it is.

Congratulations, Threshold. You've got yourself a winner! S

NYAL "MINUET-IN-A" HYBRID-TUBE/MOSFET STEREO PREAMPLIFIER

George M. Graves

"Moscode" stereo preamplifier with two tape loops, variable-gain phono input. Phono input impedance: 47 k ohms, variable capacitance. Price: $945. Approximate number of dealers: 45. New York Audio Labs, c/o Hybrit Ltd., PO Box 2146, Peekskill, NY 10566. Tel: (914) 739-6267.

Harvey Rosenberg is well known for tube-MOSFET hybrids, his NYAL (New York Audio Labs) Moscode-600 power amplifier having earned good words both in this august journal as well as in others. The "Minuet-in-A" is a revision of NYAL's medium-priced preamplifier; while its predecessor was also a tube-MOSFET hybrid, the model A has little else in common with it.

The first time I saw the original "Minuet," I thought it looked very cheap. In fact, it reminded me of a 12W Knight-kit amplifier that I bought from the old Allied Radio in Chicago when I was a kid. All of the controls were located on the front of a low chassis (you know, like the way old public-address amps were built); a screen cage, through which you could see the tubes glowing, occupied the area above the chassis beltline. In short, it looked cheap, old-fashioned, and pretty much like a DIY project. The sound of this device wasn't too bad, but the phono stage was microphonic and hissy: using it with low-output moving-coils was out of the question. Nice try, Harvey, but no cigar!

The "Minuet-in-A" is different. It looks good, with its full-height black front panel and white lettering; it also looks very fashionable. It has oodles of switching capability, facilitating the use of two tape decks. The phono stage is very quiet, and when used in its full 80dB of gain mode, full-volume hiss is only barely audible! NYAL also seems to have licked the microphonics problem, as rapping on the chassis while the unit is switched to phono elicits no howl of protest from microphonic tubes.

As implied above, the "A" features switchable phono gain. On the back of the unit, there is a block of four DIP-switches, and, depending upon how these are set, the user can select for either 60, 72, or 80dB of gain. I might add that 80dB is more than enough for almost any low-output MC cartridge (except, perhaps, the Ortofon MC-2000). The phono input also features adjustable loading: under the top cover, one finds a pair of pin sockets for each channel. These sockets will take, without soldering, the leads from ⅛W, 1%, metal-film resistors and/or polystyrene capacitor leads. This allows the user to trim the preamp for proper cartridge loading, a welcome addition to any preamp; in my opinion, this system is superior to other designs where a few loading options are supplied as a built-in, switchable feature.

The NYAL Minuet-in-A is, of course, a "budget" preamp selling for under a grand, but for that money you get first-class sound, with many of the sonic attributes of costlier units. Even though this preamp is a tube/solid-state hybrid, its sound is definitely tube-like. What you do *not* get for the money is the level of fit and finish one expects from the likes of a conrad-johnson or an Audio Research SP-11. For example, the selector switches are inexpensive pushbuttons and I have had some trouble with channels dropping in and out (nothing a little Cramolin won't fix, though). Likewise, the potentiometers are not of the best quality, but they do seem to track well and

are at least quiet. The RCA connectors on the back are the garden-variety tinned type, not gold. Inside, however, one sees premium components where they are needed, and more prosaic parts quality elsewhere. In short, this is an intelligent design, executed for maximum sonic quality at minimum price.

It is important that audiophiles are aware of the pricing of the top-quality components which they seem to demand in their audio equipment. "$5000 for a preamp?," you ask when confronted by an Audio Research SP-II. Well, believe me, ARC is not "price fixing;" it really does cost that much to market a device of the SP-II's quality. You may not believe this, but a 10uF polypropylene capacitor costs about six bucks wholesale in the quantities that small, speciality manufacturers like ARC purchase.

Have you ever thought about how much it would cost to manufacture the old Dynaco PAS-3 preamp using premium parts? Take the PAS-3 *exactly* as it was, and change nothing but the component quality. You know, use Allen Bradley, ALPs, or Noble premium potentiometers, silver-contact switches, gold jacks, metal-film precision resistors, exotic capacitors, that kind of stuff. I have thought this through to the point of actually costing it out. A Dynaco PAS-3, with the same component quality as an SP-II, would cost almost *$600* just to manufacture! This is without doing all of the other things a PAS-3 needs before it could be considered a modern design—upgrading the power supply and redesigning the RIAA phono stage, for example. Since the rule of thumb for electronics manufacturing in the USA is that the selling cost of an item will be six times the manufacturing cost, a premium-quality PAS-3 would sell for close to $4000! Remember, this is a preamp that sold for only about $200 when it was current, and the kit version was substantially less. It seems miraculous, therefore, that a modern design selling for less than $1000 can have the performance that it does have. I hope you can see why a reviewer will sometimes forgive a product for less-than-perfect switches, pots, and connectors—especially if the cost and overall level of performance warrant it.

The sound

One of the big advantages of hybrid designs is the ability to give the consumer the best of both worlds in terms of sonic quality. The Minuet-In-A is a case in point. Like most tube preamps, the NYAL seems to romanticize sound. The highs are liquid-smooth, with lots of space around the instruments. Barber's *Capricorn* Concerto (Mercury SRI 75049) is scored for flute, oboe, and strings, and has one of the best string sections on record. With both MC and MM cartridges, the NYAL reproduces this difficult instrumental ensemble in an exceedingly lifelike manner. The high overtones, far from being truncated as in many tube designs, seem to go on and on, but without the powder-dry texturing that ultra-wide bandwidth solid-state preamplifiers almost always exhibit to one degree or another in this frequency range.

The midrange and upper midrange are somewhat forward in my sample, leading to a slight suckout in the brightness (5-7kHz) range. This is especially noticeable on piano recordings. On the both the CD and LP versions of Telarc's Chopin recording with Malcolm Frager on the Bosendorfer (CD-80040, DIDZ-10030), this slight suckout manifests itself by making the piano notes in the middle and upper ranges sound a little too rounded and pretty. One must remember that piano strings are made of metal and anything which robs the piano of this metallic quality is not doing the instrument justice. This isn't a serious problem, though, and indeed may be sample dependent. It was much more pronounced when the unit was new than it is now, and is slightly worse when the preamp is cold.

The lower midrange on the NYAL is just delicious. Again, strings tell it first. Pierre Fournier's cello in Strauss's *Ein Heldenleben* (my own recording) sounds just right. The resinous quality comes through in perfect proportion, and the weight of the cello seems to exactly complement the overtones. I have seldom heard this done correctly in tube preamps; most have a "squishy" sound in this range, the sparseness of which makes the cello sound flat and somewhat emasculated.

Bass is first-rate. Good CDs are reproduced with the kind of heft on the bottom end that we all expect from them and seldom get. This is another area where the solid-state virtues of the hybrid design shine through. The bass is tight, well-defined, and goes really deep. Prop-

erly loaded—something possible with this preamp—a good moving-coil will give excellent bass performance without compromising its other attributes.

The soundstage

The Minuet-in-A produces a very wide and a fairly deep soundstage, very noticeable on very large orchestral works which have been properly miked (such as the Bruno Walter recording of Mahler's Symphony 2). The chorus on this recording is spread out in a large arc which stretches way beyond the boundaries of my speakers. At least, it sounds like an arc because the NYAL's soundstage becomes quite shallow at the edges and is very deep in the center. This effect makes the edges of the chorus seem close to the apron of the stage, and the middle of the chorus way back behind the brasses. (In my experience, almost all "inexpensive" preamps have this same foreshortening of the soundstage at the edges.)

As far as imaging specificity is concerned, the Minuet-in-A is only fair. I find it easier to locate instruments in space with my Hafler DH-110 (modified).

To summarize

The NYAL "Minuet-in-A" is an excellent preamp at an affordable price. It sounds just delicious, it's quiet, its tubes are supposed to last at least three years, and it's flexible. I know of no other preamp at anywhere near this unit's price that comes within a mile of it sonically. If it has a few flaws, I say, "What doesn't?" Perfection is for the rich. I can listen to an awful lot of great music through this preamp while others are saving their pennies (in silence) for that SP-11. **S**

LINN TROIKA CARTRIDGE

John Atkinson

Linn Troika cartridge

Low-output moving-coil cartridge with unique three-point fixing arrangement. Recommended tracking force: 1.5-1.7gm. Stylus type: Nude Vital superelliptical. Price: $1250. Approximate number of dealers: 100. Manufacturer: Linn Products Ltd., 257 Drakemire Drive, Castlemilk, Glasgow G45 9SZ, Scotland. Distributor: Audiophile Systems, 8709 Castle Park Drive, Indianapolis, IN 46256. Tel: (317) 849-7103.

Some six or so years ago, the Linn Asak cartridge set new standards for imaging and soundstage reproduction. I can remember the first time I heard an Asak in a system using Quad ESL-63s—I had never experienced such depth of soundstage and solidity of imaging from *any* system, and that with Quad amplification! The Asak was relatively quick-

ly overshadowed in this area, however, and in any case, soundstaging precision by itself didn't seem to be a high priority for the Linn design team, who were apparently more concerned with dynamics and a musical integration of the sound across the frequency range.

These two aspects of cartridge performance, precision/accuracy and musicality, often seem mutually incompatible. According to Anthony Cordesman's survey in Vol.10 No.5, such cartridges as the Monster Alpha 2, Clear Audio Veritas, and Talisman Virtuoso DTi seem to maximize information retrieval and transparency, but lose out to the musical presentation of the Koetsu Signature Red, Monster Alpha Genesis, and the Linn Karma. However, like many audiophiles, I want analysis *and* synthesis. I want—no, I *need* both the maximum retrieval of information and the integration of that information in a transparent, musically believable—and stimulating—whole.

My choice of cartridges over the last ten years has reflected either one side or the other of this apparent dichotomy, as the minutiae of my system brought one or the other aspect into relief. On the detail/accuracy front, a Shure V15 IV was succeeded by first a Dynavector Ruby Karat, then a Mission high-output MC, a Technics EPC 205 MM, another Ruby Karat, a Goldbug Miss Brier, and a Dynavector Diamond Karat. Interspersed among those engines of analysis, I enjoyed a Dynavector 20B, a Denon DL103D, two Koetsu Blacks, a Linn Asak, and finally two Koetsu Reds, the last three occupying the sharp end of my system for the last four years.

When Linn introduced their metal-bodied Karma in 1985, it seemed to me to be an attempt to increase transparency and dynamic performance without losing the Asak's consistent musical quality. I never used one in my own system, but in the two systems in which I heard the Karma regularly, I was bothered by its rather upfront presentation when compared with the Asak. Certainly the Karma is a top-rank design; it was just that its view of the music was a little out of sympathy with my own. I suppose you could say that I want a little more romance and soul; Patti Labelle rather than Patti Smith.

In March of this year, however, I visited a friend in England, John Farlowe of Exposure Electronics. His PMS Isobarik system (triamped with regulated-power-supply Exposure amps) produced both a musically satisfying *and* incredibly detailed sound. The coherence of the musical presentation was not compromised by the detailed presentation of the soundstage. (Yes, with Isobariks, no less!) I checked out his front end: there, sitting in the end of his Ittok, in place of the Karma I had expected, was a weird, rose-colored, metal-flake-finished cartridge, fastened to the headshell with *three* bolts. I had had my first experience of the Troika.

Mechanical integrity of the cartridge/tonearm system has always been a major concern of the Linn design team. The Ittok tonearm, for example, was, apart from the rare Swiss Breuer, the first high-integrity tonearm, rivaled today only by the Alphason HR100S, the Rega RB300, the SME V/IV, and the obsolete Sumiko "The Arm." The detachable headshell and azimuth adjustment were sacrificed in the effort to maximize mechanical integrity. Similarly, the Asak's flat body top and the replacement of its plastic body by metal in the Karma and Asaka were steps toward producing the tightest, least reflective/reactive interface between cartridge and headshell.

With the Troika, Linn has pushed this idea even further, but at the expense of universal tonearm compatibility. A webbed extension of the body casting extends behind the body, and a third bolt holds this plate against the headshell. (Not entirely coincidentally, the Ittok headshell has a pilot hole at exactly the right place; it just needs reaming out with a twist drill—easier than it sounds, if you're careful.) In addition, both the standard ½" centered holes and the third hole are distinguished by having a small, raised, circular land around them, machined flat. Though it seems unlikely, Linn claims the total mating area of the three raised "washers" to be greater than the effective contact area of a conventional flat body top.

Apart from the need to enlarge the hole in the Ittok headshell, mounting the Troika presents no real problems. Care must be taken, however, that the cartridge is correctly positioned when the bolts are fully tightened, as the lands indent the undersurface of the headshell, making it difficult to reposition the cartridge if it turns out that the offset angle is incorrect.

The motor is identical to that of the Karma, and a short aluminum cantilever—Linn has consistently shied away from exotic cantilever materials—holds a high-quality nude Vital stylus. A major difference, however, is that flying leads are used rather than the usual pin/lead business, removing one set of electrical contacts from the signal path.

The sound

Auditioning took place with the Troika set up in an Ittok tonearm mounted on a 1987-spec Sondek turntable (with the new laminated armboard), the player resting on a Sound Organization table with spiked feet penetrating the carpet. The player was set up by Audiophile Systems' Sales Manager, Susan Larochelle, to ensure that all was hunky-dory. The rest of the system was my usual combination of Audio Research SP-10 II and Krell KSA-50 amplification, driving Celestion SL600 loudspeakers sitting on spiked Celestion/Foundation stands. Cartridge loading was 100 ohms.

I put on the first record; the stylus hit the groove with a clean "pop." Now I am a big believer in what that first impulsive pop—God's own step signal—reveals. Without even listening to music, you can get an impression of the excitability of the LF tonearm-mass/cartridge-compliance resonance—*ie*, how much bass overhang results—and to what degree the cartridge's HF response will exaggerate surface noise. The Troika seemed pretty well matched to the Ittok, with no bass sogginess to the sound of the stylus hitting the groove, and the resultant groove noise was very quiet, suggesting a good finish to the stone. Once the music started, the first impression was of a much lighter, less "romantic" balance than the Koetsu Red that had most recently occupied the sharp end of my system. Low frequencies had less weight, but there was no feeling of insufficient bass; rather, the low frequencies were "fast," with clearly delineated leading edges to transients from such bass-rich instruments as tympani and double bass. Even with a typically modern, dry, bass guitar sound—where are you Jack Casady, now that music needs you?—on Reuben Blades's *Y Seis del Solar* (Elektra 9 604321-1), the clarity with which the bass was presented compensated for the Latin percussion's lack of kick drum.

This transparency extended up the frequency range. Susan Larochelle had brought with her a record of Britten's two solo cello suites, Opp.72 and 80 (Decca/London SXL 6393). Recorded in 1970, the cello image (a little unstable with frequency; I suspect spaced miking) hangs convincingly between the speakers. This is not just a sterile piece of cartridge soundstaging wizardry, however. Coupled with the precise presentation of the cello's tonality—the Troika allows you to hear without ambiguity when Rostropovich plays with the bow closer to bridge than fingerboard—you are drawn inexorably into the music. When I used to play the violin, I was very much aware that the essential sweetness of string tone was set against the underlying bitter astringency of the sound of horsehair scraping against gut. This tonal contradiction, akin to the fight between tartness and sugar in a really well-made sherbet or sorbet (LA's girlfriend, Laura Chancellor, is a virtuoso sorbet/sherbet preparer), would bring a tight feeling to my throat but is something I very rarely hear from recorded solo strings, the balance shifting too far to one or the other extreme. With the Troika and this Britten recording, my throat tightened appropriately, particularly when Rostropovich played harmonics.

That the Troika is capable of virtuoso soundstage delineation was demonstrated with James Boyk's recording of Beethoven's "Pathetique" piano sonata (Performance Recordings PR-5). One of my favorite test records, it was recorded with crossed 4038 ribbon mikes and, with the best equipment, is capable of creating a soundstage between and behind the loudspeakers possessing, as the Cheapskate would put it, a "palpable presence!" And presence there was, the piano image cleanly delineated between the speakers, not occupying the full stage width, but allowing the recording venue to be discerned around it, though with a little less apparent depth than I have heard before. As with the Rostropovich recording, the tonal integrity of the Troika across the band allowed the complex interplay of harmonics in the Steinway tone to be easily resolved, without any unnecessary exaggeration of detail.

Another favorite test disc is *Light Blues*, a

1982 digital recording by Tony Faulkner (Hyperion A66059). The spatial locations of individual voices in this collection of unaccompanied French part songs were exquisitely defined. A slight presence-band "edge" to the sound resulted in less stage depth than I am used to, but this may be partly due with this LP to a more transparent presentation of the artifacts of the digital recording. That it was more than that, though was revealed with the next disc to hit the turntable, *Amazing Grace* (Atlantic SD 2-906). One of the most uneven live recordings ever, a marvelous sense of "being there" is offset by a consistently over-recorded Aretha Franklin. God knows what microphones were used, the applause sounding more like the congregation was banging shingles together. But no matter: forget "Who's Zooming Who?"—this is the First Lady of Soul showing why she endures while others end up in Lake Tahoe. Although I expected the rather upfront HF presentation of the Troika to make Aretha's closeness to the mike *more* uncomfortable than with the Koetsu's more laid-back presentation, this was not so. Yes, tonally the sound was somewhat forward, but her voice had, I don't know how else I can explain it, "clean edges." I assume this is due to the improved mechanical integrity of the three-point fixing, and was even more noticeable on an LP transfer of a collection of Fred Astaire film soundtracks from the '30s (Columbia PSG 32472).

Probably the ultimate test for a Scottish cartridge is being asked to play some quintessentially *English* music, Ronald Binge's *The Watermill*, on HMV Greensleeve ESD 7063. (You may be more familiar with Binge's '50s smash hit, *Elizabethan Serenade*.) The stereo image is a little pulled to the sides in this 1970 recording, but the Troika, like the Koetsu Red, doesn't exaggerate the size of the central oboe soloist when compared with the swirling cello/bass arpeggios. It remains realistically small, benefiting from the Troika's precise imaging capabilities, and string tone was nothing less than delicious, despite a slight brightness to the sound.

Measurements

Output is about normal for a low-output 'coil (around 0.2mV at 5cm/s), and the SP-10 had no noise problems. With the downforce set at 1.7gm, the Troika just failed the 70um lateral modulation cut on the Ortofon Test Record at 78°F (it was a hot day); with the downforce increased to 1.8gm, it managed to clear the 70um cut, but 2.1gm was needed for the maximum 80um cut. It cleared the 50um vertical cut at 1.7gm downforce. While not quite as surefooted as the Shure V15 V MR (others report that the Troika is less secure at lower temperatures), this is pretty good tracking for a moving-coil design.

Channel balance at 1kHz was spot on at around 0.05dB difference, while crosstalk was both superbly low and symmetrical, measuring -34dB at 1kHz, left-on-right, and -35dB, right-on-left. Obviously, the motor geometry is very well-aligned, both with the stylus orientation and the groove (as should be expected at this price level). The cartridge designers' ability to optimize the compliance for a dedicated tonearm design was shown by the positioning of LF tonearm resonance: ideally placed laterally around 10Hz in the Ittok, and vertically a little lower, around 9Hz, both with only a moderate Q and a moderate amplitude rise. The measured frequency response, using the third-octave noise bands on Denon XL-7007, showed the ubiquitous upper-midrange depression to be 2-3dB or so deep, with a 2dB rise at 20kHz.

Conclusion

Less bass-rich than the Koetsu Red, less romantically balanced, with a more forward upper-midrange presentation, and not quite as capable of throwing as deep a soundstage, the Troika strikes an excellent balance between the analytical retrieval of fine detail and the preservation of the musical structure supported by that detail. Considerably less shrill than the traditional "high-detail" cartridges, in UK parlance it "plays tunes" across the frequency band. If its presentation of tonality is still a little upfront—within that unusual body, it is still basically a Karma—the integration of its performance from bass to high treble enables it to compete effectively with the Koetsu for pole position in my system. Ultimately, the Japanese cartridge edges ahead—I said I'm a sucker for romance—but certainly with the Troika, Linn has again managed to produce a world-beating car-

tridge, even if "world-beating" must be qualified by the fact that it can only be confidently recommended for those using Ittok tonearms. The Ittok, I feel, is still distinguished by a liveliness in the upper midrange that tends to add a sheen of "excitement," particularly to voices. Nevertheless, it is an excellent platform for the Troika, and the complete 1987 Linn player—Troika, Ittok, Sondek LP12—must be ranked as among the world's best.

How does the Troika rank against the $800 Monster Alpha Genesis, which did so well in Cordesman's survey last month? For the answer to that question, you will have to wait until the next issue of *Stereophile*! S

VACUUM TUBE LOGIC 30/30 REVISED POWER AMPLIFIER

Dick Olsher

VTL 30/30 amplifier

Stereo tube power amplifier. Nominal power output: 30W/channel. Sensitivity: 1V. Price: $990. Manufacturer: Vacuum Tube Logic, 8-14 Norwood Road, Southall, Middlesex UB2 4DL, England. US Distribution: Vacuum Tube Logic Of America, Inc., 369 Broad Street, Providence, RI 02907. Tel: (401) 331-6824.

Having swallowed a bitter pill in the form of my original review of their 30/30 power amplifier (Vol.10 No.2), the folks at VTL (David and Luke Manley) took to heart the key issue of competitiveness and decided to offer more for less. First, the US retail price has been lowered to $990 (without grille). That in itself is an important step in the right direction, but more significantly, the sonics have been dramatically improved. The driver-stage topology has been changed to that of the much more expensive VTL monoblocks, namely a two-triode configuration with one 12AT7 dual-triode, operated in parallel for higher transconductance, driving another 12AT7 phase splitter. The output stage has been modified to use 7868 power tetrodes in ceramic/silver base sockets. The power-supply reservoir capacity has been increased by another 400uF, and milspec metal-film resistors are now used throughout the signal path. According to David Manley, this is pretty much the version of the 30/30 that had been built in the UK and marketed in Germany; in order to simplify the construction and cut costs for the US market, however, the design was watered down. Presently, the amp is being built in Massachusetts and the tubes

are sourced in the US, all of which has enabled VTL to perform the minor miracle of improving the product but reducing the price.

Sonic Impressions

The initial impression was one of acceptance. Well before one had a chance to analyze its attributes, there was an intuitive sense of rightness about this amplifier. This was due in great part to its ability to project a nicely integrated sonic picture with liquid mids, nice HF detail, and a quickness atypical of tubed amps. There was good midbass control and detail, although the deep bass lacked punch. For example, the double bass on the "Tiden Bara Gar" cut on the Opus 3 Test Record 1 was tightly reproduced, with very good detail. The 30/30 does well what tube amps generally do quite well: reproduce a soundstage with great conviction.

The VTL proved very capable of revealing spatial information. With the right program material (*eg*, Opus 3 recordings), it reproduced a wide and deep soundstage, with accurate placement of instruments within that stage. The depth on the large panpipes cut (cut 6, Test Record 1 again) was incredible, with the bongos and gourd clearly placed forward in the soundstage. Focus of instruments within the soundstage was, in fact, amazingly good at this price point; the 30/30 rivals much more expensive amps in this regard. The "Insbrook" cut featured excellent specificity of individual voices in the chorus. The timbre of the nylon-stringed guitar on cut B1 was captured very nicely. Vocals generally possessed a spacious, open quality, with just a slight loss of transparency—mainly through the upper mids—and a slight darkish haze/glaze through the upper registers (though these colorations are attenuated after the amp has been cooking for about an hour). Moving on to Cleo Laine's *Live at Carnegie Hall* (RCA LPL1-5015), retrieval of audience feedback was very good. There was excellent focus of Cleo center-stage, and really clean highs—a high compliment for a tubed amp. Bass lines were very easy to follow. But the amp misses out just a bit in terms of midband transparency. All right, so it's not 20/20 in terms of sonic acuity through the mid band; but 30/30 isn't bad, is it?

Very few amps are capable of passing the "Lesley" test: accurately reproducing the timbre of my spouse's voice via PCM-F1 master tapes. I find this to be a very sensitive test. Having heard my spouse "live" for countless years has ingrained in me a very good feeling for her sound, and being familiar with the recording process gives me a handle on what the reproducing chain should output. The VTL joins that select group. There was just a slight roughness in the upper mids and a modicum of opacity in the treble. But on the whole, the performance is competitive with much more expensive amps.

The excellence of the 30/30 did not diminish on CD material. The spacious church perspective on Hildegard of Bingen's "A Feather On The Breath Of God" (Hyperion 66039) was nicely reproduced. The sopranos soared magnificently, there was plenty of hall sound, and ambience retrieval was perhaps 80% of what I think possible from this recording. Kiri Te Kanawa on "Blue Skies" (London 414666-2) was in fine focus, and lots of low-level detail was in evidence—including Kiri's breathing. Dynamic range was quite impressive for such a little baby amp. On the Jarvi Sibelius Second (BIS CD-252), the 30/30 went from soft to loud without any apparent strain or stridency before running out of steam (*ie*, it clips gracefully).

The Verdict

If you have not figured out by now that I really like this amplifier, do not pass Go—go straight to Jail. Yes, I'm going to recommend it, but it occurs to me that I should first elucidate potential applications. A small (by which I mean low-powered) tube amp is not a universal amp, and occupies a rather small niche in the market place. In JGH's listening setup/room, for example, a small amp would be less than useless. However, for my voltage-limited or appetite-limited Quad ESLs, the 30/30 would appear to be a very good match. Also, in small rooms and with speakers of at least moderate sensitivity, the VTL should work quite well. They worked very well with the Dahlia-Debras.

In my opinion, the VTL 30/30 offers a rather large slice of the best there is amp-wise, and does so at a very reasonable cost. I'll go even further—at the asking price, the amp is nothing short of the proverbial steal.

If I were looking to spend a kilobuck on a small tube amp, the VTL would be it. Let it warm up for at least 30 minutes before you do any serious listening, then sit back, relax, and enjoy. **S**

RADFORD STA 25 "RENAISSANCE" POWER AMPLIFIER

Dick Olsher

Radford Renaissance power amplifier

Stereo tube power amplifier. Rated power: 25 vacuum-tube watts nominal into 8 ohms. Price: $2495 (FOB Spring Valley, NY). Approximate number of dealers: 2. Manufacturer: Woodside Electronics, Kimberly Road, Clevedon, Avon BS21 6OJ, UK. North American Importer: Tercel Audio Marketing, PO Box 147, 7723 Younge St., Olde Thornhill Village, Ontario, Canada L3T 3N1. Tel: (416) 731-9708. In the US, 92 South Pascack Road, Spring Valley, NY 10977. Tel: (914) 623-4855.

Arthur Radford is not a household name on this side of the Atlantic, but his role and stature in UK audio are very much akin to that of David Hafler in this country. Both men designed excellent-sounding tube amps in the '60s that went on to become classics; Hafler's designs were marketed, of course, under the Dynaco label. The Radford STA 25 Series III, in particular, gained quite a following over the years with British audiophiles, and ultimately prompted a limited-edition Series IV production run just a couple of years ago. It just so happens that I own a Radford STA 25 Series III amp. It's old and decrepit, with many of the solder joints badly oxidized, and has spent so much time being fixed or upgraded with better caps that I've been unable to listen to it very much. But I've been able to glean a reasonable impression of its capabilities.

The Radford sound is classic tube sound: vivid, three-dimensional soundstaging, sweet and liquid mids, and a lot of euphonic colorations. It isn't exactly a reference tool, but is very enjoyable, nonetheless. In its day, the old Radford was quite a sophisticated design, and is still the most esoteric little tube amp around. An EF86 pentode driver stage fed Radford's patented triode-pentode phase splitter, with a push-pull EL34 output stage. One reason for the sonic success of the old STA 25 lay in the handcrafted output transformer, which probably represented the state of the art in its day. I'm told that the same tooling is used to produce the output transformers for the Renaisssance, and that the same lady who used to wind them in the old days is still doing the job. That's nice!

The Renaissance is, as the name implies, not an exact duplicate of the old Radford, but instead attempts to instill several circuit refinements into the framework of the original design. The input stage has been changed from the EF86 pentode to a cascode-coupled double-triode configuration using the 6DJ8 tube. The output stage has been changed to 6550 tubes with a rail voltage of 450 volts; this ensures a really easy time for them, and consequently a very long tube life. (All of the tubes in this amp are claimed to last as long as the paint on the chassis!) The output stage is of ultralinear design and runs in the class-AB mode, with negative bias kept to a minimum consistent with stability and linearity. The transformer output tap of the Renaissance is fixed at 6 ohms. The power rating of the amp is quite misleading. While rated nominally at 25W, it is actually capable of significantly higher power outputs.

Construction quality is high, with good-quality parts much in evidence. All signal paths are hardwired, and star-earthing is employed. The Renaissance is manufactured by John Widgery and Woodside Electronics with Arthur Radford's blessing (John being an old friend and former Radford employee). A pre-production sample was shipped directly to me for audition, which turned out to be a disappointing experience. It simply failed to live up to the expectations I had for a $2495 amplifier. It did not irritate, to be sure, but it failed to score well in the very critical categories of transparency, resolution of low-level detail, and transient delineation and speed. After discussing the matter with Barry Falcon of Tercel Audio (the North American importer), I was informed that the production version of the amp was slightly improved physically in the areas of power-supply and negative-bias circuitry, but that the sonic improvement was reportedly quite substantial. I agreed to wait for a second sample.

Indeed, the production sample (SN 019) was much better sonically, except for a slight AC mains hum. However, I'm afraid it still does not cut it at this price point; there are too many colorations and weaknesses. Here's a sampling taken from my listening notes: kick drum was anemic, lacking both weight and power; brass was slightly tizzy; there was a slight loss of focus and resolution of low-level detail through the mids; Cleo Laine's voice on the track "Send In the Clowns" (RCA LPL1-5015) was slightly phasey, being stretched across the soundstage to unnatural proportions; the extreme treble was closed-in; and brushed cymbals were slightly steely. On the "Lesley" test (my spouse's voice on tape), timbre alteration was noticeable, with a thickening and darkening of textures. Loss of clarity and purity in Lesley's upper registers was also noted. This is just a sampling. These problems were consistently present on a variety of program materials, both analog and digital. There were a few "bright spots." String tone was consistently sweet, maybe overly sweet, but kind of nice anyhow. The lower mids are sugar-coated, textures being thick and sweet. Again, very pleasant.

Soundstaging was very, very good, with lots of width and depth, but localization within the soundstage was not entirely accurate or well focused. There was a slight bunching up of left and right center material toward the extremes of the soundstage. For example, Harry James's trumpet (Sheffield Lab 3) was displaced to extreme left, almost at the left speaker.

I am at a loss to reconcile my listening impressions with what I know to be an honest and well-built product. The bottom line is that the Renaissance is entirely too colored to compete at anywhere near its asking price. It's a fun amplifier to be sure, sort of like listening in Technicolor, but I certainly cannot recommend it. **S**

MIRROR IMAGE 1.1S POWER AMPLIFIER

J. Gordon Holt

Mirror Image power amplifier

Solid-state stereo amplifier. Rated power: 200W/ch. into 8 ohms; 400W/ch. into 4 ohms. Power bandwidth: 1Hz-1MHz. Distortion: Less than 0.025%. Quiescent power consumption: 350 watts. Dimensions: 19" W by 7½" H by 18" D overall (includes 2" D for front handles). Weight: 47 lbs. Price: $2995. Approximate number of dealers: undisclosed. Manufacturer: Mirror Image Audio, Ltd., 700 Springvale Rd., Great Falls, VA 22066. Tel: (703) 759-4774.

I never cease to be amazed at the courage—or is it foolhardiness?—of those who launch new companies in a field already so overcrowded with superb products that there would seem to be no demand for another one. Yet it happens all the time. Just a few years ago, the high-end power amplifier leaders were Krell, Threshold, Audio Research, and Mark Levinson. Yet since then, we have seen first products from lean-and-hungry new companies called Rowland Research, Jadis, Boulder, Wingate, and now, Mirror Image. None of these firms has yet to threaten the ascendance of the established names (although Dave Wilson, of Wilson Audio, chose the Rowland 7 as his standard amplifier for the WAMM system), but that doesn't stop them trying. The latest aspirant, Mirror Image, comes as close as any to date.

Mirror Image makes four models of power amp: the 1.1S, 1.1S+, 1.9S, and 1.9S+. The 1.1S and 1.9S units differ only in rated power output: 200 watts/channel for the 1.1S and 325 watts for the 1.9S, into 8 ohms. (Into 2 ohms, they are rated at 750W.) The "+" designation refers to separate power transformers—dual-mono designs, as distinguished from the "basic" models which share a single power transformer.

All models use hefty toroidal power transformers, a heavy solid-copper buss in the high-current circuits (such as between the power supply and the speaker terminals), and lots of regulation and isolation. Except for the power transformer, all other parts are on two identical left and right circuit boards, and the amp's internal construction is very impressive looking—nicely laid out and ruggedly put together.

Reading the manufacturer's description of my sample 1.1S gave me a spooky case of *deja vu*. It reads almost exactly like Rowland Research's description of their Model 7, even to the DC-coupled circuitry that won't pass DC for more than a second because of a built-in output offset-correction circuit. The amplifiers even *sound* remarkably alike.

The design of the 1.1S is basically straightforward but functionally complex. Each channel consists of an FET input voltage amp, a bipolar voltage amp, and a bank of MOSFET output devices. The complexity lies in the support circuitry, which sets all the operating parameters and holds then in place despite changes in line voltage, operating temperature, and so on. As mentioned above, the entire signal path is direct-coupled; there are no capacitors in the signal path. Direct-coupling

has the immediate advantage of eliminating capacitor "sounds" from the signal, but it puts the loudspeakers at high risk—particularly from a powerful high-current amplifier like this—by converting any small DC offset at the amplifier inputs into a destructively powerful DC offset at its outputs. With 30dB of available gain, a mere tenth of a volt of DC at the amplifier's input becomes 3 volts of DC offset at the output. This can translate into a current of 3 amperes through the woofer voice-coil (whose DC resistance is a small fraction of its rated impedance), and very few voice-coils can take that for more than a short time without burning up.

Mirror Image's twofold solution to this potential problem is essentially the same as Rowland's: Use a servo circuit to eliminate small DC offsets at the output by applying a countervoltage at the input, and an offset timer and a muting circuit to shut down the amplifier if an uncorrectable offset lasts longer than a second. Since the timer/muter is in a sidechain paralleling the signal path, it has no effect on the sound when not actually muting, and thus provides all the sonic benefits of DC coupling without any of its hazards.

Despite this safety feature, Mirror Image's instructions warn the user several times about the destructive potential of their amplifier. In fact, they make such a point of it that I felt fortunate in having used the 1.1S for three weeks without having anything blow up in my face. (Under "Trouble Shooting" in their manual, the symptom "Speakers blown" is followed simply by the helpful comment "We warned you!") It's true that the Sound Lab A-3s I'm using are particularly rugged speakers (having no tweeters, in the conventional sense), and I didn't do anything stupid like setting the stylus down on the outer edge of a disc, but nevertheless, the repeated warnings in the instructions made me rather uptight about using the amp. It's always better to be cautious than careless, but my personal experience with high-powered amplifiers has been that they are *less* likely than lower-powered ones to destroy loudspeakers.[1] (A friend of mine recently burned out four woofers in a pair of speakers rated at 200W/channel with an amplifier rated at 70W per, by playing Telarc's *1812 Overture* at an intemperate volume. I *did* warn him.)

The 1.1S's instruction manual left something to be desired, "preliminary" or not. It runs to 23 pages and is generally well written and useful, but has neither an index nor a table of contents. Worse is the fact that the manufacturer includes neither his address nor phone number. The address is on the warranty card, but once you mail that in, the only way you can contact Mirror Image is by looking up their listing in *Audio* magazine's *Annual Equipment Directory*. (Or by referring back to this report.)

The instructions also mention that, because of the amplifier's extremely wide bandwidth, it may oscillate when used with certain high-capacitance speaker cables. And just in case this should be a problem, special output filters are supplied with each amplifier. Each filter consists simply of a 5 ohm resistor and a 0.1uF capacitor in series across the output, and plugs between the amplifiers's dual 5-way binding posts and the speaker plugs. I was unable to induce the amplifier to oscillate, so I'll have to take Mirror Image's word that the filters work as claimed.

The MI 1.1S has a very effective signal-muting circuit that operates instantly when you turn the amp on or off. On turnon, it prevents charge-up thumps (the amp produced only a faint thud); on turnoff, the signal is muted again, instantly, allowing one to change input connections or otherwise mess around with the system without having to stand and twiddle your thumbs while the storage capacitors discharge. (One amplifier I recently tested took almost four minutes to go dead, and it didn't fade out; it retained its full sensitivity for that period, then died suddenly. This was a tribute to its supply storage capability, but a pain in the butt when I needed to disconnect its inputs.)

MI's instructions also include a stern caveat against shorting the loudspeaker terminals when the amplifier is turned on. You shouldn't, even when no signal is being fed to it, because serious damage can result. You may wonder what harm it could possibly do, but the answer relates to the amp's DC coupling and offset correction. If there is any DC offset *at all* at the amplifier's inputs—and many preamps will present it with at least some offset—the

1 See "Consultation" elsewhere in this issue.

amplifier will try to produce an equivalent offset at the shorted output. Since the short prevents any voltage from developing across the output terminals, the amp will dump its entire current capability into trying anyway. Obviously, something's gotta give! At the very least, it will blow the amp's internal power-supply rail fuses. (Output shorting does no harm to a tube amp or a non-DC-coupled solid-state one, as long as it is not being driven with signal.)

The 1.1S has several indicator lights on the front panel, which show 1) muting due to an uncorrectable DC offset at the inputs, 2) shutdown due to overheating, 3) positive-going or negative-going overload clipping, and 4) the presence and location of a blown power-supply rail fuse. An overheat shutdown, which according to the manufacturer should never occur unless the amplifier is starved for ventilation, leaves the cooling fan running, and operation will resume automatically as soon as the fan has brought the affected heatsink back down to its normal temperature. Nonetheless, an overheat shutdown should be investigated to make sure nothing is blocking the ventilation path.

Before getting into a nit-picky discussion of the MI 1.1S's sound, I should explain where I'm coming from on power amplifiers. Through many years, I have observed the sounds of tube amps and solid-state amps becoming increasingly similar. Just a few years ago, it was easy to characterize any tube amp as being warm, rich, sweet at the top and forward in the upper middles, while being deficient in deep-bass energy and reproduction of inner details in complex material. A solid-state amp, on the other hand, could be generalized as having a deep, solid low end, a rather cool sound, and somewhat over-etched detail with an excessively crisp and rather edgy high end. Today, the same characterizations still apply, but the differences, in some cases, have become so slight that drawing distinctions is more of an exercise in hairsplitting than a real help for potential equipment buyers.

I have mentioned in these pages before (and offended some insecure audiophiles by so doing) that it is impossible to know with certainty what a power amp sounds like, because the only way you can listen to it is through a loudspeaker, and the only way you can evaluate a loudspeaker is to listen to it with an amplifier. As for amplifiers, I accept certain things on rather shaky faith, and one of them is that The Ultimate Truth in the area of spectral balance probably lies somewhere between the best tube sound and the best solid-state sound. Knowing the limitations of power transformers, I would guess that the "correct" LF quality combines the heft and extension of the best solid-state amps with the best bass detail and impact any of them have achieved to date. I also guess that the "correct" mid- and upper-range contribution lies about midway between that of the least forward tube amps and the least laid-back solid-state amps. And I have long considered the extreme high end of the very best tube amps to be the paragon of absolute perfection, if not the next best thing.

There are indications that most serious loudspeaker designers have been thinking along similar lines, because the less compromised the loudspeaker system, the better it tends to sound on those amplifiers which meet the abovementioned criteria. I have adopted the Threshold dual-monoblock SA-1s as my "reference" amps because they meet these criteria as well as any I have found, as well as doing everything else superbly. I use the Sound Lab A-3 loudspeakers because they produce from the SA-1s what sounds to me like a pretty good semblance of live music. But the comments which follow, about the sound of the Mirror Image 1.1S, should be read with the understanding that they relate to what I hear from the SA-1s, and the average of other top amplifiers, rather than as absolute statements of fact.[2]

First off, the MI 1.1S has as magnificent a high end as any amplifier I have heard! It combines remarkable detail, crispness, and focus with the kind of sweetness, openness, and delicacy I have always admired from the best tube amps. Only one other solid-state amplifier that I have heard, the Rowland 7, has equalled it in that area—for $4500 more!

Otherwise, I found the Mirror Image 1.1S to be somewhat colored. Its upper-middle and lower-high ranges are fairly recessed, although not enough to make the amplifier sound laid-

2 Any equipment reviewer who tells you he knows the absolute truth about the sound of an amplifier or loudspeaker is lying or kidding. He doesn't.

back. Instead, it gains a little richness and warmth, which benefits large-scale classical works but doesn't help pop or rock material at all. And, compared with my SA-1s, the 1.1S's bass is distinctly lean. This is largely offset by a strong impression that it goes substantially deeper and has better detail and pitch delineation. In other words, while the 1.1S's bass differs from that of my reference amps, there is no way I could describe it as inferior. It's just different, and will need a fatter-sounding loudspeaker in order to sound satisfyingly full at the low end.

Soundstage breadth and depth from the 1.1S are both slightly compressed, in comparison with the best I've heard (which include the SA-1s). Some recordings threw definite and stable images 'way beyond the physical limits of the loudspeakers, but for reasons I cannot explain, the same thing did not seem to happen to hall ambience. A wider soundstage is almost certainly something you would get with the dual mono "+" versions of the Mirror Image amplifiers (because of improved power-supply isolation), but I don't know whether they would render depth any better than the 1.1S did.

Inner detailing is excellent, on a par with that from the SA-1s and Rowland Research 7s, both of which cost a respectable $7500 a pair. Through the entire listening session with the Mirror Image, though, the prevailing impression I got was of tremendous smoothness and ease. The amplifier is a delight to listen to, and even though it has only about twice the power of the SA-1s (good for about 3dB of extra output before clipping), it gave an impression of having virtually limitless output capabilities. (Never did I see the front-panel clipping light come on, even at the loudest levels I ever listen at—around 103dB. And that is with *very* inefficient loudspeakers.)

The cooling fan, while quite unobtrusive, raises the noise floor in a very quiet room, but only very slightly. As long as you don't sit within a few feet of the amp, the muted rushing noise is easily swamped by the background noise of any analog signal source. (It is not swamped by CD noise.)[3]

At almost $3000, the Mirror Image 1.1S has some very stiff competition, much of which we have not as yet tested (and may never). Of the ones we have, the most worthy contenders would probably be the BEL 2002 (lean low end, neutral mids, very good highs, slightly vague imaging) and the Krell KSA-100 (one of the most tubelike of solid-state amps—warm, a little rich, somewhat recessed, with great depth and spaciousness, softly silky highs, and superb LF extension with very good control). Clearly, none of the three is an unqualified winner, but they are sufficiently different in sound to afford a wide variety of choice in loudspeakers for use with them. (My reference amps, at $7500 a pair, are hardly comparable. Neither is the similarly priced Rowland 7, yet the Mirror Image comes surprisingly close in some areas to equalling the Rowland's performance.)

The Mirror Image 1.1S does not "redefine" any standards of sound quality. What it does redefine is our concept of value for money: it offers very high power capability and close-to-state-of-the-art sound at less than half of the price one would expect. Would I recommend the Mirror Image 1.1S to my best friend? Without a doubt, but subject to the usual caveats about not mating it with loudspeakers that have the same kinds of colorations. **S**

3 Mirror Image has already announced the availability (free on request for present owners) of a fan-muting pad which reduces the noise "appreciably."

Incidentally, in case anyone is interested, the ambient noise level in my listening room, with all electrical devices turned off, measures 42dB. This is slightly less than the 44dB level of a very quiet intake of breath a foot from the General Radio 1565-A SPL meter.

A TALE OF TWO CD PLAYERS

John Atkinson spends time with the Accuphase DP-80/DC-81 and Stax CDP Quattro CD players

Accuphase DP-80/DC-81: Two-box CD player with infra-red remote control. Specifications: Frequency response: 4Hz-20kHz ±3dB. THD: 0.002% (1kHz), 0.008% (20Hz-20kHz). S/N ratio:

106dB. Channel separation: 100dB. Output voltage (DC-81): 2.5V balanced/unbalanced (fixed), 0-2.5V, variable. Output impedance (DC-81): 50 ohms balanced (XLR sockets), 50 ohms unbalanced (RCA sockets), 1250 ohms variable (RCA sockets). Interface between the two boxes is to the serial digital-audio data standard; connection is via optical fiber or coaxial cable (75 ohms, 500mV p-p). Dimensions (each box): 18 23/32" (475mm) W by 5 5/16" (135mm) H by 14 11/16" (373mm) D. Weight: 33.1 lbs (DP-80), 34.2 lbs (DC-81). Price: $8000. Approximate number of dealers: 30. Distributor: Madrigal Ltd., PO Box 781, Middletown, CT 06457. Tel: (203) 346-0896.

Stax CDP Quattro: Single-box CD player with infra-red remote control. Specifications: Frequency response: 0.7Hz-20kHz ±0.5dB (Direct output); 0.4Hz-18kHz (filtered output). THD: less than 0.002% (0dB, 1kHz); less than 1.6% (-60dB, 1kHz); less than 12% (-80dB, 1kHz). S/N ratio: over 104dB. Channel separation: more than 100dB (5Hz-20kHz). De-emphasis deviation: less than ±0.5dB. L/R channel phase difference: 0 degrees, 5Hz-20kHz. Output impedance: 1k ohm. Output voltage: 2.1V. Dimensions: 443mm W by 165mm H by 290mm D. Weight: 8kg. Price: $2495. Approximate number of dealers: 20. Distributor: Stax Kogyo Inc., 940 E. Dominguez Street, Carson, CA 90746. Tel: (213) 538-5878.

It is now five years since the initial launch of Compact Disc in Japan, and, as mentioned by the Audio Cheapskate elsewhere in this issue, we are well down the road to the day when the CD replaces the LP. Yet, even having lived with CD for nearly all those five years and built up a collection of nearly 300 discs, I still have doubts about the medium's ability to communicate the musical message. It seems that when you know either the music or the performance well, as with nearly all classical works, then everything is fine: it is possible to enjoy the sound from CD. Yet with unknown music, such as a new rock or jazz recording, the music is less accessible, almost as if the listener has to do more work to appreciate what is going on. There seems to be a threshold effect, as in chemical kinetics, where even an exothermic (energy-emitting) reaction needs some energy to be supplied before anything literally catches fire. With CD replay, the listener is committed to a greater mental workload before the music can have the desired emotional effect.

This threshold effect seems not only to be different for different kinds of music, but also for different people: LA, for example, has almost never been drawn into the music from silver discs, whereas I think it fair to say that JGH gets a considerable amount of musical enjoyment from CD. Yet with LP, even with a modest player, there is *no* such effect. All audiophiles must be familiar with the chain of events in which putting on even a totally unknown LP leads inevitably to another occupying the turntable platter, then another, and another. With black disc, the listener's defences are down; he or she is wide-open to the music.

Whether the threshold effect is due to the silver disc revealing more clearly the inadequacies of professional recording equipment, to the problems of consumer-priced digital electronics, or to fundamental errors in the implementation of digital audio, I wouldn't like to say. There is no doubt in my mind, however, that this effect exists. It explains, for example, why classical music enthusiasts are more forgiving of CD replay. But one way to check it out is to audition CD players intended to decode the digital signal as accurately as possible; if the effect persists, then the problem must lie with the source material. I therefore obtained samples of two players said to be (by those whose ears I respect) among the best-sounding models: the two-box Accuphase DP-80/DC-81, and the Stax CDP Quattro, both products from smaller, enthusiast-run Japanese companies.

Up to that point, I had been using two players, the Meridian Pro-MCD and the Mk.II version of the California Audio Labs Tempest. The English player was almost always enjoyable (in CD terms), but at the expense of detail. The tube player, the first version of which I enthusiastically reviewed for *Stereophile* exactly a year ago, was true to the spirit of the music and excelled in the presentation of soundstage, but, despite my initial willingness to accept its mechanical idiosyncrasies, that plastic Philips drawer mechanism really started to irritate after a while.

I started off the audition feeding the two CD players into the line-level inputs of the

my Audio Research SP-10, the preamp being connected to the power amplifier by about 15 feet of Monster Interlink Reference A. There was no sense of anything untoward, but when I plugged either player *directly* into the power amplifier, there was a disturbingly noticeable increase of transparency. All the auditioning for this review was therefore done with both players connected directly to the Krell KSA-50 power amplifier, either via an experimental spaced LC-OFC cable loaned by Stax representative Michael Detmer, or via one meter of Siltech. (The steel locking sleeves were removed from the WBT phono plugs; their presence was noticeable, particularly with the Stax player.) The Accuphase was auditioned via its variable output; the Stax via an audio-quality 10k ALPS potentiometer.

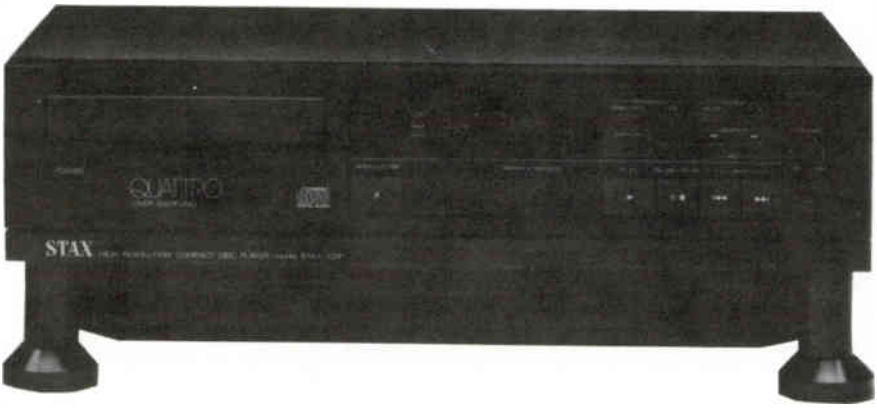
Speakers were Celestion SL600s, connected to the amplifier with Monster M1 speaker cable. Each player sat upon an improvised table, consisting of a Sound Organization turntable stand (spiked to the floor through the carpet), upon which rested a sheet of thick glass, called the Focal Point and supplied by Michael Detmer. Michael also helped me determine the polarity of my mains sockets by connecting one lead of a high-impedance crystal earpiece to the player chassis, holding the other in one hand, and listening to the hash produced. The player plugs were oriented so as to minimize the noise in the earpiece. Both players were powered up for 48 hours prior to any critical listening. The Stax, in particular, improved significantly after being left on overnight, almost completely losing a slight hardness to the sound.

The new CBS CD1 test disc was used to check out the objective performance of the players. As well as the usual test signals to examine frequency response, de-emphasis error, channel separation and S/N ratios, this has unique test signals to check both the low-level linearity and the monotonicity of the players' DACs. In particular, it features a fade on a 500Hz tone from -60dB to well below the noise floor, the tone still being detectable at -120dB. Not only is this 24dB *below* what was felt to be the theoretical cutoff point for a 16-bit digital system, it is actually a *pure* sinewave, despite apparently having insufficient coding levels to be described at all! This is achieved by using digital dither on the signal cut on the CD so that, at the penalty of a slightly increased noise floor, the severe distortion which would otherwise be mandatory for such low-level signals is turned into broadband noise, leaving the signal pure.[1] Checking the low-level monotonicity of the players' DACs requires a digital storage 'scope; unfortunately, *Stereophile*'s had yet to arrive at the time of writing. The disc was useful, nonetheless: listening to the fade track gives a good subjective idea of a player's low-level linearity, any distortion heard being due to the player, *not* to the digital process. I feel this disc will become increasingly important in the testing of CD players, at least as far as *Stereophile* is concerned.

Stax CDP Quattro

The Quattro starts life as a latest-generation Yamaha chassis. Stax adds their own front, top, and side panels in discreet black, and a bottom tray, the latter constructed from a nonferrous material (aluminum and wood) to minimize magnetic hysteresis distortion. Four legs at the corner of the unit provide the necessary ground clearance, each with a height-adjustable foot. The sprung Yamaha transport, with its three-beam pickup and track-selection/programming logic, are left intact, but that's about it. Although the Yamaha board is still present, it is only used to control the transport, extract the digital data, and perform the eight-to-fourteen demodulation, 4x oversampling, and digital filtration. Although the original Burr-Brown DAC used by Yamaha in time-shared mode to decode *both* channels has been left on the Yamaha board, the 176kHz stream of digital data is taken to a new Stax DAC/analog board in the tray. This is completely independently powered: not only is there a separate power transformer, but a second mains cable is taken to the wall outlets; Stax recommends that the plugs for the digital and analog sections be placed as far apart as possible. Stax uses two Burr-Brown 16-bit DACS, one per

1 For full details of the rationale and theory underlying these new test signals, see "On the use of computer-generated dithered test signals," AES Preprint #2396, by Robert Finger. The disc is available from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458, and from CBS Special Products, CBS Inc., 8th Floor, 51 West 52nd St., New York, NY 10019, priced at $45 plus $3 handling.



Stax Quattro CD player

channel, followed by a high-quality sample-and-hold chip and the two analog output stages.

Two outputs? Yes, the user chooses between a conventional low-pass filtered output—using a well-behaved third-order Bessel-type filter—and one which is basically the raw pulse output from the DACs, stretched into a staircase by the sample-and-hold circuitry. Stax makes a good case that, with the transverse digital filter removing most of the ultrasonic spuriae and the natural sin(x)/x HF rolloff given by the aperture effect of the sample-and-hold circuit (see "Zen & the Art of D/A Conversion," Vol.9 No.6), further low-pass filtering to reconstruct the analog waveform is unnecessary. There is also a switch on the rear panel which inverts the digital data stream, resulting in absolute phase inversion. As with the Meridian Pro, carrying out the polarity change in the digital domain ensures that anything heard will be due to the inversion and not to some aspect of the analog circuitry.

Construction is excellent: Ero polypropylene-dielectric caps; leadless, surface-mounting resistors; high-quality, gold-contact relays for muting and de-emphasis; and all-discrete output circuitry—no op-amps here! Selecting a track number and pressing Play on the infrared remote control closes the drawer and starts play; pressing Stop once puts the player into pause; pressing it twice stops the action. Apart from that idiosyncrasy, all functions and facilities that one could want are supplied and self-evident in their operation.

Sound quality: As might be expected from a company that, in my opinion, makes the most transparent headphones in the world—the Lambda Pro—the sound of the Quattro, auditioned via the Direct sockets, is immediately identifiable by the wealth of detail apparent. This is a highly transparent sound. On the *HFN/RR* test CD, for example, there is a recording of my ex-Deputy Editor at *HFN/RR*, flautist Ivor Humphreys, playing Gluck's *Dance of the Blessed Spirits*, accompanied by a Steinway piano. The recording, engineered by Martin Colloms and Mike Skeet, was made in a small London concert hall during a torrential rain storm. While the sound of the rain striking the glass-dome roof of the hall, though soft, was nevertheless quite audible on the PCM-F1 master tape, I rarely hear it on CD. For whatever reason, with most players it is just not audible. With the Quattro, however, you not only hear it; you also know that it is not electrical noise, but has a texture all its own. You can also hear that it is less audible with the Calrec Soundfield mike than with the pair of 4038 ribbon mikes!

This detail may be true to the music—when a clarinettist blows, there is always a small amount of air that escapes his embouchure, and the Stax lets you hear it as air and not as additional noise. On the other hand, it may not be true—I think I was made more aware of multimiking anomalies and poor edits during the time I used the Quattro than with any other player, including the Accuphase. But with multimiked recordings respectful of the notes, the Stax allows the

music to communicate. The Ashkenazy Sibelius Symphony 2 (London 410 206-2), for example, though one of my favorite performances, is a very early digital recording, dating from 1979. With the Stax, you can hear that the digital artifacts *are* artifacts and can therefore be ignored, allowing you to get into the rest of the sound. It is almost as though the things wrong with the sound are presented on a different plane.

And what a sound! This player's high degree of transparency, coupled with low frequencies that are deep, powerful, and well-defined, enables such stirring moments as the *divisi* pizzicato double-bass passage at the end of the Sibelius's third movement—*poppa pumpum Pum, pa pum, pa pa pa pum, pa pum, PA PA PA PUM, PA PUM!*—or the declamation by the brass choir two-thirds of the way through its first movement, to be effortlessly presented. This player produces true Kingsway Hall sound—brilliant, bass-rich, and exciting! Regarding the brass passage, inexpensive CD players render this as the sound of a giant harmonica: as the instruments play louder, their tone colors start to run into one another. Not so here; trumpet, trombone, tuba, and horn are all distinguishable within the choir, each with its tone color intact.

Tonally, that transparency lets you hear through the hi-fi aspects of the recording. At CES, Peter McGrath kindly gave me an early copy of the first CD on his Audiofon label: Ivan Davis playing Grieg piano music. You will probably be aware from my writings in *Stereophile* that I have philosophical difficulties with Peter's preferred spaced-omni mike technique. But who cares, when the result is as musical as this Audiofon CD played on the Quattro—a weight and clarity to the piano sound which I have rarely heard. (This disc is worth getting for the *Holberg* Suite alone.)

The Quattro's soundstaging is deep and wide, though apart from those rare records with sufficient information in the grooves, it doesn't extend beyond the speaker positions. To expect a system to do this automatically, as is sometimes implied in other magazines, is to misunderstand the nature of stereo recording. But with recordings possessing information that enables the soundstage to be realistically decoded—the Marni Nixon Gershwin song collection, for example (Reference Recordings RR-19CD)—there is a delicacy of image presentation and apparent depth rivaled only by the tubed Tempest player.

Lest you think the Quattro faultless, it is not kind to recordings that are already fizzed up in the treble. I admit it, I am a big Dolly Parton fan, and when she joined forces with Emmylou Harris and Linda Ronstadt for *Trio* (Warner Bros. 9 25491-2), I couldn't help myself; I rushed out to buy it. And, losing control, I bought, not the good old reliable LP, but the CD! Yes, I know, I should have known better, but it is an analog original and it did say "Analog and digital mastering by Doug Sax" on the sleeve; I thought that his involvement would have added some kind of class. Well, to judge from the quality of the highs, it's time for engineer and producer George Massenburg, whose name is emblazoned everywhere on the disc, to send out for new hearing-aid batteries. You want lower treble boost on the voices? You got it! You want Bill Payne's piano on your lap? Here, take it. (Oops.)

I may be exaggerating a little, but I find this disc to be virtually unlistenable much of the time; a tragedy. Contrary to all the reviews I've read, there is some fine material here, including Kate McGarrigle's "I've had enough" and Linda Thompson's "Telling me lies" —true tear-jerker country singing. Putting the Accuphase to one side for a moment, the Stax gets closest of any CD player I have heard to allowing me to get off on this recording—you just don't hear English women singing with such a sob in the throat. But it is still unforgiving of the recording's bright character, and if it wasn't for Dolly, Emmylou, and Linda, I would put on another record.

Measurements: The maximum output level from the Direct sockets was dead on spec at 2.1V, but very slightly higher from the Filtered outputs at 2.13V. Frequency response was essentially flat from 31Hz to 8kHz, being 0.9dB down at 4Hz, 0.8dB down at 20kHz, via the Direct outputs, and -1.8dB at 20kHz via the Filtered outputs; nothing that would indicate any exaggeration of recorded brightness. De-emphasis error was a little greater than I would have liked, at +0.4dB at 4kHz and -0.4dB at 16kHz via the Direct outputs; this

would be noticeable in a direct comparison. Track access was reasonably quick, taking 5s to reach Track 15 on the CBS disc, while error correction was effective, not being fazed by the artificial fingerprint or "black-dot" tests on the Philips 5A disc, and muting only very occasionally on the 800 and 900um interrupted-data test tracks.

Trying to check level error at very low levels proved problematic via the Direct outputs, as relatively high levels of very-high-frequency noise led to misleading readings. Checking with an analog scope proved fascinating, as the Direct-output waveform had a "join-the-dots" appearance compared with that from the Filtered outputs, owing to the lack of analog low-pass filtering. Via the Filtered outputs, level accuracy appeared good, with around 5dB of compression apparent at -90dB. This is a very hard measurement to make with noise present, however—we are talking about fractions of a millivolt—and it proved impossible to check the shape of the -90dB waveform, buried as it was within the ultrasonic noise. However, I feel that the Stax player's low-level performance is excellent, as listening to the 500Hz fade revealed *no* apparent harmonics or heterodyning-type whistles; the signal sounded very pure and just faded into the noise. Although the noise level is higher than I would like, being fairly audible behind the -60dB starting point of the tone, it is innocuous, and has obviously been traded off against the excellent low-level distortion performance.

Conclusion: Compared with the similarly priced CAL Tempest II, Stax's Quattro offers a more transparent, more detailed presentation of the music, while not quite matching the tubed player's musical presentation of the soundstage. Its slightly upfront character and total transparency to whatever is wrong with the recording will make demands on the rest of the system in which it is used—if you only have a cart to be pulled, you needn't buy this thoroughbred—and the quality of the software seems more critical than with the Accuphase. With recordings that have true high-end sound engraved within the bumps/pits, however, the Stax Quattro will reveal more music than you thought could possibly survive the rocky path from microphone to silver disc.

Accuphase DP-80/DC-81

This is the two-box $8000 player that caused LA to gibber in Vol.9 No.8: "For the first time, this digiphobe wanted to hear CD after CD. For the first time, I marveled at low-level detail in abundance. For the first time, I was tempted to remark that CD reproduction . . . reminded me most of master tapes." What *is* this miracle-worker among microprocessor-controlled machinery? Can *any* CD player be worth the price of 40 old Magnavoci, 12 new Sonys, or even three Stax Quattros?

The front end of the player is the DP-80 CD transport. Big by CD-player standards, the DP-80 is solidly constructed and beautifully finished, with a full-width hinged panel at the bottom of the front panel concealing the programming controls. A red-LED display shows track and timing information, and an infra-red remote control duplicates the front-panel controls. (This is of the more logically ordered type, where Stop means stop and a separate Pause button is provided. Selecting a track number closes the drawer if open, and puts the machine into play without Play having also to be pushed.) As one might expect, all the programming facilities one could wish for are provided.

Inside, the unsprung transport is to the left, in front of a large toroidal power transformer. To the right is a single large single-sided pcb which retrieves the data, applying a little massage along the way. The data off-disc are error-corrected and eight-to-fourteen demodulated (when the original 8-bit data words are recorded on the master disc, they are translated to 14-bit words so that, among other things, the low-frequency content of the RF signal is reduced), using LSI chips from Accuphase and Sony. I also suspect a Sony origin for the fast-access, metal-chassis transport. The DP-80 is not in itself a complete player, as the only outputs are digital. The resultant stream of data, with left- and right-channel information interleaved, is available via both a conventional RCA phono socket on the rear panel, or an optical-fiber output.

The DC-81 D/A processor is the same size as the DP-80, and styled visually to match, with rosewood endcheeks and a gold-colored anodized aluminum front panel. A full-width

Accuphase DC-80/81 CD player

hinged panel identical to that of the DP-80 conceals both the variable output control and the switch to choose between the three digital data inputs provided on the rear panel, two optical and one coaxial. Inside, despite the large box, there is no wasted space. Separate toroidal transformers for the digital and analog sections are on the left, with two sets of circuit boards on the right, plugging into a vertical motherboard via high-quality edge connectors. Electrically, apart from the transformers and digital filter/oversampling circuits, the construction is totally dual mono, with separate regulated power supplies for the digital circuitry, left and right DACs, and the left and right analog circuits. That care has been taken to minimize what most other manufacturers have yet to recognize as problems is shown by the fact that the lower board is contained within an electrostatic shield to reduce stray RF radiation. This board handles the reception of the digital data, followed first by 2x oversampling, then low-pass filtration with a 121-coefficient digital filter. The output from this board consists of two sets of parallel, 88.1kHz-sampling-rate, 16-bit digital data, fed to the DACs via optoisolators to prevent noisy digital ground spuriae from affecting the analog signal.

Above the digital board lie the two analog boards. As far as I know, these are unique for a consumer product in having discrete D/A sections, one for each channel. Rather than the ubiquitous integrated-circuit DACs, Accuphase chose to construct a discrete current-multiplying converter from very-high-precision resistors and precision current switches. "Very high" precision? The selection of these resistors is not a trivial task, if you consider that in a current-multiplying DAC, the LSB switches in a current one 32,728th the size of that switched by the MSB, *ie*, a tolerance of ±0.0016%! The DAC output is taken to a sample-and-hold chip, followed by a 9-pole, GIC-type (General Impedance Converter) Butterworth low-pass filter, said not to have an amplifier in series with the signal. The final analog amplifiers are discrete, and both conventional unbalanced outputs, via nickel-silver phono sockets, and balanced outputs, via XLR sockets, are provided. The unbalanced outputs are paralleled by variable outputs. Construction is to a superb standard, and high-quality parts, such as copper-foil, polystyrene-dielectric capacitors for the low-pass analog filters, are used throughout.

Sound quality: As explained earlier, the Accuphase player was auditioned via its variable outputs, the sound straight into the power amplifier then being better than when the direct outputs were taken to the preamplifier. The best sound was obtained with the direct outputs fed straight to the power amplifier; unfortunately, the high output then drove the Krell into clipping rather too often. (The sound was then also *very* loud!) The two units were connected via optical fiber.

Well. What can I say? This player can play music. Its presentation of CD sound is the most accessible I have ever experienced. I started with two torture discs: Prefab Sprout's *Two Wheels Good* (Epic EK 40100) and *Trio*, from those good ol' gals Dolly, Linda, and Emmylou. Both albums are distinguished by heavy hands on the mixing desk's high-frequency equalization controls, yet, without

any apparent HF rolloff—if anything, the fundamental Accuphase sound is itself a little bright—the discs were rendered listenable. This didn't always happen: the London *Scheherazade*, from Dutoit and the Montreal SO (410 253-2), one of the few modern recordings of this work to approach the subtlety of soundstaging captured on the old HMV Beecham, remained hard in balance. Similarly, the Sibelius Second, which sounded so overwhelming on the Stax, was revealed on the Accuphase as a very early digital recording, with a cloud of what sounded like quantization distortion floating above the string players' heads.

But those London recordings aside—as noted by Harry Pearson in his *TAS* review (Issue 43), London/Deccas did not sound at their best with this machine—the Accuphase had that magic ability to make you want to just keep putting discs on, one musical experience leading naturally to another, a train of association unbroken by intrusive hi-fi considerations. To give an example, one evening's listening went as follows: the Previn Britten *Four Sea Interludes* (EMI CDC 7476672); my friend Mike Skeet's Soundfield mike recording of Britten's *Missa Brevis* for boys' voices and organ (Proudsound CD11402); Part the First of the live Harnoncourt *Messiah* (Teldec 8.35617ZB); the Barbirolli Vaughan Williams's *Fantasia on a Theme of Thomas Tallis* (EMI CDC 7475372); Charles Ives's *Unanswered Question* on Sheffield Lab (CD-27); finishing up with the 1966 Elisabeth Schwarzkopf recording of Richard Strauss's *Four Last Songs* (EMI CDC 7472762). This latter I had felt to be a disappointing transfer to silver disc when compared with the black, with Schwarzkopf unnaturally forward. Played on the Accuphase, however, from the first breath of *Frublung* to the transcendental calmness of *Im Abendrot*—"O spacious, tranquil peace, so profound in the gloaming"—your emotions are ripped one way, then the other, at the end leaving you limp in your chair: "How tired we are of traveling—is this, perchance, Death?" With the Accuphase, no way was it possible to follow that piece with anything but silence—which is how Strauss planned it.

Common threads running through the albums listed, as heard on the Accuphase, included a wonderfully spacious presentation of the soundstage, on a par with the CAL Tempest, and a tonal balance on the bright side of neutral, with a less well-defined bass than I would expect, particularly when compared with the Stax Quattro. The clear, unforced delineation of detail, instead of being artificially thrust forward, is integrated within the musical whole, producing goosebumps. In the Britten *Missa Brevis*, for example, the chiff of the wind on the flute stops is unambiguously part of the organ sound, not separate from it. Bass-guitar fret noise, as the player lifts off with wirewound strings, again is part of the overall sound and doesn't float in front of the body of the tone. If I say that this aspect of the Accuphase reminds me of the musical qualities of the Koetsu Signature Red or Linn Troika cartridges, then you will sense that I liked this aspect of the player's performance a lot, even if, at the end of the day, I have to own up that analog goes even farther in this direction.

Measurements: Output voltage was very slightly below spec at 2.46V, but is still a very audible 1.8dB higher than the standard 2V, so should be compensated for when making direct comparisons with other CD players. The response was flat through the bass, midrange, and lower treble, being 0.25dB down at 8Hz and 10kHz, and an innocuous -0.5dB at 20kHz. De-emphasis was spot-on, being generally ±0.05dB but reaching a maximum error of -0.25dB at 16kHz. Track access was astonishingly fast, up with the top Sonys at just 2s to reach track 15 on the CBS disc. Effectively, it was instantaneous; no sooner had you punched up a desired track on the remote control than the music started. Error correction was not quite up to the standard set by the Stax, however: though the fake fingerprint caused no problems, there was noticeable muting with the 800um and 900um interrupted-data and 800um "black-spot" tracks on the Philips test disc.

The low-level sound was noise-free compared with the Stax, so I had more confidence in the +10dB level error measured at -90dB. The undithered -90dB waveshape was square, as it should be, being encoded with just three levels, though it had a slightly asymmetric duty cycle. The 500Hz fade test proved interesting: While free from heterodyning whistles, there was a noticeable haze

of high-order harmonics audible, even at the -60dB level. As the tone faded, the third harmonic became predominant, before the signal slid under the noise. While not up to the standard for low-level linearity set by the Stax, this was still very good compared with some of the other CD players with which I have tried this test.

Conclusion: The Accuphase DP-80/DC-81 may not quite offer the extraordinary resolution of detail that I found so impressive with the Stax, and has less well-defined low frequencies, but it has the most "accessible" sound of any CD player I have ever listened to. If the California Audio Labs and Stax players are superb hi-fi components in CD-player terms, then the Accuphase is superb in true high-fidelity terms. The threshold that I mentioned at the beginning of this review, if not quite nonexistent with the Accuphase DP-80/DC-81, was low enough to provide no impediment to the meaning of the music. But at what price! It being impossible to put a value on a single component costing more than 80% of *Stereophile* readers' systems, I shall hid behind the words of 19th-century English writer John Ruskin: "All works must bear a price in proportion to the skill, time, expense and risk attending their invention or manufacture. Those things called dear are, when justly estimated, the cheapest." **S**

NAD MODEL 6300 CASSETTE RECORDER

George Graves II

NAD cassette deck

Specifications: three-motor, three-head cassette recorder, incorporating Dolby HX-Pro headroom expansion and Dyneq. Price: $898. Approximate number of dealers: 300. Manufacturer: NAD (USA), PO Box 98, Norwood, MA 02062. Tel: (617) 762-0202.

The Compact Cassette has come a long way since Philips introduced it as a portable dictation medium in 1963. It seems that hardly a month goes by without some audio manufacturer introducing a new, up-market recorder which promises to "redefine the state of the art." NAD, long known for their "purist" approach to audio component design, is no exception. With the introduction of the Model 6300 cassette deck, NAD has joined the high-end cassette market, and, as with most NAD products, has done so in a fairly unconventional manner.

The appearance of this unit belies its technical sophistication. First of all, one is struck by the utter simplicity of the front panel. It sports a large black knob (the record level/balance control) on the upper right-hand section of the fascia, and a cassette-well door on the left side. Sprinkled across the rest of the front panel are a few controls and switches looking much like those sported by any other modern cassette deck. A casual inspection, in fact, will reveal nothing which would grab the observer's attention, or in any way distinguish this unit from any of a dozen or more similar-

looking machines from competing Japanese suppliers—nothing, that is, except the price! This plain-Jane-looking unit costs almost $900!

Why the high price for what appears to be a fairly ordinary cassette deck? The 6300 is anything but ordinary. First off, it has a sophisticated, servo-controlled, three-motor, three-head transport with dual capstans and vanishingly low wow and flutter. Secondly, it features both Dolby HX-Pro headroom extension and the Tandberg "Dyneq" circuitry, the former adjusting the bias, the latter the equalization, according to the HF content of the record signal, both circuits squeezing the signal's high frequencies within the cassette tape's limited HF MOL. The NAD deck also sports a built-in compressor for making restricted dynamic range recordings for use in automobiles, where high ambient noise levels make wide-range recordings impractical.

The deck does have a dual-range counter which will count tape or time as you wish, but this latter feature is not that useful, in my opinion, because it counts *up* from zero rather than down, which would indicate remaining time. An unusual feature (and a welcome one) with this deck is the inclusion of a six-function infra-red remote control. This strangely shaped remote controls all transport functions (Stop, Play, Pause, Fast Forward, Fast Reverse) and Record. And that's it! There are no more "features." Although there are "Bias Trim" (which affords some adjustment over the factory-set tape selections) and "Play Trim" (about which, more later) controls, there are no "music-search" features; there isn't even a pair of microphone inputs—just line level.

What one gets for $900 is a very-good-performing, no-frills cassette deck of excellent design and construction which should give its owner many years of sterling, trouble-free service.

How does it work?

The NAD 6300 costs more than twice the price of decks with similar features that promise the same level of performance. If one ranked this machine on looks and features alone, it would end up being on a par with the Aiwa F660 (three heads, fixed bias/EQ with trim, HX-Pro, etc.). In fact, however, the 6300 provides a much higher level of performance than just about any other seemingly similar machine.

However, as mentioned earlier, this machine does not sport the now common microprocessor-controlled automatic biasing and EQ circuitry. Instead, a front-panel switch allows the user to select between three settings: Normal (low bias, 120us equalization), CrO2 (high bias, 75us equalization), and Metal (Metal tape bias, 75us equalization), corresponding to the international Type I, Type II, and Type IV formulations. (Type III is the long-obsolete dual-layer ferrichrome tape.) Alongside this selector is a fine bias adjustment which allows for some deviation from the factory settings. I find this omission of auto-biasing circuitry to be this deck's greatest shortcoming.

Basically it's a question of cassettes needing all the help they can get. Tape formulations have come a long way since the dark ages of the early '70s, but the fact remains that cassette tape creeps along at 1⅞ inches per second (ips), recording four tracks of information on a magnetic band only ⅛" wide. This puts real physical limits on the amount and quality of information that can be placed on the tape itself, in stark contrast to professional recorders, which often put only two tracks on a ½-inch-wide tape traveling at 15 or even *30* ips!

Anyone who has ever done any serious recording knows the importance of biasing and EQ, even for the more forgiving open-reel tape formats. Just a few dB one way or the other can make a big difference between having a tape which sounds just like the microphone feed, and having one which sounds different, therefore wrong. The problem is that even with professional recorders, it takes knowledge and test equipment to set up a machine for a particular tape. The same is true for cassette decks, except that having the machine properly set up is even more critical than with reel-to-reel because the cassette medium is so much less forgiving. The average consumer, even the most knowledgeable, does not have access to the type of equipment required to set up a cassette deck. It is also true that most would probably not be inclined to do so even if they *had* the equipment because it is a lengthy and time-consuming procedure.

Yet, it needs to be done if the user is to wring the last measure of performance from the humble cassette. The buyer of an $800 recorder capable of fantastic performance is not going to get that performance unless the deck is biased for the tape he or she is using. Of course, one could stick with the tape brands and types that the machine was factory-set for—if one knew what they were.

NAD sent me three Maxell cassettes to use in the evaluation of the 6300, but the instruction book gives the average buyer no clue as to what tapes his machine was set up to use. Since it is a fact that the major tape manufacturers change their formulations almost as often as Audio Research used to revise their preamps, the particular type of tape that your machine was biased for may be unavailable in a year or so. The only viable and practical answer to this problem is a machine that in some way allows the user to bias his own machine without the need for extra test gear and without a complicated methodology requiring that the user pry the chassis open every time he wishes to change tape parameters. Such a system can be manual (as with some of the Nakamichis) or automatic.

The curious "Play Trim" control is described in the NAD's somewhat skimpy manual as a control for restoring the highs often lost when playing a tape made on a machine with a slightly different tape-head azimuth. This is a real concern with tape decks. The Play Trim control does brighten a tape when rotated clockwise from its indented position, but I prefer Nakamichi's system of allowing the user to actually trim the playback head alignment to the NAD system of using a high-frequency peaking control (with all of the phase problems and extra tape noise[1] that such a circuit brings with it).

I ran some "round-trip" frequency-response tests (recording a signal on tape, then measuring the signal at playback to see how the record/playback process has altered it), using two of the three Maxell tapes supplied me by NAD (Maxell XLII-S high-bias tape and Maxell Metal) to check the effectiveness of the "Dolby HX-Pro" and Tandberg "Dyneq" circuits for increasing high-frequency headroom. Decks without the headroom-extension features afforded by the above two circuits tend to self-erase frequencies above 10-12kHz at fairly low recording levels (usually -20dB on the VU meter or lower). This means that as your music gets louder it contains less and less high-frequency information, accompanied by increasing midrange distortion. (For a more detailed description of the phenomenon of high-frequency saturation in cassette decks and how Dolby HX-Pro helps overcome it, see my review of the Aiwa F-770 in Vol.9 No.7).

I recorded a 16kHz signal on tape at increasing levels until the level of the signal coming off the tape started to drop in relation to the signal going on tape (they were matched at -30dB using a Hewlett-Packard 400 audio voltmeter). Starting with the Type II tape, I found that the 16kHz signal started to saturate at -5dB as measured on the H-P 400 (all tests are relative to 180 nanoWebers/meter of fluxivity, the standard "0 VU" reference level for cassettes). Then I decreased the frequency of the test signal until the input and output signals were again equal in level at 0 VU. With Maxell XL II-S, the frequency at which full output was restored was about 14.2kHz on the left channel, 14.5kHz on the right. Repeating this procedure with the Maxell metal tape showed that saturation at 16kHz did not occur until almost exactly 0 VU on the right channel, and at +0.5 VU on the right. These are astounding figures, and all the more remarkable when you realize that professional reel-to-reel machines of a couple of decades ago could accomplish this feat only at 15ips or faster!

How does it sound?

From the above frequency-response and high-frequency headroom figures, one would expect this deck to make marvelous recordings. Well, one would not be disappointed! Copying both symphonic master tapes and direct-to-disc recordings was a joy with the 6300. Using Dolby-C noise reduction, it became damn difficult to tell the source from the copy. When there *was* a difference, it manifested itself as a slight loss of detail, a smoothing or glossing-over of the sound. At no time was the slightly bright and hard sound which characterizes my Aiwa AD-F770 evident with this machine. When compared with my master tapes, the NAD did lose

1 Although, as the NAD's HF control comes *before* the Dolby decoding, noise boost should be minimal. —JA

something in soundstage presentation—it shrank somewhat in all dimensions. I might add that even though this shrinkage is very evident on direct comparison with the source, it is nonetheless *much* less severe than the soundstage shrinkage I heard when these same masters were transferred to disc in some test pressings I had made. Good material still images very well on the NAD.

To check the importance of the aforementioned lack of auto-biasing facilities on this deck, I made some tests on other brands of tape, such as TDK Type II cassettes and some Sony metal cassettes. Using my ears (as presumably the average user would be required to do), I tried to trim the bias control for best results on these other tapes. I then measured the results and they were unimpressive. I could get nowhere near the headroom figures that I obtained with the NAD-supplied tapes. The copies of master tapes that I made with these "hand-tuned" Sony and TDK tapes were either brighter (Sony Metal) or duller (TDK Type II) than the source. The sound with these tapes was not as clean as the Maxells, either. Lest you think that perhaps the Maxell tapes were just better tapes than the others tried, I was able to get equal results with all tapes (of the same type) on my reference Aiwa AD-F770, which *does* have auto biasing.[2]

Conclusion

I found the NAD 6300 cassette deck to be easy to use, well built, and capable of flawless performance. It has among the most stable transport of any that I have ever heard, and with one exception, the lack of features usually sported by decks in this range is a minor annoyance. However, I feel that the lack of a real user-accessible tape optimization feature is a serious omission in a deck of this price and performance potential. If NAD were to replace the "Car" compressor feature and the high-frequency trim control with a good, microprocessor-controlled tape-optimization circuit, they would have a first-class winner. Recommended, therefore, but with reservations: the buyer must rely on the manufacturer and/or dealer to properly line up the deck for suitable tapes. S

2 One way to adjust the bias with a three-head deck is to record pink noise or interstation FM tuner noise at -20dB and monitor the signal off-tape while adjusting the trim control. The bias will be about right when the off-tape sound has the same amount of HF as the original, determined by switching between source and tape. However, this is an approximate method at best. —JA

MARANTZ ST551 TUNER

Don Scott

Marantz ST551 FM tuner

FM stereo/AM tuner with digitally synthesized tuning. Usable sensitivity: 1.9uV/10.8dBf mono, 5.5uV/20dBf stereo. 50dB stereo quieting sensitivity: 35uV/36dBf. Capture ratio: 0.9dB. Selectivity: 60dB alternate channel, 4dB adjacent channel. S/N ratio at 65dBf: 82dB mono, 80dB stereo. Stereo THD: 0.1%. Separation: 53dB. SCA rejection: 40dB. AM suppression ratio: 60dB. 19 and 38kHz products: -40dB. Power consumption: 12W. Dimensions: 16.5" W by 10.25" D by 2.375" H. Weight: 5 lbs. Price: $299.95. Approximate number of dealers: 500-700. Serial Number: 66U010199. Manufacturer: Marantz Co., Inc., 20525 Nordhoff Street, Chatsworth, CA 91311. Tel: (818) 998-9333. Toll-free: (800) 423-5101.

The Marantz ST-74 tuner, reviewed in Vol.8 No.7, was described as having "butter-and-marmalade appearance and AM performance, but dry-toast FM." The latest offering from Marantz, the ST551,[1] has not quite as sweet AM or appearance, but its more palatable FM makes it one of the best-sounding tuners encountered. It doesn't lack features either—remote control of manual tuning, scan, band selection, and presets—all at a fairly low price.

A functional layout is used on the front panel. From left to right are the power switch, band selectors, combined mute-mono/stereo switch, four-digit off-white-against-aqua frequency display, infrared remote-control sensor, controls for eight AM and 16 FM presets, and a large, easy-to-use tuning rocker switch. Overall, the tuner looks businesslike and is highlighted with dull gold lettering. The rear panel contains terminals for AM and 75- or 300-ohm FM antennas, RCA audio jacks, a switched 500W accessory outlet, and provision for marriage to a complete remote-control Marantz system.

Circuit Design

As is often the case with audio components, simple circuits yield the best results. The ST551 was designed to work reasonably well with a minimum parts count.

The tuner contains an unusually well-filtered and regulated power supply to support the remote-control circuitry. Also included are an active low-pass filter from the PLL synthesizer to the FM local oscillator, and an effective buffer amplifier to isolate the local oscillator from the PLL controller. The above three factors contribute to the excellent low-noise audio heard from this tuner.

Specifications

Above-average specifications tell the story of why this tuner has the sonic finesse of a fine tuner. Capture ratio is 0.9dB, meaning the tuner can effectively mask a weaker station on the same frequency. S/N ratio in stereo is claimed as 70dB. It was actually an outstanding 83dB, about as quiet as the McIntosh MR 80 and the Luxman T-02. Stereo separation is stated as 45dB. It proved to be 53dB—as good as I have ever measured. Subcarrier and SCA rejection were barely adequate at 40-45dB. However, little high-frequency grundge or SCA birdies were heard. Sensitivity and selectivity ratings are typical, and were not exceeded.

FM Audio Quality

The ST551 was compared with an unusual number of tuners (those in italics were courtesy of the Connecticut Audio Society): Onkyo T-9090, Kenwood KT-8300, Sansui TU-9900, Magnum Dynalab FT-101, Pioneer F-77, *Sequerra Model 1*, *Yamaha T-85* and *T-70*, and the tuner section of a *McIntosh MAC4200* receiver. Also used as a secondary reference were tapes made from Onkyo T-4087, Bogen TP-100, and Luxman T-02 tuners. In no case was the ST551 outclassed in projecting a full-bodied soundstage with accompanying low intermodulation distortion. With the exception of the Mac, all tuners in the first group sounded restricted and choked in the critical midrange voice area when directly compared with the ST551. However, the MAC4200 had an extra 2dB under 500Hz when listened to via the record outputs. (I hope to test this same tuner section in the separate MR7082 to make a more proper judgment.)

Although not available for direct-air comparison, I'm sure the Quad FM4, NEC T-6E, Bogen TP-100, Mission Cyrus, and the Arcam Alpha would have strutted equal sonic virtues. What I *can* conclude is that, when fed a clean moderate or strong signal, the ST551 produces audio equal to the best. A word of caution here: if a tuner won't receive a station at all, its sonic worth is limited. For instance, the Yamaha T-85, Onkyo T-9090, and McIntosh MR 80 will receive stations clearly in stereo that can be heard, at best, as hopeless splatter on a wide-band tuner such as the ST551. In actual use, therefore, it is very rare for a less selective tuner to produce the best audio, due to the crowded RF environment in the US. And I'll be damned if it isn't always the hardest stations to get that have the best programming. (This must be an FCC regulation closely adhered to.)

AM Section

As received, the FM alignment on this tuner was on the money, but the AM was not. After alignment, the loop sensitivity was a typical

1 The actual model reviewed was the slightly cosmetically different but otherwise identical ST560, which is not available in the USA.

300uV/m. But, believe it or not, the high-frequency response extended to nearly 7kHz, with unusually low distortion. Music actually sounded like music on the ST551.

Conclusion

One of the curious things noticed was how good this tuner sounds with my 24-year-old Marantz Model 5 amps. Slowly this gap of excellence is being filled. For instance, the 1986 Marantz PM-74 amplifier, though not widely marketed in the US, and the CD-74 CD player, are serious audio components. (The amp was found by Sam Tellig to be a winner when using the line inputs.) And the 1987 June CES saw the introduction of products that will ensure the pioneer Marantz label will not go entirely the route of the rack-system bandwagon.

The ST551 will not go down in history as one of the greats, as did the Marantz 10B, but it receives my recommendation in order to appreciate what can be heard from properly engineered stations. With a good antenna and no strong signals on adjacent channels, the ST551 will perform well. **S**

NIKKO NT-950 AM/FM TUNER

Don Scott

Nikko NT-950 FM/AM tuner

FM stereo/AM tuner with digitally synthesized tuning. Usable sensitivity: 1.9uV/10.8dBf mono, 6uV/20dBf stereo. 50dB stereo quieting sensitivity: 32uV/35dBf with noise reduction, 35uV/36dBf without noise reduction. Capture ratio: 1.5dB. Selectivity: 80dB alternate channel, 12dB adjacent channel. S/N ratio at 65dBf: 82dB mono, 78dB stereo. Stereo THD: 0.08%. Stereo separation: 50dB. SCA rejection: 65dB. AM suppression ratio: 60dB. 19 and 38kHz products: -67dB. Power consumption: 12W. Dimensions: 17.25" W by 10.4" D by 1.75" H. Weight: 7 lbs. Price: $259.95. Approximate number of dealers: 200. Manufacturer: Nikko, 5830 South Triangle Drive, Commerce, CA 90040. Tel: (213) 721-1168.

Japanese company Nikko was manufacturing electronic products for the home market as early as 1947, and became the first high-end (no junk) firm to compete with early American hi-fi in Japan. Nikko has had a presence in the US for 25 years, and has shown some spurts of serious audio activity. Within the past year, however, the company was purchased by Asti Pacific Corporation, a large Japanese manufacturer specializing in satellite technology; access to Asti's technical resources, if used correctly, should evoke some innovative audio and video products.

Two of Nikko's latest products that caught my attention were the multi-function AVR-65 AM/FM/VHF/UHF/CATV, remote control, 65W/channel receiver and the NT-950 tuner. The tuner is of particular interest because it has both high-blend and dual IF bandwidth for a usually discounted $259. The nearest-priced competition also featuring these features is the Onkyo T-4087 ($380). The Sansui TU-D33X and the H/K TU905 (both $229) have only high-blend, and the Denon TU-600 ($350) has only IF bandwidth selection. Consequently, the Nikko NT-950 scores points as having the most features at its yen-point—if all works well. As it turns out, all does work well; this is a good tuner even for the semi-serious FM buff. Other features include a 5-level signal-

strength meter, 10 AM and 10 FM presets, and scan of preset stations.

The substantial, very slim, black metal cabinet has an attractive front panel with white and red lettering. From left to right are power on-off, red stereo indicator, green signal meter, 5-segment blue frequency display, memory and scan controls, the infamous combined FM muting and stereo switch (also controls AM muting),[1] IF bandwidth selector, separate AM and FM switches, and a rather narrow—just OK—tuning rocker. Tuning is in 100kHz steps on FM, and the short route from 88.1 to 107.9 is provided. Overall feel is good, interior construction is medium grade, and the power transformer is larger than found in typical low-cost tuners.

Specifications

Nothing to be ashamed of here: noise figures of 78dB in stereo, 80dB alternate and 12dB adjacent-channel selectivity, 65dB AM suppression, 60dB SCA rejection, and adequate sensitivity add up to quiet reception for all but 30% of the situations requiring more RF moxie. Tuners such as the aforementioned Onkyo T-4087, the Yamaha T-85 ($449), and the Luxman T-02 ($500) are upgrades in RF performance and have lower distortion in the narrow mode. THD of the NT-950 is 0.08% in stereo wideband and 1.5% in narrow. Most notable is the tuner's good AM limiting, which minimizes the harsh increase in noise with weak and fading signals.

AM Section

Buzz-saw AM. While no self-inflicted digital noise from the frequency synthesizer or microprocessor is present on FM, it obliterates all but the strongest station on AM. Nikko has been made aware of the problem and promises a solution. Of course, if you don't care about AM, this may be of little concern.

1 Like many tuners tested, the muting level is set too high to take advantage of the tuner's good fringe stereo quieting. The NT-950's control is labeled mute and is near the rear-center of the circuit board. Clockwise rotation decreases the muting point on AM and FM.

FM Sound Quality

The NT-950 is not the ultimate-sounding tuner, but neither is it the poorest. The Bogen TP-100, Quad FM4, and Denon TU-767 edge it out slightly in deep-bass extension; low frequencies, however, are taut, dynamic, and well controlled. The Marantz ST-551 exhibits 3-5dB more stereo separation, which would never be noticed in typical listening. Mono sum audio appears dead-centered in the soundstage, and the 1kHz pre-emphasis/de-emphasis hingepoint is accurate, contributing to midrange clarity. Highs are extended, with just the right amount of crispness for clarity without "ess" ing. Good filtering of 19, 38, and 67kHz components allows quiet, birdie-free listening and/or taping from this tuner. Low noise levels are aided by a unique high-blend. Its action is very mild, just taking the fuzz off many stations and decreasing stereo separation at high frequencies by only 5dB. What would be nice would be a two-step high-blend similar to that featured by the McIntosh MR-78, which would allow use over a wider range of signal levels. In short, the NT-950 has few FM faults and sounds particularly articulate with voice.

Conclusion

The narrow-bandwidth position sets the NT-950 apart from its similarly priced competition: high-value products like this will shorten *Stereophile*'s "Recommended Components" list. I hope, however, that Nikko can get the AM working properly. It is recommended for all locations except where extreme selectivity is needed. The manufacturer has two additional tuners in development, models Gamma 60 and 80. Indications are that Nikko is a company to watch. S

MAGNUM / DYNALAB 205 FM SIGNAL AMPLIFIER

Don Scott

Slim frequency-selective outboard FM booster. Gain: ±30dB. Noise floor: 3-4dB. Power consumption: 15W. Dimensions: 19" W by 11½" D by 15/16" H with feet. Price: $229. Approximate

Magnum aerial booster

number of dealers: 100. Manufacturer: Magnum/Dynalab, Brampton, Ontario, Canada. US: 1971 Abbott Rd., Lackawanna, NY 14218. Tel: (800) 448-8490 outside NY state.

Magnum/Dynalab is one of the few companies that takes good FM reception seriously. Their models 95 and 105 outboard FM boosters (still available) were reviewed in Vol.7 No.7, the 95 winning the highest marks.

One of the problems with these, or any, outboard devices is where to put them. With this in mind, Magnum has come up with a solution both cosmetically and aesthetically pleasing. The new model 205 is the width and depth of a typical tuner, but half its height, allowing it to serve as a platform for a tuner, giving a more professional look than the usual auxiliary boxes and dangling cables. The design cosmetics dictate an oversized cabinet for the 3-RF-stage booster circuit board and power supply. However, the extra footroom does no harm, allowing distant dress of input and output leads; these can cause RF-feedback, resulting in oscillation and noise if placed in close proximity to each other.

From left to right, the sleek, white-lettered-over-black front panel contains power on-off, bypass-amplify switch, variable frequency select, and gain control. Looks and feel are professional. In fact, the overall design would make an attractive tuner. The unit must be powered even for bypass, which seems a waste of a few watts. Antenna connectors are F-type, the 205 is rack-mountable, and it carries a five-year warranty.

Specifications

The manufacturer claims -30 to +30dB control of RF signal level. The noise floor of the 205 is no more than 3-4dB, leaving an effective gain of 26-27dB. A method of cancelling the already low noise floor is to use two 75-300 ohm transformers back-to-back at the output, which will have about 4dB loss, leaving pure signal. Bandwidth is about 400-500kHz, slightly improved over the 600kHz of previous units.

Performance

26dB of available gain is not a panacea for weak signals unless the input signal is low-noise—it can amount to 26dB of boosted garbage instead of the desired signal. Of course, the best place for a signal booster is at the antenna, but no one makes a tunable mast-mounted device; and it would be most inconvenient to run up on the roof to change the booster tuning for each change in station frequency. The next best thing would be to use coaxial cable from the antenna to the booster, keeping the run as short as possible to minimize signal loss and noise induction.

Incidentally, the best coaxial I've used is RG-6, which has foil shielding instead of the braided surround of common RG-59U, and 1dB less loss per 100 feet. I've also noticed large differences in the transfer efficiency of 300-75 ohm transformers (most consist of 3 capacitors without any coil windings). The Radio Shack outdoor 15-1140 ($2.79) is okay, but I have been able to get about 10% more signal with some indoor matching transformers. No particular brand has been consistently good; therefore, some experimentation is in order. Weatherproofing is needed for indoor types if used outdoors.

The added selectivity of 400-500kHz of the 205 does much to dampen or eliminate in-band or out-of-band interference and aids alternate-channel selectivity. Unfortunately, adjacent-channel selectivity may be hindered because the booster passes twice the FR-window of the US 200kHz spacing. Sliding the booster tuning to one side to attenuate one station may help in some instances, depending on the unaided selectivity of the tuner used. The 205 can also be used as a variable signal attenuator to help stanchion a powerful local that may be causing RF-overload and resulting distortion.

Other factors determining the successful use of the 205 are the RF-gain and RF-signal/noise. Only slightly noisy stereo and dead-quiet mono of stations previously heard in noisy mono resulted with the Onkyo T-4087 and

the AR T-04, but negative results were noted with higher-gain tuners such as the Onkyo T-9090 and Yamaha T-85.

Conclusion

I can give Magnum/Dynalab a clean bill of health on the 205 except for two items (one brought to my attention by a *Stereophile* reader). First, where are the fuses? Their omission seems an undesirable cost *vs* safety measure. The other problem, in my opinion, is excessive price. Some *tuners* cost less. On the other hand, the manufacturer is the only kid on the block in the tunable FM-booster business, and their series of boosters may be the only means of obtaining less offensive reception of desired station(s). I estimate that the models 95, 105, and 205 will give positive results in only 30% of reception situations because of the S/N contributions cited in the above companion equipment. The 205, however, seems to be the best of this Canadian manufacturer's crop, and it looks exquisite.

CRAMOLIN, TWEEK, & SIGNET CONTACT ENHANCERS / CLEANERS

George M. Graves II offers a few words on hi-fi housekeeping

Sumiko Tweek

I admit it: I hate RCA connectors! I hate those horribly unreliable and irrational "pin-plugs" (and jacks) that the audio industry has been lumbered with since WWII. They are, as I am sure all of you know, unreliable because they get loose, the strain reliefs break, the contacts are not gas-tight, allowing air to oxidize the conductors, etc. They are irrational because in an audio system, connecting the "hot" side of a cable to a component before making the ground connection almost always results in a loud, speaker-destroying hum (or worse, a humongous ultrasonic amplifier-frying oscillation). With the exception of expensive, carefully thought out designs from Tiffany and WBT, the RCA plug, with its protruding "hot" pin, does just that stupid thing! Unfortunately, we are stuck with them—apart from the Europeans who wisely came up with the DIN plug (also not without its problems, but that's another story)—so we might as well learn how to get the most from these cantankerous troublemakers.

Even the best, most expensive connectors (aerospace stuff) share with the worst (RCAs) two characteristic faults: 1) the mating surfaces get dirty (and the contact resistance goes up); and 2) when examined under a microscope, it becomes apparent that, because of surface irregularities in even the most highly-polished metal, very little actual metal-to-metal contact between the mating surfaces is taking place (compared to overall mating surface area). Both are no-nos in audio where any "diode effect" on connectors introduces audible distortion.[1] With very high-resolution systems, this situation becomes even more problematical, due to the fact that the diode effect can effectively cancel out the advantages of the high-cost, first-rate interconnects which are likely to be used there.

1 Diode effect is a phenomenon which occurs when connections are less than optimum. Dirt, or an oxide film on one or both of the conductors, causes the mating interface to have a resistance to current flow which is higher in one direction of flow than it is in the other. This is the main characteristic of a diode.

Clearly, some form of maintenance is required here.

There are many products on the market which address these problems, and most are successful to a greater or lesser extent. The question here is: How to use them to their best advantage? I shall discuss three such products, then a methodology for using them.

Tweek

I'm sure that there aren't three audiophiles in the country who haven't at least heard of this product from Dayton-Wright, which is marketed by Sumiko.[2] What it is is a contact *enhancer*. Tweek is applied to both parts of the connector and, when the mating surfaces are *clean*, this fluid—paradoxically, an insulator—flows into all of the irregularities between the two metal surfaces. Essentially, by keeping all contaminants and air away from the freshly made metal-metal contacts, it makes the connector disappear! Over time, the fluid appears to become slightly conductive, thus enhancing the contact further. Tweek's contribution can easily be heard. I have never seen an instance where it didn't audibly (and with a video system, visibly) improve things.

Cramolin

As great a product as Tweek is, it can do little for dirty or corroded contacts. In order to enhance a connection, there must be maximum clean-metal contact area to begin with. Using an abrasive such as garnet paper may make a plug shiny again, but it also roughs it up, thus limiting the contact area between two mating surfaces. A cleaner is needed here, and there is none better than Cramolin R-5. This stuff is manufactured by Caig Laboratories in Escondido, CA, and is used widely in the computer field and in aerospace and military electronics. It is a cleaner, a lubricator, and an antioxidant. It will remove all types of dirt and corroded crud from any metal surface. Cramolin R-5 might be difficult to find; most hi-fi shops have never even heard of it. I found mine in a store which wholesales spare parts to the TV-repair industry. Cramolin R-5 comes in a 6 oz. spray can with a long plastic nozzle and retails for less than ten bucks. A complete kit, containing two different Cramolin fluids (one for old contacts and one for new), lint-free applicators, and instructions, is available for $20 plus shipping from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458.

Signet Contact Cleaner and Restorer (SK302)

Signet[3] sells a kit, called the SK302, for doing both of the above. The phial of liquid supplied contains both a cleaner and a contact enhancer in one solution. In my opinion, formed after exhaustive tests of all of these products, it neither cleans as well as Cramolin, nor enhances as well as Tweek. What it does have going for it is the cleverest set of cleaning "tools" I have ever seen, literally worth the price of admission. When you buy this kit, you get a phial of the aforementioned liquid, a plastic wrapper with "Q-tips," and a small plastic zip-locked bag containing the cleaning tools. All of these tools are made of a porous, white, slightly rough-to-the-feel plastic, and there are two of each. Four are just round dowel-like pieces of different diameters, for cleaning banana jacks and the like. The other two look like RCA plugs! That's right, here are two plastic RCA plugs, complete with a long center pin. The back of the tool is a female RCA connector.

To use this device you put the fluid on the tool and insert it into a female RCA jack. Rotating the tool rubs the slightly rough surfaces of the tool against all of the mating surface of the jack: the barrel, the pin, the works. To clean a male RCA, you coat the other end of the tool with the cleaning liquid and insert the plug. Again, turning the tool cleans both tip and barrel. I wish I could tell you that Signet offers these tools separately, but alas I can't. To get them, you'll have to buy the SK302 cleaning kit for about $15. Keep the tools, throw out the liquid (or keep it for cleaning the TV tuner).

How to use this stuff

If you're starting from scratch here, set aside a Saturday. You will have to do *all* of your connections.

First, start with the preamp. Set the unit,

2 Sumiko, PO Box 5046, Berkeley, CA 94705. Tel: (415) 843-4500

3 Signet, 4701 Hudson Drive, Stow, OH 44224. Tel: (216) 688-9400

complete with all of its cables (phono tuner, amplifier, tape deck, CD, etc.) on a table in front of you. Starting at one end, remove a pair (right and left). Taking the RCA tool from the Signet kit, spray a *tiny* amount of Cramolin on the tip and inside barrel of the male end of the tool. (If the terms male and female as applied to connectors are unfamiliar or confusing, put your mind in the gutter and you'll figure it out!). Now, rotate the tool back and forth several times and withdraw it. Notice the dirt (the mating surfaces of the pristine white tool are now tinged with grey). Repeat this with the other jack and then clean each male plug using the female end of the tool in the same manner. Remember, use very little Cramolin. Too much will gum up the works.

After cleaning the male and female connectors of this first pair, you then use the brush in the Tweek bottle to coat the male and female mating surfaces. Again, use the liquid sparingly. Not only is it expensive, but too much is actually detrimental. Next, re-mate the connectors as they were originally, being careful not to swap right for left accidentally. Rotate each male plug in its socket several times to work the two coatings of Tweek together. If the plug is loose, disconnect it and bend the barrel ears in slightly with a pair of needle-nosed pliers and remate. Then go on to the next pair of interconnects, doing each the same way as the first. Finish with the speaker connectors. You'll have to improvise here, because there are so many ways to connect speakers.

After completing your entire system in this manner, remember a rule of thumb: Re-Tweek a connection every time you break it, and go through the entire clean/enhance routine at least once a year (every six months is better still).

Now sit back and take a listen. I guarantee that your system will sound better than you have ever heard it before. **S**

FOLLOW-UP

Kindel P100 II loudspeaker

Since my review of the Kindel P100 Mk.II loudspeaker appeared in Vol.10 No.4, the manufacturer has made two significant modifications to the design: the dust cap on the cone tweeter has been changed, and the woofer now has greater excursion capability.

I listened to a pair of the revised loudspeakers (unfortunately, the original review pair were no longer to hand): the new tweeter did not lead to any clearly obvious difference, but the woofer change has resulted in a significant improvement. I was no longer able to bottom the woofers at any reasonable level. The low-frequency performance is now quite respectable for a speaker of this size and price ($325/pair West Coast, $345 East); it may not be competition for state-of-the-art designs, but it will be satisfying in the right environment.

In other respects, I found no reason to alter my original review conclusions. The overall balance of the revised P100s seemed even more laid-back than the original, and the speakers still sounded a little "small" in a large listening space. I did try the P100s briefly in a room about one third the size of my main room, and found them to be better suited to this smaller space. The bass was a bit firmer, the soundstage "larger," and the balance less recessed (though still far from forward-sounding).

Since the Kindel P100 Mk.II is more likely to be used in a small space, I feel it definitely worth an audition if your listening room is less than large. **—TJN**

Snell C/i & A III

Since my original review of the Snell C/i loudspeaker (Vol.10 No.2), I have had the opportunity to investigate its performance in the bi-amp mode and to compare it with the Snell A III in my own listening room. The results in both cases are, I feel, useful additions to the previous report.

Bi-Amping: Except for its application with a subwoofer, bi-amping hasn't been very popular with audiophiles. Theoretically it should provide a cleaner bass response (the elimina-

tion of the passive crossover means there is now no choke in series with the woofer), and a cleaner upper range (elimination of power-robbing bass from the high-frequency amplifier). But the benefits must be traded off against increased expense, system complication, and the addition of an electronic crossover network—a component that often has its own electronic signature. Snell manufactures two versions of their own electronic crossover—one for the Type A III and another for the C/i.

For my bi-amp setup I chose to drive the high end of the C/i's with the Adcom GFA-555 and the low end with the Yamaha B2X. If two Adcoms had been available I would have used them: in using dissimilar amplifiers, care must be taken over consistent signal polarity and that your observations don't merely reflect the characteristics of a different amplifier used for one of the frequency ranges. To minimize this possibility, I used the Adcom for the full-range auditions in comparing the bi-amp and the mono-amp modes.[1] The mono-amp auditions were also done in a bi-wire mode to minimize the variables.

There is one significant limitation to bi-amping the C/i's: there is no way to bypass the passive crossover choke in series with the woofer. That may account for the fact that I did not find the bass in the bi-amped mode to be significantly different from the bass in the single-amped mode—at least not in ways that couldn't be explained by the change in woofer amplifier. The bi-amped bass was a bit more robust (which you might expect), but also a bit warmer and looser (which you wouldn't). The subjective superiority of one mode over the other was very much a function of the recording; the differences in any case weren't dramatic.

The same might be said for my overall impressions. On balance, I felt the Snell C/i in the bi-amped mode was a bit more detailed, but with a small loss in sweetness and soundstage accuracy—depth was not as consistently well rendered. I wasn't at all put off by the losses—the Snell crossover is as transparent as any I have tried—but neither was I overly impressed by the gains.

I can conceive of one situation in which bi-amping of the C/i's might be worthwhile. I wasn't able to try it myself, but the tube-amp fan could use a tube amplifier to drive the high end and a solid-state amp for the low end—making optimum use of the strengths of each.

But if resources are limited, it's my opinion that you'd be better off driving the C/i's with the best single amplifier you can afford[2] instead of splitting your resources on two (stereo) amps and an electronic crossover. At the very least, if you're tempted to bi-amp, start with the mono-amp setup and get your friendly Snell dealer to loan you an electronic crossover and suitable amp. That way you can determine for yourself if bi-amping of the C/i makes sense in your particular installation.

The Type A III *vs* the C/i: Beginning with the most mundane difference between these systems, the A III is about 3dB more sensitive than the C/i. The C/i really needs a powerful amplifier to "open up": the A III is also at its best with plenty of power, but can make do with somewhat less.

I auditioned both the A III and the C/i in the bi-wire (not bi-amp) mode, using the Adcom GFA-555 amplifier. The Type A III was distinctly superior to the C/i in two areas: clarity and definition from the bass through the midrange, and dynamic range.

The A III did not, in my listening room at least, appear to be deeper in bass than the C/i, but its bass reproduction was noticeably more open and defined. Detailing was also superior to the C/i in the all-important midrange. The A III had a lively, dynamic quality, a projection through the upper midrange that the more laid-back C/i could not duplicate. I'm not entirely sure this is a total plus: the A III is forward-hall, the C/i mid-hall; on many recordings I preferred the latter's perspective.

Imaging of the two systems is comparable. But I preferred the overall soundstaging of the C/i to that of the A III; the former appeared superior in its reproduction of program depth. I also consider the high end of the C/i to be superior to that of the A III. In my environment, I wasn't able to duplicate the high-end "tizziness" to which JGH objected in his review

1 It was during the early bi-amp auditions that I came to prefer the performance of the *well warmed-up* Adcom to the B2X in driving the C/is.

2 I had a similar experience with the Apogee Scintillas. Driving them in bi-amped mode, using a Krell KSA-50 and KSA-100, while better than using just the KSA-50 alone, proved less good in terms of transparency and imaging than using a single biwired KSA-100. —JA

(Vol.9 No.3), but the C/i did have a slightly softer, sweeter high-frequency response, particularly in the lower treble.

While I felt the A III to be superior to the C/i in a number of important areas, I did not find it to be universally superior, and in many ways actually preferred the C/i. You may not feel the same way, but if you audition the A III you should also give its little brother a serious listen. **—TJN**

Beard P-35 power amplifier

Shortly after I finished my listening tests of the original sample for my review in Vol.10 No.5, Mark Siebert of Beard America called to discuss my findings. He expressed surprise at the treble quality I obtained with the Beard driving my Quad ESLs, and assured me that these results were not representative of the upper-octave performance possible with the P-35. I agreed to audition another sample. Weeks passed, with no second sample in sight. Finally, at the June 1987 CES, I was informed by Beard that very likely the HF problems I encountered with the Quads were caused by output-stage instability or oscillation brought about by the highly reactive nature of the load. To solve the problem, the P-35 had been revised, a Zobel network being added to the output stage to improve its stability into reactive loads. A sample (SN 8687) of the revised amp was shipped after the show, too late for me to revise my findings in Vol.10 No.5.

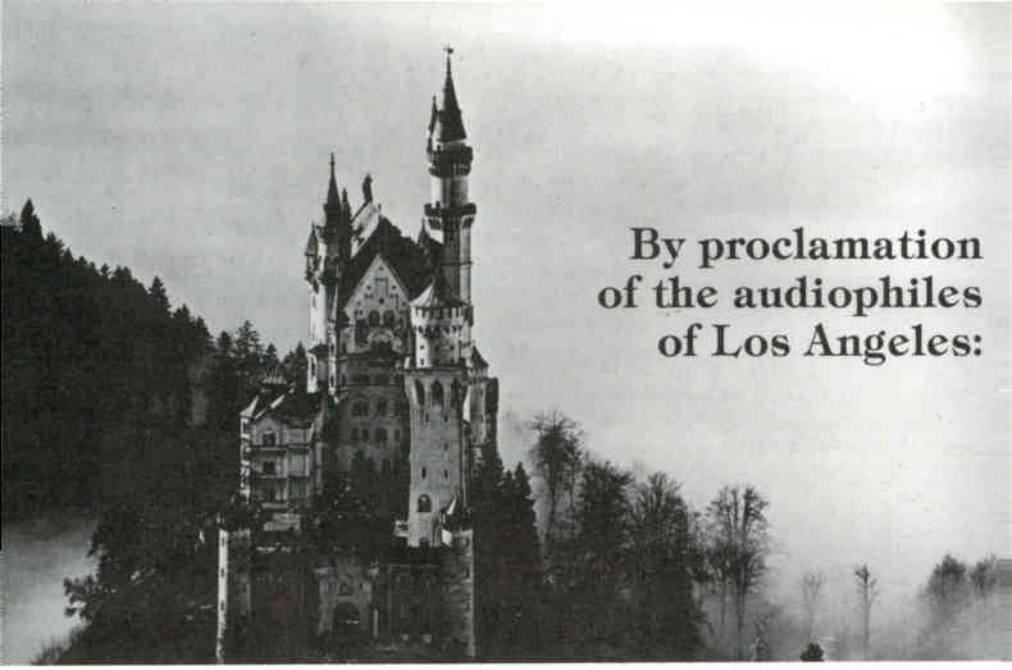
All of the listening tests this time around were performed with the Dahlia-Debra speakers. My Quads are temporarily out of service—I zapped the tweeter panel in one channel with a DC-coupled 150W amp (whose identity I will not reveal at the moment).

The excellence of the mids was still very much in evidence on both analog and digital program material. Midrange transparency and recovery of low-level detail impressed me, then and now, as deserving of a Class B appellation. Its capabilities in the area of soundstaging remained undiminished. For example, on the Opus 3 "Test Record 1" CD, the soundstage was portrayed with excellent width and depth. Instrumental spaces within the soundstage were resolved with good focus. Individual voices in the chorus on the "Innsbrook" band were nicely resolved, with the sopranos clearly soaring, as they should, above the crowd.

Yet, the Beard's performance at the frequency extremes was still troublesome; all the more irritating because of the fine standard it sets in the mids. To be sure, the highs are cleaner and smoother than with the Quads—but not clean and transparent enough for a $1900 amp. Having kept the original sample, I was able to compare the old and new. The sound of both samples—at least through the Dahlias—was essentially identical. That's not to say that with the Quads, the revised version would not have sounded better. It probably would have. But with the Dahlias, the added Zobel network apparently is not needed and did not make any difference. Getting back to the quality of the treble, it's true that treble transients were cleaner through the Dahlias, but brushed cymbals still lacked delicacy. Brass sound still did not have the requisite amount of bite, being too polite or soft, and the extreme treble was still noticeably closed-in and grainy. This sort of treble performance might sit well with a cheap and gaudy dome tweeter, but that is not the kind of tweeter the P-35 is likely to be used with. To settle the matter, I decided to enlist my spouse Lesley's help, or at least Lesley's voice via PCM-F1 master tapes. With both amplifier samples, it was obvious that the timbre of Lesley's voice was being altered through the upper octaves, there being added grain and grundge, and an opaque, closed-in quality about the upper registers. In my opinion, this is no better than Class C/D performance.

Then there's the bass. The performance of both samples was identical, and no better than what I recall of the original sample on the Quads. Somewhat underdamped and anemic pretty much sums up the Beard's reproduction of bass lines.

Although unable to verify whether the revised version performs any better into a capacitive load, I did find that with a typical speaker load, the P-35 is still an uneven performer: Class D bass, Class B mids, and a C/D rating in the treble. On an average basis, the quality of the midrange pulls the amp into Class C. In retrospect, however, does a $1900 amplifier belong in Class C, populated as it is with so many decent $1000-and-under amps? **—DO**



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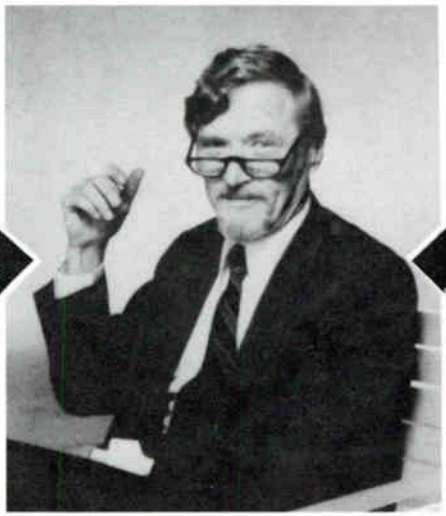


Beware little amps!

Editor:

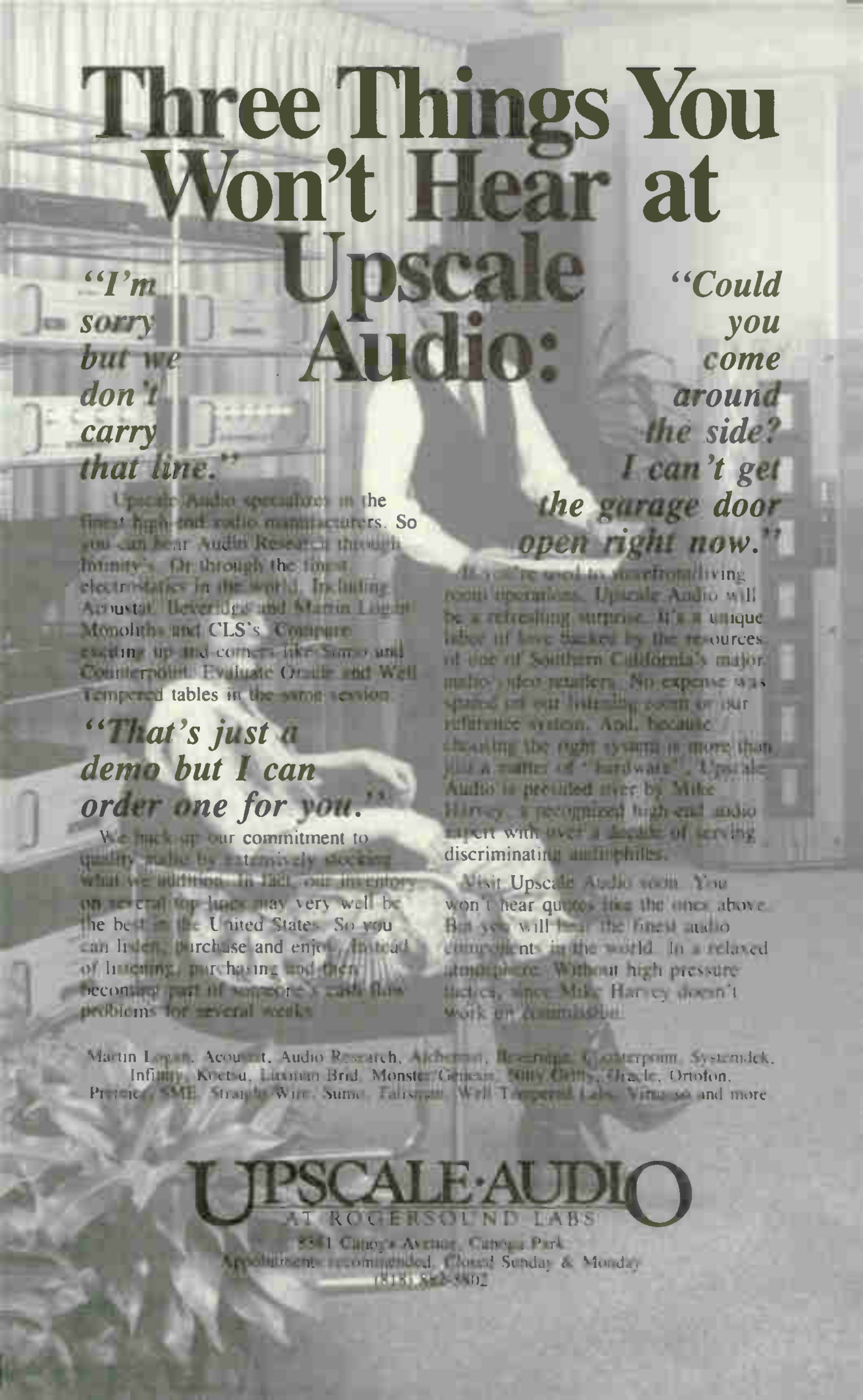
I have read in hi-fi magazines that underpowered amplifiers blow up loudspeakers. Surely this was a misprint—the larger the amplifier, the more chance it will over-stress the loudspeaker, correct?

Sidney Weisfeld
Peoria, IL

The functional part of a loudspeaker consists of a coil of copper or aluminum wire suspended in a magnetic field; pass a current through that coil and it will move. If that current has an alternating waveform (which is the case with music), whatever the coil is attached to will vibrate, producing sound. The heating effect of the current causes the coil's temperature to rise, but as loudspeakers are, in general, notoriously inefficient devices at turning electrical power into acoustic—under 1%—most of the amplifier power is wasted as surplus heat. As long as the signal level is within that intended by the drive-unit's designer, however, the coil assembly will be in a state of thermal equilibrium, waste heat being conducted away as fast as it is induced. Although the temperature will often be high—up to 200°C for the coil, perhaps, and around 50-60°C for the chassis/magnet assembly—no damage will be done.

If the drive-unit's capacity for conducting heat away from the coil is exceeded, though, the glues will melt, the coil former (often made from a heat-resistant paper) will deform or char, and ultimately the coil will melt. Long before this final stage is reached, however, the sound will have become audibly degraded as the distorted coil rubs against the magnet pole-piece. Although it is possible to damage drive-units mechanically, by causing the voice-coil to "bang against the end-stops," overheating is the primary cause of failure.

The user does have an ally in the form of electrical theory: heating power is proportional, not to the peak level of the signal, but to the RMS (Root-Mean-Square) level. This is equal to the peak level only for a squarewave;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for other waveforms it is lower. For a sinewave, for example, the RMS level is almost half the peak, 1/root 2 in fact, and for music, because of its spiky nature, it is even lower. Thus high signal levels don't automatically imply overheating of the coil. Unfortunately, however, another factor comes into play.

*Nearly all speakers use more than one drive-unit to cover the audio band. The HF units are physically much smaller than those for low frequencies, and therefore have a much lower thermal capacity and can handle very much less power before overheating. It is not uncommon for a tweeter to have a power handling of around 5-10 watts, even when used in a system capable of coping with over 100 watts. The reason the designer is able to get away with this apparent shortfall in high-frequency power handling is again due to the nature of music signals. With the exception of synthesizers, musical instruments and voices contain very little energy above 2-3kHz, so less power is required in this region for accurate reproduction. Under normal conditions, the tweeter is not usually called upon to handle high levels of power and can be relatively fragile (*ie, *inexpensive, and, more to the point, light and easily accelerated).*

If the music waveform departs from normality, however, the tweeter can be fed high-power, high-frequency signals with the expected result. The commonest way this happens is when the speaker's proud owner uses an amplifier not quite large enough for his needs, and accordingly drives it into clipping for perhaps a large percentage of the time. A sinewave fully clipped is effectively a squarewave, and as the latter has an RMS level root-2 times as high, it requires twice as much power to be reproduced. This will eat away at the tweeter's power-handling margin for high-frequency signals, but, more seriously, could destroy it with clipped low-frequency *signals.*

That may sound like a paradox, but consider a speaker crossing over to the tweeter at a not-uncommon 3kHz, being fed 100 watts of 1kHz sinewave, none of which is being handled by the tweeter. Drive the amp fully into clipping, however, and the speaker as a whole is now passing 200 watts of 1kHz squarewave, half of which, consisting of odd-order harmonics—3, 5, 7, 9kHz—at or above the crossover frequency, fries the tweeter. Small amplifiers, not big ones, destroy tweeters! **—JA**

Hi-Fi Headache

Editor:

Every time I listen to my system for more than an hour I get a headache. What am I doing wrong?

Reuben Gold
Phoenix, AZ

There are several possibilities.

You might be playing incompatible recordings. Early stereo recordings were made to be compatible with mono players, but most current stereo releases are not. Without the compatible mono, they will produce poor center fill on incompatible stereo systems, and this has been experimentally proven to produce headaches in some laboratory rats. You may be one of them.

Check your loudspeaker wiring. Speaker cable wears out with use, due to the abrasive action of all those high-current electrons flowing through it. (Digital sources are much more destructive to wires than analog sources, because of the sharp-cornered binary Ones.) Eventually, the electrons wear through the thin conductors and start to escape into the listening room, causing loss of detail, poor soundstaging, and AIDS. Wire integrity can be checked with an electron counter; at least 80% of the electrons going into the wire at the amplifier end of the cable should be coming out of the speaker end. If not, replace your wires or return them to the factory for redrawing.

Check your own wiring. Your ears may be connected in reverse polarity; if they hear a rarefaction when they expect to hear a compression, this is bound to cause a stress-related headache. Reversing your loudspeaker polarity will correct for this.

Take a Tweek and phone me in the morning. **—JGH**

Test CDs

Editor:

Could you recommend a good book on Compact Disc and a test CD both for setting up

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systems and for checking out the performance of CD players? Incidentally, I have been trying without success to contact Finial Technology to obtain details of their optical LP player. Could you let me know their address?

Robert J. Cook
Chicago, IL

Finial Technology can be contacted at 707 East Evelyn Avenue, Sunnyvale, CA 94086. Tel: (408) 720-9800.

The best book on Compact Disc and digital technology we can recommend is by Ken Pohlmann, the writer for Audio *and* Digital Audio *magazines. See the review in this issue.*

Companies marketing good test discs are Denon and RCA (widely available retail). JA produced a Test Disc for his old magazine, Hi-Fi News & Record Review*; this is still available from the HFN/RR Accessories Club, PO Box 200, Bedford MK40 1YH, England. Tel: 011 44 234 741152. Philips offers professional discs for testing players (contact them at NAP Consumer Electronics Corp., PO Box 309, Greeneville, TN 37744-0309), as does CBS (their professional disc, which includes signals to test the low-level linearity of the DAC, is available for around $40 from Old Colony Sound Lab, PO Box 243, Peterborough, NH 03458).*

Why no Television?

Editor:

In Vol.10 No.1, John Atkinson mentioned that he felt it important not to have a television in the same room as his hi-fi system. Why?

Patenotte's
Yazoo City, MS

I do not have a TV in my listening room because it is not a good idea to have any large resonant and/or reflective structure between or near the loudspeakers. With speakers having wide dispersion, the reflections of the sound from a TV positioned between the speakers will obscure the stereo imagery, and its cabinet resonances will both add colorations and affect the precision of imaging in the frequency region where the resonance lies. With a system capable of very high resolution, this will negate a lot of the additional performance for which you've paid. **—JA**

Toeing-In

Editor:

My question concerns the "toeing-in" of loudspeakers. Is this only appropriate for some kinds of loudspeakers and not others? Is it dependent on room acoustics? Or neither? Or both?

The manufacturer's polar patterns for my KEF R104/2 speakers indicate that the full range of sound lies on the reference axis. This being the case, would it not be appropriate to toe-in the speakers so that their axes face the listening seat?

What effect does toeing-in have on the various attributes of fidelity: imaging, soundstaging width and depth, the size of the listening "sweet spot," and so on?

Avron D. Gershen
No address supplied

The main purpose of aiming loudspeakers so their axes converge on the listening seat—so-called "toeing-in" —is to broaden the width of the seating area from which one can hear a balanced stereo stage. It is consequently appropriate for any speaker system, in any room, where parallel aiming causes a loss of channel balance when one listens from off center. It is also, frequently, of value in tightening and stabilizing the center image when listening from dead center.

Obviously, if you rarely listen in the company of others, imaging should be optimized for the center "sweet spot." This is often best accomplished by toeing the speakers in so as to converge those segments of their polar patterns which produce the smoothest, widest-range output. (In all cases, both speakers must be toed-in by precisely the same angle.) Then, because phasing interference can often introduce severe cancellations unrelated to polar response, some additional tweaking of toe-in angle may be necessary to optimize the imaging.

Remember that center imaging is also affected by the distance between the loudspeakers. With most speakers, it is advisable to experiment with separation distance when they are aimed parallel, before trying to tweak their toe-in. With minimally miked recordings, proper speaker separation will give a fairly even distribution of stereo images from left to right across the stereo stage. (Start with the

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speakers about as far apart as they are from the center listening seat.) Then experiment with toeing-in to optimize center imaging for the listening position(s) you wish to provide.

Practically all loudspeakers exhibit some vertical-venetian-blind effect—a tendency for images to hop back and forth as you move your head to either side of center. (VVB is as much a result of phase interference as amplitude differences.) The effect is most easily heard when the signal source is a mono A+B pink-noise or white-noise hiss, as from an unmuted FM tuner set between stations. For center-seat listening, the name of the game when toeing-in is to minimize VVB for the speakers you are using. For off-center listening, aim for the toe-in which gives the least apparent change in spectral and channel balance when you move off center. If this is best achieved with the speakers' axes crossing somewhere in front *of the center listening seat, don't hesitate to use that setup, but bear in mind that extreme HF response may be diminished when the speaker axes are aimed more than about 15° to either side of a listener. The ideal toe-in angle is that which best balances the imaging without compromising overall frequency response any more than necessary.* **—JGH**

Biwiring

Editor:

Would you please devote a few words to the subject of loudspeaker biwiring? What are its advantages? Disadvantages? Is it of more value with low-impedance loudspeakers? Should one always use the same kind of wire for both circuit lines? If not, which should get the better wire: the LF or HF leg?

Michael H. Prager
No address supplied

The most important advantage of biwire loudspeaker connection—running separate full-range cables to the LF and HF sections—is that it allows you to use the best kind of wire for each signal path. The LF line should be as heavy as is practical, to get as much of the amplifier's damping factor as possible to the woofers. Wire gauges of 10 and below are not overkill here. The HF line can be lighter (12 or 14 gauge), and should be designed for the best possible transmission of highs. That is, wires should consist of many small strands rather than one or a few heavy ones (as in the LF line).

As you guessed, biwiring is of greatest value when the speaker's woofer section has a very low impedance within the audible range, because the power and damping loss can be minimized by the heavy cable without compromising HF performance. The only disadvantage of biwiring is the slight additional cost, usually far outweighed by the ability to use cheaper cables that don't have to be designed to handle a wide range of signal frequencies. **—JGH**

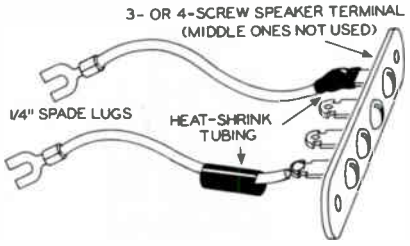
It has also been proposed, by Martin Colloms and Malcolm Hawksford, that by returning the ground connection for each leg of the crossover to the ground terminal of the amplifier, biwiring minimizes interaction between the high-pass and low-pass signals. This appears to maximize transparency for a given loudspeaker design. **—JA**

Spade-Lug adaptors

Editor:

In his review of the McIntosh MC 7270 amplifier in Vol.10 No.2, JGH mentioned that the ¼-inch terminal strips on its rear didn't allow spade lugs to be used. Could he suggest how to make up an adaptor?

Will Dunnett
Tallahassee, FL



Oversized (⅜-inch) spade lugs can be attached to a ¼-inch barrier strip by means of a simple adaptor harness.

The harness is made up from a three- or four-pin screw terminal, two short (4-inch) pieces of heavy gauge (#12) zip cord, and two ¼-inch spade lugs. Solder these together as shown, and if you need instructions about

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how to use the harnesses, you shouldn't be let loose with a screwdriver, let alone a soldering iron!

For added durability, you might add ½-inch lengths of shrink-fit tubing over the soldered connections at both ends of the leads. For added convenience, you might also mark one screw head and its corresponding wire (right behind the spade lug) on each harness with bright red nail polish, to identify that circuit as the Hot (+) connector. **—JGH**

Accuracy or music?

Editor:

There seems to be something wrong with *Stereophile*'s assessment of CD players. If a Compact Disc more closely approaches the sound of master tapes, then surely it will be more enjoyable? The better CD players, therefore, will be those that always produce the most enjoyable sound. How, then, can JGH contrast "accuracy" with musical enjoyment, implying that the more accurate players consistently produce a less enjoyable sound?

Marvin Katz
New York, NY

There have been CD players through here which have had some very appealing sonic qualities, but which I suspect of being rather highly colored. Three that come to mind were the California Audio Tempest, the PS Audio PS-1A, and the Meridian Pro-MCD. There were things about the sound of each of these that I loved, but there were also compelling reasons why I was not prepared to declare any one of them to be a state-of-the-art CD player. These reasons had mainly to do with the fact that each player had a unique sound, none resembling any other player I have heard. For that reason, I can state with absolute assurance that at least two of them were "inaccurate" reproducers, leaving open for the time being the question of whether or not the third was equally inaccurate.

Now, contrast those players with the situation shaping up with the high-end Japanese players. As the Japanese incorporate into their designs more and more of the features we now believe to be important for sonic quality (oversampling, digital filtering, DC-coupled audio circuitry, improved power-supply integrity, and so on), their players manifest fewer and fewer of the active irritations we heard from the first three generations of Japanese players, and *their sound becomes more and more alike. Converging sound quality is usually a sure sign that components are approaching Ultimate Truth—suggesting that what we hear from these CD players is probably quite close to what the original master tapes sound like. Of course, we are not required by moral or civil law to* like *the sound of those master tapes, but if we truly believe that the objective of perfectionist audio is to reproduce the signal source as literally as possible, then I believe we must acknowledge the likelihood that the cream of the Japanese CD players better qualify as perfectionist devices than do the apparently more colored (if sometimes more pleasant-sounding) players from the smaller, more "audiophile-oriented" firms.*

For example, I am convinced that some of the remarkable soundstaging characteristics attributed to exotic CD players are the result of nothing more than apparent frequency-response aberrations. They don't, in fact, have any such aberrations, but they sound *as if they do. In particular, I refer to what sounds like a withdrawn midrange. This makes everything seem more distant, but also tends to heighten the illusion of front-to-back depth and to make the acoustics of the performing hall more apparent. That these are attractive qualities is unarguable; that they are distortions is, to me, equally unarguable. Thus, we have a rather odd situation where CD players which I would conjecture as being more accurate (the top Japanese ones) are seen by perfectionists as being inferior to those apparently less so.*

Humanity has a long and colorful record of believing what we want to believe, rather than what reason and observation tell us to be true; it is not surprising that we should do the same thing with CD players. On the other hand, I feel I there is every justification for stating that the audiophile's professed desire for accuracy and his frequent preference for colored sound are inconsistent. In fact, at the risk of incurring the wrath of many of our readers (and the raised eyebrows of the As J and L), I will even dare to say that it is wrong!

Even more wrongheaded, in my opinion, is the average perfectionist's blithe assumption

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that the sound he hears from analog discs is more "correct" than what he hears from the best CD players. Anyone who tries different phono cartridges from time to time knows how different they are in terms of spectral distribution alone. With CD sound becoming more and more uniform from one player to another, it becomes increasingly difficult to support the view that any given phono cartridge is the standard against which CD should be judged, no matter how musically agreeable it may sound. It is quite easy to assemble a system which makes CDs *sound astonishingly musical, at which time your favorite cartridge is likely to sound quite poor. Yet, to me, that is the only sensible way to go these days, simply because the evidence suggests that it is more often the sound of the CD, not the analog player, which most closely approaches "reference quality."* **—JGH**

JGH and I have been round and round on this one only about 1000 times, but the simple fact is that he has no basis to make the argument he is making. A comparison between different CD players compared directly to a master tape will *give you an indication, but you won't know what has been indicated. Comparing various renditions of phono playback to their master tapes* does *tell you how accurate the phono playback is (up to a point, because you have to ignore the degradations introduced by mastering electronics, cutterhead resonances, vinyl differences, etc.), because the record is made from that tape playback. This is not the case with digital. The master tape machine could have quite inferior playback electronics, but it would have no effect on what went on the CD, since it goes on the CD in digital form. Therefore, there actually* is *no way to know what the digits on the master tape are supposed to sound like. We can even posit the hopeful situation that digital playback will keep getting better and better, just as analog has (though we realize that there are limits to what improvements can be made).*

Even if master-tape playback were a reliable standard, JGH has no right to assert that the "one-face" Japanese machines are more accurate than the "many-faces" high-end machines, because he has not done any of these master-tape comparisons. I'm sure the recording engineers would be in a decent position to evaluate the quality of various CD players, but they ain't talkin'—and probably don't spend a lot of time playing back CDs in any case.

JGH does make a good point, however: it is *possible to set up a system in which CDs sound, if not astonishingly musical, at least musical. He has done so. What is really interesting, though, is that his LP reproduction on this system, using a cartridge carefully chosen to approximate the* tonal balance *of CD, does not sound "quite poor." In fact, it sounds, as JGH will reluctantly admit from time to time, significantly better than even the best of CD reproduction, particularly from the "one-face" players. Even more interesting is that in this listening room the best sound of all, to my ear, comes from JGH's 20-year old analog tapes made on a venerable, but not audiophile-tweaked, Revox A77.* **—LA**

BUILDING A LIBRARY

The Vaughan Williams London Symphony

Illustration by Jim Wood

Christopher Breunig

Just how subjective our conclusions about music performance are was brought home to me when I began to prepare this piece on Vaughan William's second symphony, *A London Symphony.* (The work was prompted by fellow composer George Butterworth, to whom the final revised score was dedicated, after he was killed in the futile trench fighting of the Great War. Before that, a manuscript was sent to Fritz Busch at Aachen, in 1914, and lost; the score was reconstructed from parts, then cut and

and revised by the composer, in 1918, 1920, and 1933.)

The English conductors most closely associated with Vaughan Williams's music are Sir Adrian Boult, who first performed the *London* in 1918 (he premiered Symphonies 3, 4, 6), Sir John Barbirolli (to whom 8 was dedicated—he also premiered the *Sinfonia Antartica*), and Sir Malcolm Sargent (first performance of 9). Sargent recorded very little; Boult and Barbirolli each made two recordings of the *London*: with the LPO, Decca 1951 and EMI '71, and with the Halle, 1957 and '67. respectively. The earlier, and more vital, Boult version is long gone, while the older Halle recording, curiously, has just reappeared on Compact Disc in the UK. The other two recordings can be had on LP, Barbirolli's happily remaining in analog on EMI Eminence (EMX 41 2087 1), Boult's digitally remastered, to its disadvantage I think, on EMI 29 0331 1. On CD the Boult, with the *Tallis Fantasia*, is on CDC 747 2132. Encouraged by the success of the *Antartica*, conducted by Bernard Haitink, EMI has now recorded him in the *London*—and that will be interesting.

Two other "second-generation interpreters" are Vernon Handley, given much encouragement in his formative years and now a conductor of real stature, and Andre Previn. I was lucky enough to attend the sessions for the latter's later RPO *London*, coupled with a rather fine *Lark Ascending*, and assessed by George Graves in Vol.10 No.4 (p.185). The sessions were held in Croydon's Fairfield Hall, a post-war auditorium 20 minutes away from central London often used as a "dress rehearsal" venue for the City's under-prepared, under-financed concert performances.

Telarc's Jack Renner found the sessions interrupted first by tinkling cups and saucers, from the kitchens under the stage (!)—the building also houses a theatre auditorium—then by fan-noise, heard as spurious LF signal. One could only admire Previn's good humor and professionalism while these gremlins were "ghost-busted."

Another visitor to Fairfield, critic Edward Greenfield, talked of how he'd enjoyed RCA's CD release of the early '70s LSO cycle. Previn's response was not exactly ecstatic: he'd changed his way with the pieces, not least the *London*, of whose *finale*, with its moving Epilogue, he now took a more sombre view. "That's one hell of a funeral march," he observed.

Although EMI, Decca, and RCA were recording, experimentally, in stereo in 1954-5, they were beaten to the start in the UK by the small group Pye. Barbirolli began a series of recordings with his Manchester orchestra, the Halle, for Pye in June '55 (Vaughan Williams 8, allegedly with a Mercury production team—something to excite *TAS*!). He returned to the EMI fold for a remake of Vaughan Williams 5 in May '62. Generally, the stereo imaging on Pye was vague; the reproduction of first-desk players was close and thin, and LP pressing quality was notoriously poor.

But "Glorious John," as Vaughan Williams dubbed him, was a key figure in my musical upbringing, which at that time included the new-music festivals at Cheltenham, in the Cotswolds. The '58 concerts climaxed in a radiant account of the *London*, after which the composer appeared on the platform. A large figure in bulky tweeds and hearing-aid, he seemed, if anything, bewildered by our acclaim. In a characteristic speech, Sir John confided that at the morning rehearsals he had said, "I wish I could score like that now!" Did he mean score, I sometimes wonder, or was he referring to the actual musical material? For some, the orchestration will seem rather opaque and heavy, the dynamic scale calling for plenty of low-register winds, percussion with side-drum, glockenspiel, jingles, cymbals, and bass drum.

Therein lies my excuse for trying to coax you away from the sumptuous Telarc (CD-80138), with its massive power, almost unsociably wide dynamic range (all as GG has said; in fact, this wasn't just the "classic" miking job—the hall was too difficult for that), and toward the dated, almost subfusc '57 Halle production. The latter is certainly inferior in execution to the RPO's, with some abrupt left-right separation, relatively narrow dynamics, and tape hiss. But this PRT CD (PVCD 8375), with its lack of liner notes and quite abysmal artwork, is not only a high-quality remastered disc, but enshrines a quintessentially Barbirollian reading that says so much more than the new Previn.

You can hear this not just in detail, in the contrasting ways these conductors present the Westminster ("Big Ben") chimes, or etch in the

cymbal part at the beginning of the *allegro risoluto* (i), but in the characterization of the *scherzo* (described by the composer as a Nocturne), too bland by far in the Previn/Telarc. I have to add that the superior London Symphony Orchestra produces finer, more atmospheric work in the opening Introduction on Previn's old RCA disc, although there the interpretation is more episodic, less consciously "symphonic."

There you have, though still structured "classically," a movement conjuring images of street urchins at play, with imitations of accordion. In Barbirolli's later recording he had become more expansive, but there was more orchestral refinement: in, for example, the quasi-fugal writing, the string attack is rougher on PRT. (Incidentally, the PRT is a bargain-priced CD in the UK and, uniquely, offers an access point for the Epilogue.)

It is hardly surprising that Barbirolli excels here: he used to delight in describing himself as a true Cockney, born in Southampton Row over a baker's shop, the son of Italian and French parents.

And what of Sir Adrian's *London*? Yes, I have enormous respect for his two recordings; the older Decca has a scherzo that fizzes along, with brilliant wind voicing, triangle touched in to perfection, the whole nicely suggestive of the music's darker undercurrents. Later on, though, he came to sound as if "presiding" over the score. The later scherzo is clear, accurate, responsible, but lacks the "demon." I don't think the *London* is meant to be this coolly detached, though one understands Boult's abhorrence of wallowing.

The EMI recording comes from a technically very fine cycle, but, digitized, has lost the feeling for spatial layout, and fidelity to timbres. Nor has the nasty little edit which comes at the end of the horn passage in (ii), just before the magical, quiet, throbbing figures for strings lead to viola solo with woodwind dialogue, jingles suggesting the old Hansom horse-drawn cab, been disguised.

Even so, there is some magnificent playing at the brassy climax to (iv), stilled by the return of the Big Ben (harp) and the murky waters of the Thames. But the old Halle version renders still more detail, and reflects the greater commitment. **S**

BOOK REVIEWS

***The New Sound of Stereo* by Ivan Berger and Hans Fantel.**
265 pages, hardcover $12.95. Published by New American Library, 1633 Broadway, New York, NY 10019.

Ho-hum. Another book about hi-fi, right? Well, yes, but while this still isn't *the* book about high end we've all been waiting for, it's the newest and best about mainstream audio to date, and does at times acknowledge the existence of a higher end.

Ivan Berger is technical editor of *Audio* magazine, and one of the few people in mainstream audio who listen. Hans Fantel, audio and video columnist for *The New York Times*, is a recognized mainstream tin-ear who ingratiated himself with the high-end fraternity a few years ago by declaring, for the edification of millions of *Times* readers, that there was no justification for ever paying more than $1000 for a pair of loudspeakers. It's hard to tell who wrote what parts of this coauthored book, but the result is a surprisingly good balance of solid technical information, clarity, and readability.

Published in 1986, the book is as up-to-date as it could possibly be, excepting any mention of the new digital audio tape format (which, as I write this, has yet to be introduced to the US, and may not be if Congress continues to press for a Copycode bill—see Vol.10 No.5). I did not find a single factual error in the whole book, nor did I find many expressions of opinion with which I could disagree. The authors recognize the existence of high-end audio and its sometimes extremist views, and often cite them as an alternative outlook. For instance, there is mention, without comment, of the fact that some critics of CD believe it to be inherently flawed by inadequate standards of sampling rate and digital word length.

The book's only lack is at the ends of the knowledge spectrum: there is nothing about basic physics or electronics, and no one topic is treated with the depth necessary for full understanding. Otherwise, it's an excellent primer, both for the beginning audiophile and for the practicing aficionado who needs to fill some holes in the fabric of his audio knowledge. **—JGH**

***Principles of Digital Audio* by Ken Pohlmann.**
285 pages, softcover $19.95. Published by Howard W. Sams, 4300 W 62nd St., Indianapolis, IN 46268.

Ken Pohlman is best known in the audiophile fraternity as the author of the digital audio column in *Audio*, and as a frequent CD-player reviewer for *Digital Audio* magazine. Professionally, he is an associate professor and Director of the Music Engineering Department at the University of Miami, Florida. He is also, possibly, the most knowledgeable writer on the subject of digital audio in the US.

Principles of Digital Audio covers the field thoroughly and in considerable depth, omitting nothing even remotely pertinent to digital audio recording and reproduction. The index is unusually complete and useful, and is sensibly organized in a semi-chronological manner, following the course of an audio signal from the air through the digital mill and back to the air. The book can be read from stem to stern as a textbook, or—thanks to the complete Index and Table of Contents—used easily as a reference source.

The writing, however, is amazingly variable in quality. Some of the explanations are inspired in their lucidity; others are so convoluted as to be almost (but not usually totally) incomprehensible. The problem in many cases is the author's assumption that readers are as familiar as he with advanced mathematics. A simple "See Figure 1" will, for example, refer the reader to a staggeringly complex full-page electrical matrix, for which no explanation is offered. It is usually possible, with some perseverance, to figure out what is going on, but a few words of clarification here and there would make the going a lot easier.

In other cases, awkward syntax and inappropriate punctuation frequently make it necessary to go back and reread a sentence several times before you can figure out what it is trying to say. The chapters on error correction will be particularly hard going for the average audiophile, but the principles are spelled out clearly enough to provide a conceptual grasp of the subject, which is really all that anyone not actually into digital design will need to have.

The fact that the book is not consistently well-written or edited, however, pales before the monumental feat of writing such a thorough text on such a complex subject. I hope all the copies of this sell out, so Ken can consider writing a revised edition.

If you will ever want or need to know anything or everything about modern digital audio recording, this is the book to have.

—JGH

***Audio IC Op-Amp Applications* by Walter G. Jung.**
225 pages, softcover $17.95. Published by Howard W. Sams, 4300 W 62nd St., Indianapolis, IN 46268.

Walt Jung, familiar to many from his informed writing in *The Audio Amateur*, is that ideal audio person: the engineer who listens. Now in its third edition, his book will be found immensely useful by any audiophile interested in the practical implementation of audio circuits using op-amps. New material covers servo control of low-frequency response, the use of such high-performance chips as the Analog Devices AD711, PMI OP-27, and Burr-Brown OPA606 (introduced since the publication of the second edition), and, most interestingly, a chapter on the design of CD-player output circuitry. Highly recommended. **—JA**

***Electronic Connections* by Martin Clifford.**
429 pages, hardcover $24.95. Published by Prentice Hall, Inc., Englewood Cliffs, NJ 07632.

Martin Clifford is one of the most prolific home-electronics authors in the US, having turned out enough general titles (*Complete Guide to Satellite TV*, *Complete Guide to Video*, *Complete Guide to Car Audio*) that he can now consider turning to more abstruse subjects, such as this one.

Electronic Connections is unique: a book devoted entirely to hooking things up. Starting with a discussion of different kinds of wire and plugs and their applications, including a section on making your own interconnects, it then takes up component hookups of increasing complexity, from AC outlets to complete audio/video systems.

This is a goldmine of information for the person who installs audio, video, and car hi-fi equipment for himself or for others, but the book has little to offer the serious audiophile beyond the basics of system hookup. The section on audio interconnects, for instance, shows that the author has not kept abreast of our field, and reads like something Julian Hirsch might have written. The only things deemed important are DC resistance and whether or not the wire flexes easily.

This is a useful book for the beginner and the casual dabbler in home electronics, but at $24.95, it is of very questionable value to the advanced audiophile. **—JGH**

MANUFACTURERS' COMMENTS

Souther tonearm

Editor:

Talk about coincidence! In the latest *Stereophile* recommended components listing for our Triquartz tonearm (Vol.10 No.3), there is a note by JA that our spindle locator could ". . . introduce an arbitrary degree of rumble." We had investigated this potential problem of induced noise during the original design period. After much testing we finally concluded that, when using a Delrin or Teflon insert for the spindle locator, the problem could only occur when the spindle was rough and/or eccentric.

Having no control over turntable manufacturers, we could only attempt to correct the spindle by polishing. To this end we have included a spindle-polishing kit with all Southers for about three years. Spindles with gross surface defects or major eccentricities could, of course, not be corrected by surface polishing, and we contacted turntable manufacturers so that they might address this problem from their end. (In all fairness, the condition of the spindle will be an issue only with our tonearm. We thank those manufacturers who have kindly addressed our unique problem in their turntable production.)

Now the coincidence. We were just in the process of announcing to our dealers the final solution to this potential problem at almost the exact moment the *Stereophile* listing was published!

During one European show, our Swiss distributor, Reza Oskoui, showed us a way we might move the spindle locator up off the spindle, after the initial set-up adjustment. Back at the lab, we set about improving upon this concept. Now we offer an accessory which is compatible with any vintage Souther SLA to eliminate the potential for spindle-generated rumble. Once this simple, 15-minute installation is complete, the spindle locator can be unscrewed up off the spindle, thus eliminating all contact.

We haven't come up with a "suitable" name for this new accessory. Current candidates are "Souther Supporter," "Reza Riser," "Lou's Lifter," etc. Any help with this most difficult problem will be appreciated greatly. We have established a rather modest price of $60 for this accessory.

Lou Souther

Souther Engineering Corporation, Canton, MA

Marantz ST551 tuner

Editor:

Thank you for the kind review of our ST551 tuner. We are pleased that you found our rated specifications to be conservative and that the unit actually measured much better than spec. This represents the conservative design philosophy we follow when developing Marantz products.

When designing the ST551, we gave our attention to those circuit characteristics which have the greatest influence on sonic quality. A product designed to be affordable cannot be optimized for every aspect of its performance, so we concentrated on those elements that are most significant to the majority of our customers. Therefore, the ST551 shines in the crowded airwaves of the urban environment where signals are strong and close together.

We believe that the AM section of your sample was not misaligned. All alignment procedures involve trade-offs among the various parameters. We fear that the realignment you describe may result in an unacceptable deterioration of the signal-to-noise ratio.

Your readers should also be aware that the availability of the ST551 may become limited. Marantz Company has recently changed ownership, and the new management has a strong commitment to increasing the overall quality of the Marantz product line. Therefore, all of our products are being re-evaluated, and some may be replaced with improved models.

Verle Rader

Marantz Company, Inc., Chatsworth, CA

Magnum Dynalab 205

Editor:

Most will agree that it is a good day when a magazine of *Stereophile*'s stature feels that your product is worthy of public interest by making

comment, be it good, bad, or indifferent.

However, it becomes a little difficult to take when, for no reason other than pure speculation, we find ourselves accused of putting cost before safety—a submission that has absolutely no basis in fact!

The issue DAS appears taken with is the fact that our 205 is not equipped with a fuse. He says this is "an undesirable cost *vs* safety measure." If this is true, he must also advocate that every appliance, including his toaster and coffee maker, should be equipped with a fuse as well.

Every component we manufacture that operates off 120V AC must first be certified "safe to operate" by our own Canadian Standards Association (CSA). In the case of the 205, their assessment is that a fuse is not necessary for safe operation. As a matter of fact, it draws only about as much current as a plug-in AC power supply for a portable radio. Come to think of it, I haven't seen a fuse holder in one of those either.

In the case of our FM tuner, with its somewhat higher current draw, safety standards require that we fuse our FM tuner internally. Incidentally, our components have been submitted to UL as well.

Like any conscientious manufacturer, Magnum Dynalab takes the safety of its customers very seriously. While I am quite sure the concern DAS has expressed is sincere and well placed, perhaps you would consider having him do a piece called "Fusing for Safety," within which he might explain his views more fully, to the benefit of us all.

Again, thanks for your policy which allows us the opportunity to comment in parallel.

M. C. (Marv) Southcott
Magnum Dynalab, Brampton, Ontario, Canada

NAD 6300 cassette deck

Editor:

Thanks for reviewing the 6300 cassette deck. We are delighted that Mr. Graves was so impressed with the sound, but disappointed in his infatuation with automatic biasing. Our goal with the 6300 (and all NAD products, for that matter) is to produce a product which sounds better than competitive products in the same price range. To do this, we focus on basic, efficient circuits which enhance the real-world performance of a product and eschew more costly items which, while they may offer some convenience, are not necessary for superior sound reproduction.

While we agree that automatic biasing is handy, it would add significantly to the cost of an already expensive deck. In our experience, autobiasing is of limited benefit to the serious recordist, who will buy the best-sounding tape he can find and stick with it until it is discontinued by the manufacturer.

On the other hand, the "car" circuit cost-effectively addresses the everyday problem of squeezing a wide-dynamic-range CD into the limited dynamic range available in a moving auto. The play trim circuit is a simple and effective way to correct not only for azimuth errors, but for differences in EQ (common in cassette decks) and the progressive high-frequency losses in favorite tapes that have been played too often.

In the final analysis, the 6300 is the alternative for critical audiophiles who want the sonic performance of a world-class cassette deck but are put off by a $1500 price tag.

Peter D. Tribeman
NAD, Norwood, MA

Kevek ES.6 speaker

Editor:

Thank you very much for the excellent review of the Kevek ES.6 (Vol.10 No.5), especially since we were included with more expensive models from some of the larger speaker companies.

It is also gratifying to discover that you were accurate in most of your comments, indicating that indeed you really did listen carefully for the subtleties an untrained ear would not notice. Generally, I am skeptical about the abilities of many reviewers. Several months ago, the crossover was modified to improve the central stereo image, and the vertical dispersion that you noted also improved. This had come to our attention when certain program material was auditioned; we discovered that some respected recordings were not as revealing as others. Your earlier comments about source material and its dubious accuracy is probably the most valid statement I have ever seen in print. We now pay particular attention, at all frequencies, to phase shifts that affect imaging, and have developed our own technique for testing this specific area. At the

same time, we incorporated a minor change to the crossover in order to flatten the tweeter, removing the sibilance noticeable on certain programs.

I have to state that we too had reservations about the effects of the graceful grille shape, but numerous tests, including the placing of reflection absorbers, have shown no significant changes. We did discover that an irregular response, with peaks and troughs, became more exaggerated with certain cabinet and grille configurations. This confirmed that a smooth balanced response is still of prime importance, for many reasons other than just this one.

I am encouraged by your comment that you would like me to design a speaker in the $900 region. In June we introduced the three-way ES.8 at $900, with the criteria of retaining all the virtues of the ES.6, plus more dynamic range and authority in the lower octaves without the added chestiness encountered when larger woofers are utilized.

John Bradford
Reference Monitor International, Inc.,
Carlsbad, CA

Snell A-III and C-i speakers

Editor:

Thank you for the opportunity to comment on TJN's review of our Snell Type A-III and C-i loudspeakers used in their bi-amped configuration. TJN has a misconception which is shared by many audiophiles and reviewers alike. He states that: "Theoretically it (bi-amping) should provide a cleaner bass response (the elimination of the passive crossover means there is now no choke in series with the woofer)." He goes on to note: "There is one significant limitation to bi-amping the C-is: there is no way to bypass the passive crossover choke in series with the woofer . . ." The fact is that in any well-designed loudspeaker, the resistance of the choke is carefully integrated into the total design. Any change from its proper value would actually degrade performance. For example, if a speaker had an inductor with a DC resistance of 2 ohms, a system "Q" of 0.7 (the generally preferred value providing low-frequency extension with minimal ringing), and a -3dB point of 24.9Hz; eliminating the 2 ohm inductor would change the "Q" to 0.553, and raise the -3dB point to 33.3Hz! This is obviously not an improvement.

Any Snell Acoustics loudspeaker can be bi-amped *without* an electronic crossover, or with the crossovers specially made for Snell Acoustics loudspeakers by several manufacturers. This means that bi-amping is not necessarily a complicated or expensive proposition. In many cases bi-amping with two inexpensive amplifiers is a better value than using one more expensive amp. Two important problems which we have found in moderate- and low-cost amplifiers are their ability to drive real-world loads properly, and their channel interaction (as is apparent from the image change when comparing a stereo amp *vs* a channel from each of two identical amps). We have devised a neat solution which we call "vertical bi-amping." In this configuration, a stereo amp is used for each speaker, rather than the typical set-up with one amp for low frequencies and another for high frequencies. Each amp's power supply only needs to deal with one low-frequency channel; and the left and right high-frequency channels are totally independent, offering ideal imaging characteristics. (Of course, the two amps must be identical.)

We are pleased that TJN feels, as we do, that the Type C-i manages to retain many of the Type A-III's sonic qualities at a much lower cost, but it must be pointed out that TJN's room was an excellent acoustic environment for the C-i's. A speaker's "power response" (the total energy it puts out into the room) has a great effect on its tonal quality and the way it interacts with its acoustical environment. This is an area in which the Type A-III beats the C-i (and anything else we know of) by far. The Type A-III is able to sound neutral in a wider range of listening rooms than the C-i; therefore, comparisons of the two must be taken in the context of the room used for the comparisons. A less flattering room for the C-i would have revealed the A-III outperforming it by a wide margin.

Kevin Voecks
Snell Acoustics, Haverhill, MA

Klyne SK-5A preamplifier

Editor:

With respect to Mr. Holt's review of our SK-5A preamplifier, we offer the following comments to help put it in proper perspective. First, we

are perplexed as to why you would give this preamplifier, a design which enables the listener to optimize the performance from any MC cartridge, to a reviewer who, as judged by the comments and conclusions in this review, is largely ignorant of, and apparently unappreciative of, the moving-coil cartridge technology. As Mr. Holt has suggested, the low-output moving-coil cartridge is, in fact, the preferred cartridge for the highest-quality phono playback. His categorical denunciation of this class of cartridges serves only to confirm his misunderstanding of their proper use and to demonstrate his unfortunate bias with regard to their value.

Second, his premises and (illogically derived) conclusions suggest to the reader that the flexibility designed into the SK-5A are "tweak" features, that they are unnecessary items that "appeal to the needs of what I (Mr. Holt) call compulsive tweaks"; and, that their existence encourages the user to "endlessly diddle" with the settings. These ideas are fabrications by Mr. Holt and are simply wrong. Let us look at the facts.

Every cartridge, whether it be moving-coil or moving-magnet, has a mechanical resonance. Because these transducers are velocity generators, this mechanical resonance will produce an increased output voltage at the resonant frequency. For most low-output MC cartridges, this resonance typically occurs between 30 and 50Khz, and typically ranges between 6dB and 12dB in amplitude. Also, as with any resonant system, there is a corresponding phase shift below the resonance frequency which extends well into the audible high-frequency region.

Users of moving-magnet cartridges are familiar with the need to load these cartridges with a capacitive load. This load, the value of which is usually specified by the cartridge manufacturer, appears in series with the relatively high inductive impedance of the MM cartridge and contours the high-frequency response of the cartridge to be nearly flat. This is the function of the input capacitors of the SK-5A which should be used in conjunction with MM cartridges. This is hardly a "tweak" feature, and I am sure that Mr. Holt has himself loaded MM cartridges in this fashion for many years.

MC cartridges, however, have a very low source impedance (frequently less than 10 ohms), so that loading them with a capacitive load has virtually no effect on their high-frequency response. Thus the user must either live with the resonant peak (as is the case with most phono preamplifiers on the market) or have some electronic method by which the cartridge output at resonance can be flattened. This is the function of the set of High Frequency Contour switches in the SK-5A. The switches provide a passive high-frequency contour of 6dB per octave with a wide range of available corner frequencies, as well as providing a considerable degree of phase correction in the upper portion of the audible high-frequency range.

These switches, and those that provide capacitive loading for MM cartridges, provide the opportunity to achieve a flat frequency response from most cartridges, as well as some degree of phase correction for MC cartridges. We cannot understand why Mr. Holt would consider features enabling the user to achieve flat frequency response from their phono pickup to be "tweak" in nature. We can only conclude that he simply does not understand them.

The phono gain settings allow the user the opportunity to optimize gain of their MC cartridge relative to the level line input sources such as tuner, tape, and CD. Low-output MC cartridges which are likely to be used with the SK-5A range in output from 0.1mV to 1.0mV, depending on the manufacturer. This translates to a 20dB difference between these extremes. While the majority of low-output MC cartridges falls within the 0.2 to 0.5mV range, we feel that it is a valuable convenience to be able to adjust phono level to match the level of other sources.

One of the most misunderstood aspects of MC cartridge use is the proper input-impedance load. Some have argued that very low loads be used because it reduces treble distortion by subjecting the cartridge to electrodynamic damping. Others have argued that higher loads (*eg*, 47K) result in a more open, "live" sound. The fact is that both these arguments have merit, but neither approach provides the best solution for the problems involved. Proponents of low loads have correctly recognized that MC cartridges do benefit from some electrodynamic damping, both reducing treble distortion *and* taking some of the edge off the

high frequencies due to the resonant rise in the response curve as described above. However, a very low load will not elicit frequency-response correction or phase correction. Also, by the time the load is low enough to tame the problems in the treble, the middle frequencies have become overdamped, and the sound (and soundstage) becomes restricted and lifeless. Thus, proponents of high cartridge loads are also correct, particularly in systems that may need a little extra high-frequency energy.

We feel that the best way to solve both these problems is to correct for the resonant rise electronically as described above; and then to *judiciously* use resistive loading at the cartridge input to provide a small amount of electrodynamic damping only to aid in controlling high-frequency distortion, without interfering with the cartridge's natural ability to trace the lower frequencies. As you might guess, we typically recommend cartridge loadings considerably higher than those recommended by "low-load" proponents. (Frequently manufacturers of low-output MC cartridges suggest on their specification sheets a load of 30 or 40 ohms. This number refers to the input impedance of a step-up transformer and should not be used as a resistive load.)

Mr. Holt implies that the phono amplifier settings are to be determined by ear. This is incorrect. The settings of the switches depend primarily on the cartridge used and to some extent on ancillary equipment in the system. The operating manual includes recommendations for settings of the most commonly used MC cartridges. These settings are based on measurements we make on samples of each type of cartridge. If a user has a cartridge type which we have been unable to measure, the manual provides guidelines on how to make the settings, based on technical specifications of the cartridge. Of course, anyone wishing to optimize the sound from their system will want to experiment because cartridges vary somewhat sample to sample. Also, each system will have a different high-frequency character and will require differing high-frequency characteristics from their cartridge.

Mr. Holt uses the Ortofon 2000 cartridge running straight in to the SK-5A and finds the noise level of the SK-5A to be too high. His statements in this paragraph are very misleading, as the Ortofon 2000 was designed *only* to be used with a step-up transformer. This cartridge has an output voltage of 0.05mV (50 microvolts); 12 to 18dB *below* most "low-output" MC cartridges. (When the output voltage is cut in half, the level drops by 6dB.) While it is possible to design a preamplifier with sufficiently low noise to work with such a low signal level, the sound would be of poor quality. (The technical reasons for this are beyond the scope of this discussion.)

Finally, we would like to know more about the test methods used to determine the "flavor" of the high-level section of the SK-5A. Our own bypass tests indicate that the line amplifiers are exceptionally neutral. Indeed, the primary difference between the SK-5 and 5A is the line amplifiers, and we have received unanimous raves from those who have had their preamps updated.

I sincerely hope that the above information is helpful in correcting much of the unfortunate misinformation in Mr. Holt's review. The ultimate irony of his comments is that the SK-5A has features which allow for the correction of those very shortcomings of low-output MC cartridges which he so abhors (although he disproportionately exaggerates their significance), and then refers to these features as "tweak," and criticizes the SK-5A for having them. Incredible?

Thank you for the opportunity to comment.

Stan Klyne, *et al*
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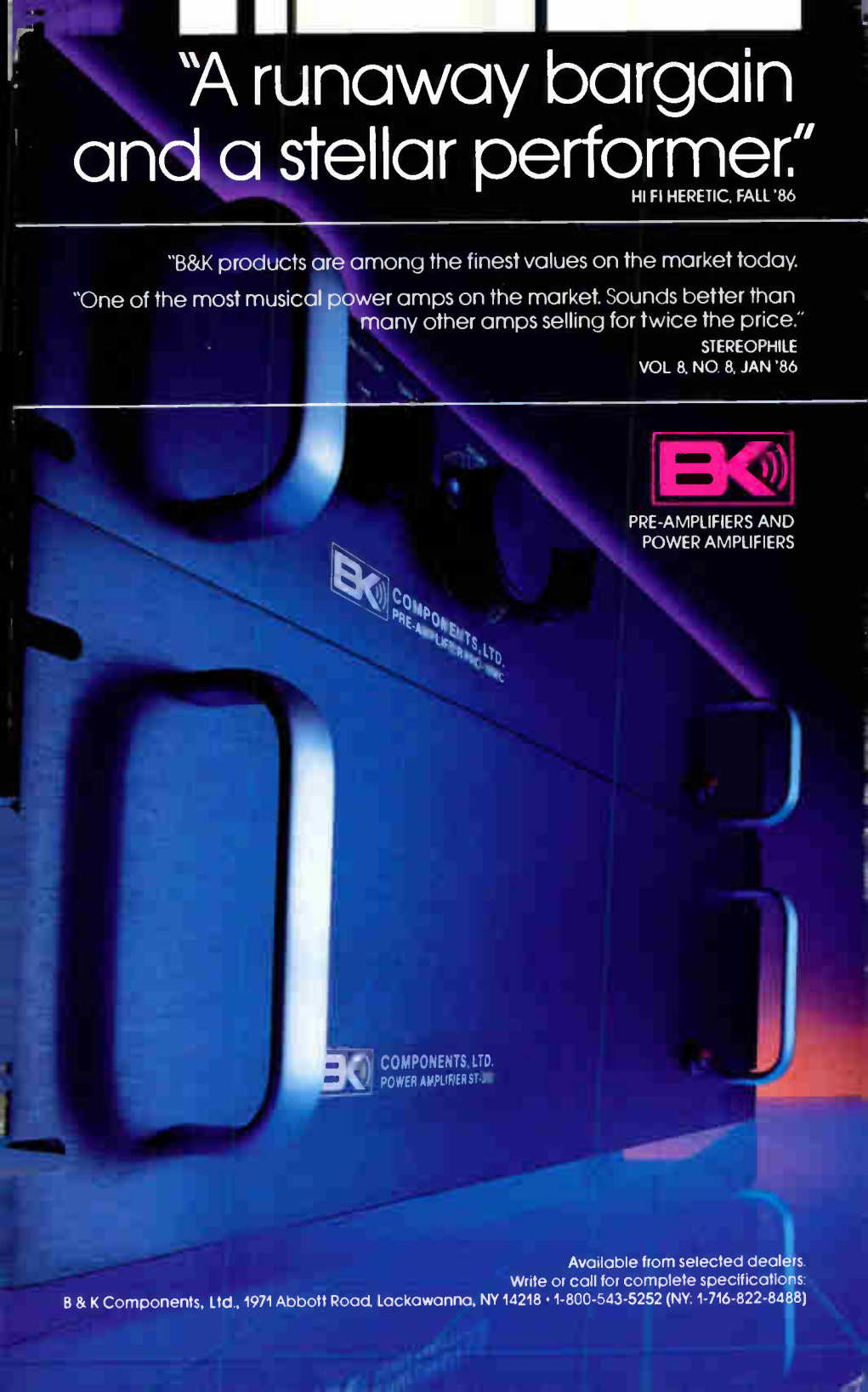

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