THE B.A.S. SPEAKER

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THE BOSTON AUDIO SOCIETY
P.O. BOX 7
BOSTON, MASSACHUSETTS 02215

VOLUME 6, NUMBER 4
JANUARY 1978

In This Issue

Winter is upon us again (in Boston, anyway), so pull a chair up to the fire and join Jacob Rabinow as he takes on Shreve, Slindee, and other SL8 modifiers. Jake suggests a few approaches of his own, and a wildly unconventional way of dealing with record warps.

We have a goodly number of reviews this month, especially of records. Equipment reviewed includes the Audio/Pulse Model One, the Fons CQ30, the Nakamichi/ADS car stereo system, the Paragon 12 and Trevor Lees preamps, the Acoustat X ESL, the ADS 810 and 910 loudspeakers, and the Fried loudspeakers. David Reiter follows up by questioning whether the equipment really matters all that much.

You'll also find more discussion of the RFI question and some tips on resonance fighting. That may not sound like fun, but it beats shoveling snow.

Coming attractions -- Look for commentary on the proposed BSO mike switch in the February Speaker and for an article by Tom Holman on FIM distortion.

Membership dues are $14 per year (October 1 to September 30) or portion thereof. Dues include a one-year subscription to the BAS Speaker. (Note that almost the full amount of dues is allocated to production of the Speaker. The local activities of the BAS are strictly self-supporting.) For further information and application form, write to: The Boston Audio Society, P.O. Box 7, Kenmore Square Station, Boston, Mass. 02215.

Ads are free to members. No commercial or non-hi-fi advertising will be accepted.

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*Quad ESL's, serial #36932, 36933, new condition, bronze grilles, original cartons, $700. B. Schurman, 100 Biscuit City Road, Kingston, RI 02881, phone (401) 783-3255 evenings.
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*Trevor Lees preamp, black face plate, silver knobs, extra set of Telefunken tubes, used eight months, $250; Netronics 450D turntable, $60. Harold Hopkins, (314) 474-7004.
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*Quad ESL speakers, condition almost immaterial; also, Quad 303 tube power amplifiers; will buy, but will also consider trade for Hermeyer ESL amplifier (original behemoth version), old oscilloscope, other test and audio gear, or design work or troubleshooting. Peter Cade, 33 Bentley Avenue, Jersey City, NJ 07304, phone (201) 589-3499 (days), (201) 432-9896 (evenings).
*Information on obtaining or using Decca or Fane ribbon tweeters. Robert C. Williams, 5279 Sunnyside Road, Mounds View, MN 55112.
*Trade Pioneer RG-1 dynamic range expander (unused) plus cash for used Soundcraftsmen equalizer. Bob Unternrink, 418 West Third Street, Lexington, KY 40508.

Background on BSO Mike Proposal

Elsewhere in this issue (BAS December meeting summary), you will find references to a proposal to change the mikes in Symphony Hall--Boston from Neumann KM-83's to Schoeps CMC-52UG's. The majority of the membership, living out of state, know nothing about this matter, and it is possible that locals also are unfamiliar with the details. Since the BAS has been approached and asked to contribute money toward the purchase of the new mikes, some background is necessary.

Last summer, WGBH-FM used Schoeps mikes (loaned by BAS member John Newton) for some of its Tanglewood Festival BSO pickups. A number of the people involved were impressed with the sonic improvement (relative to the Neumanns believed used before). This led Victor Campos, a BAS member and employee of Acoustic Research, to suggest on a local hi-fi oriented radio program, Shop Talk, that a fund be established to make possible the purchase of Schoeps for use in Symphony Hall. He estimated that about $1000 would be needed.

Victor has been able to collect some of the money through donations and that is about where the project stands. At the December meeting of the BAS it was suggested that the BAS contribute a few hundred dollars to the fund.

Meanwhile, Mike Riggs has been discussing the idea with several people and has uncovered some interesting information bearing on the idea of the exchange. The bulk of this material will appear here next month, but here is a brief overview.

Generally, those who have used both mikes think the Schoeps are "flatter," "smoother," and have a better line-driving capability. They also believe that much depends on the application to which the mikes are put and that one cannot be sure of results without some experimentation.

Riggs does not see why raising the money should be a problem -- at least that part (if any) which might be appropriate for the BAS to contribute; passing the hat to the tune of $5 a head at a typical meeting would go far. But people need to know whether they are getting their money's worth. At this writing, Campos is in Las Vegas at the Winter CES and thus no information is immediately available from him.

But it would seem that a test could be arranged. New York member (and Schoeps distributor) Jerry Bruck offers to lend Schoeps mikes for that purpose and is rather enthusiastic about the idea. He also has suggested experimentation with various pickup patterns. (The present setup uses spaced omni's, a system which many experienced recordists feel optimal. Others prefer co-incident miking techniques such as ORTF, or X-Y/Blumlein.) Riggs has not had direct offers but knows people closer to home who would probably do the same as Bruck.

If the Schoeps mikes were to prove superior, the substitution could be made permanent. Also, it is possible that the Neumann's might be sold to offset some of the cost of the Schoeps with the mike fund making up the difference. But again, this is only speculation; more information is needed.

Also, there are a number of not necessarily coincident interests at work here: that of the nationwide audiophile community, The Boston Symphony Orchestra's Transcription Trust, WGBH's, and that of WCRB/Charles River Broadcasting. The latter's may be the most overlooked to date; CRB is responsible to the Transcription Trust for the production of both the archival and distributed BSO performance tapes. It is a matter of money to them as well as of quality -- and potentially at least, a matter of effort if significant experimentation is undertaken.

As if this were not enough, the issue is surrounded by other questions. The frequency response of the mikes in question differ from each other in ways that might appeal differently to different listener tastes (the Neumann has a rather broad peak centered near 9 kHz, for example, while the Schoeps has a small amount of rolloff at 20 kHz). The Schoeps may need a windscreen to protect them from low-frequency air movements, and if this is the case, the high end rolloff might become noticeable even on FM. There is a question as to whether the mikes or downstream equipment should be improved first; some feel that the noise and distortion elsewhere in the system would "drown out" improvement in mike type or technique. Some favor enclosing the broadcast loop in Dolby-A or dbx encoding, and loudly state that this would make more difference than anything else that could be done. Then there is the question of how much difference the listener outside New England would hear after the rigors of tape duplication.

So think about all this, and be prepared to read more about it in the next issue. We audiophiles, perhaps as opposed to the BAS Treasury, could do more.

-- Mike Riggs/Jim Brinton (Massachusetts)

Parts Source Recommended

Readers interested in building or modifying things may be interested in this really excellent parts source. They maintain a good stock of high-quality parts at reasonable prices and have always given me good service and advice. A catalog is available for the asking from A-G Equipment Company, 5400 Mitchelldale, Suite B-8, Houston, TX 77092.

-- Tom Tyson (Texas)

More Praise for Sound Reproduction

Like Massachusetts BAS member Dennis Boyer, I have found Sound Reproduction, Inc., a good place to do business. They are fast and courteous, and their prices are unbeatable. They have a shop if you wish to avoid the mail, and they issue a useful quarterly catalog with comparative prices. Their address is 460 Central Avenue, East Orange, NJ 07018, phone (201) 673-0600.

-- Peter Cade (New Jersey)

I agree. My Pioneer RT-701 arrived exactly seven days after I mailed my order.

-- Michael Riggs (Massachusetts)

IAR, Audio Forum, Etc.-- Further Thoughts

Recent ads have made it appear as if the BAS were endorsing an audio magazine, International Audio Review, and on two other occasions BAS Speaker originating articles have been reproduced in other organs without credit to The Speaker.

However, in a moment that made me feel supremely foolish, I realized that International Audio Review doesn't even appear in Peter Mitchell's July ranking of audiophile magazines. Of course, IAR doesn't claim otherwise, just that they're "second" based on Peter's figures, plus addition of a set of their own -- which apparently is true enough. But I think the casual reader is left with the impression of an endorsement.

The case of Audio Forum seems a little cleaner -- just a lack of courtesy, which editor/publisher John Bertoglio is rectifying. Roy Cizek offered the article to him (perfectly all right with us), but neither he nor Audio Forum thought to mention the source. Something similar appears to have happened with Dan Shanefield's July. 1976 article, "A New Method for Estimating Phase Distortion and Experiments on Phase Audibility," which recently was reprinted in Wireless World. Dan did ask them to credit us, but they either forgot or simply chose not to do it. We will be writing them about the omission.

Our failure to articulate a reprint policy may be part of the reason for the problems we've been having, so I want to remedy that here. In our view, what you write is yours. You may do with it what you wish, and we will do everything we can to help. Our copyright is to protect contributors and the BAS from misuse of material printed in the Speaker, not to maintain exclusivity. We do, however, require that you make a credit to the Speaker a condition of publication elsewhere and that you notify us in advance. The latter, by the way, is important, because in some cases we prefer that certain specific procedures be followed, so do clear things through us.

-- Michael Riggs (Massachusetts)

News on the RFI Front

To help the new RFI panel and other interested BAS members with their research on RFI legislation, I'd like to offer the following information and analysis.

Three RFI bills were introduced in the first session of the 95th Congress. You can find the sponsors' introductory remarks on Goldwater's Senate bill (S. 864) and the House bills of Messrs. Benjamin (H. R. 8079) and Vanik (H. R. 8496) in back issues of the Congressional Record, dated 3/2/77 (pp. S3315-17), 6/29/77 (p. H 6712), and 7/22/77 (p. H 7680), respectively.

Senator Goldwater thoughtfully inserted the text of his bill and the text of the section of the Communications Act of 1934 that his bill would modify into the Record along with his comments. The Benjamin bill is identical to Goldwater's bill, so its absence is not a problem. The Vanik bill, however, does differ from the other two.

The difference in the Vanik bill is not as important as its sponsor suggests. Unlike the other two bills, which would explicitly authorize the FCC to require the use of "protective components" (filters), the Vanik bill merely allows the FCC to set minimum rejection standards. But nothing in the language of the bill expressly prohibits the FCC from requiring the use of filters as a means of achieving RFI rejection. So even the modified Vanik bill is not as good, as might seem; it would still let the FCC meddle in audio design.

Fortunately, Mr. Zwicker is mistaken in assuming that the enactment of such legislation within the next two years is inevitable. In fact, it is unlikely to pass during the term of the present Congress. The House Subcommittee on Communications is busy preparing a new basic communications law to replace the much-amended 1934 act. Chairman Lionel Van Deerlin has made this project the subcommittee's top priority. And he would prefer to incorporate any changes deemed necessary into the omnibus re-write, instead of considering them as amendments to the existing law (the form in which the three bills above are presented).

So it is the form that this new basic law takes, and not the progress of these three bills, that should concern us. Still, I recommend that you familiarize yourselves with the details of the bills, for they will be considered as possible clauses in the new Communications Act.

At the moment, work on the new act is progressing so slowly that it is unlikely to be finished before the end of 1978. And when it is reintroduced in 1979, it will take at least two years to get through both houses of Congress, since there are likely to be many controversial provisions affecting broadcasting, cable, and common carrier regulation.

And there's more good news. According to the consumer electronics trade papers I read, public interest in CB radio is waning. Several companies heavily involved in CB are in trouble (investors take note). It appears that the broad base of support for RFI legislation is drying up. This leaves only the hard-core CB subculture, the FCC itself, and the amateur radio lobby to support FCC regulation of the audio industry.

But it would be a mistake to underestimate hams. Like the gun lobby, ham groups are quite adept at organizing massive letter-writing campaigns and engaging in grass-roots politics. These techniques give both groups a degree of influence far beyond what one might expect of a relatively small number of people. We audiophiles can give the hams a run for their money in influencing Congress on question of RFI legislation. But the influence of ham groups is so entrenched at the FCC that, if such legislation is passed, we will have little hope of influencing the kinds of regulations the FCC would adopt.

Moreover, the FCC itself has never been very sympathetic to audiophile concerns. One need only look at how they've handled FM to see what I mean. The agency has not upgraded the audio performance requirements for mono FM and TV sound since 1945. When new frequency allocations were being made in that year, the FCC kicked FM out of its prewar band and relocated it in the present 88-108 MHz band. This action, which made all existing FM transmitters and receivers obsolete overnight, was nearly fatal to the then-nascent high-fidelity medium. Fifteen years later, the FCC chose the GE-Zenith FM multiplex instead of the Crosby system, although the latter was considered superior by virtually all component manufacturers. FCC officials defended the debacle by pointing out that the GE-Zenith, system, unlike the Crosby system, allowed for the retention of SCA services. From this we can only surmise that they consider the "needs" of the purveyors of tinny background music more important than those of the critical listener.

Please consider all of these factors before making any endorsement, however qualified, of legislation to give the FCC regulatory authority over the rest of audio technology.

-- Jack Hannold (New Jersey)

Postscript to RFI Panel Motion

In reading the summary of the September meeting, I was surprised that the "Comments: Personal Opinion ..." section was published as it appeared on the motion document. This section was initially intended for the members to whom I mailed a copy of the proposed motion two months prior to the meeting; it was my opinion only and was simply intended to help explain the intent of the motion. A second version of these comments was included with the motion that was brought to the floor, but again it was intended to be my opinion only. The passage of the motion did not in any way endorse these comments.

I feel that publishing my comments gives too much weight to my opinions versus the others expressed at the meeting and those I know reputable individuals in the manufacturing business have. The RFI panel should in no way feel bound by these opinions; they must be fully objective and come to their consensus independently.

I hope the RFI panel will actually come to fruition, and I hope they will seek to modify the September motion in any way they see fit (subject to a vote of the members) to suit their plans.

-- Harry Zwicker (Maryland)

Verion Cables

As part of the audio industry, I tend to be skeptical of many of the alleged audiophile products that appear on the market. My friends and I have often been disappointed. However, I recently stumbled upon a unique product: the manufacturer doesn't promise the world on a silver platter, but merely suggests that his product is an honest attempt to solve the growing audiophile problem of RFI. The manufacturer is Verion; the product is a cable.

I used to experience RFI problems through my tuner, both from CB's and from telephone dialing. It got to the point that I expected to have a conversation, sooner or later, with someone through my speakers. The Verion cables solved all my problems. I now enjoy clear, clean FM reception without an elaborate antenna set-up. The cables are not cheap, but for me they were well worth buying.

Incidentally, although I am an audio distributor, I do not handle Verion products. I am writing this letter as an audiophile who is glad to see this product available. Thank you, Verion.

-- Tom Yankewicz (Manitoba)

Stereo Review Replies to "In the Literature"

Re: "In The Literature," Stereo Review, October 1977 (p. 20, October Speaker). Hirsch's answer to the question of RFI testing is neither a cop-out nor does it contain a logical fallacy. The problem is simply that there are too many external sources and types of RFI, too many ways for it to get into a system, and too many individual variations in installations and random variations from sample to sample of a given product, for anything less than a totally rigorous test of RFI sensitivity to be helpful. The random factors would more than swamp out any differences in a given component's internal anti-RFI design.

To restate the case, I'm sure that if all products were tested under the same highly-controlled conditions in a screen room, some would show up better than others in regard to RFI rejection. However, the complexity and expense of the test would in no way be justified by the usefulness of the information that could be provided to audio consumers using their equipment out in the real world of RFI.

-- Larry Klein (New York)

Correction on Decca Arm Mass

On page 13 of the October 1977 Speaker James Lin quotes Hi-Fi Sound as giving an effective mass of 18 g for the "old" Decca International arm. This is in error. The Decca arm used to be available with two different headshells, a skeleton type (this is the one usually seen) weighing 5 grams and a more massive one weighing 13 grams. Hi-Fi Sound clearly states that the effective mass of the arm is 18 grams with the heavier headshell. This means that with the lighter headshell (which is the one universally used) its effective mass is of the order of 10 g, a very respectable figure. I have no similar information on the "new" Decca International arm.

-- Stanley P. Lipshitz (Ontario)

Scott Kent on Records--IV

An X-Y oscilloscope display of a stereo signal provides considerable information. I have mentioned observing peak levels and the relative balance of in-phase to out-of-phase information before (Speaker, October 1977). On a record, the out-of-phase information is cut in the vertical plane of the disc. The amount of vertical information is limited by playback and side length considerations. Generally, it does not exceed 5 cm/sec, rms, or 7 cm/sec peak. Lateral peak levels are often as high as 20 cm/sec referred to 1 kHz.

As many who have recorded live music are aware, two omnidirectional microphones of reasonable quality, spaced ten to fifteen feet apart, can make a very listenable recording. Distance from the sound source (be it musicians or motorcycles) is not terribly critical to obtaining a sound balance on playback as one might have heard if standing midway between the two microphones. In fact, this is not a bad technique to use for placing two omnis far enough back to hear a pleasant balance of direct sound and hall ambience -- as far apart as most peoples' stereo loudspeakers. Depending on microphone choice, deviations from flat frequency response may be apparent, but loudspeakers are far less flat in real listening rooms than the response curves of most omni microphones. Overall, the results will usually be remarkably realistic compared to what is generally available on records. If the microphones are placed with a much wider spacing than the playback speakers, loss of image and some problems of localization can occur. These problems can be largely corrected with a bridged center-channel loudspeaker. An additional center-placed omni microphone would also resolve the localization, but create a loss of depth in the direct sound. (Steinberg and Snow, AIEE, December 1933)

If it's all that easy, why don't record companies simply tack up two omnis ten feet apart, cut the results, and give us impressive, undiluted, live-sounding recordings? Most record company spokesmen allege that nobody would buy records that sound like real musicians playing in acoustically appropriate surroundings. For a non-concertgoing listener, preconditioned by unreal sounding recorded music, they may be partly correct. Many cynics say it would put too many recording engineers out of work and close down a lot of manufacturers of multi-track hardware. In classical music recording, this may also be partly true.

Another reason, however, is that this "fine" sounding omni-made recording cannot be cut onto a lacquer to give a pressing with twenty-minute plus sides, which can then be played back with a real cartridge with realistic dynamic range. Exceptions are music played on instruments with small dynamic range (such as harpsichord or clavichord) or music calling for a modest dynamic range which is then recorded at, perhaps, an excessive distance. (An example is reviewed this month.) Why the two-omni tape cannot be cut satisfactorily is evident if the signals are viewed X-Y on an oscilloscope. There is as much out-of-phase information as in-phase. Peak levels appear either in or out of phase, depending on how their frequency contents relate to microphone spacing, arrival time, and room standing waves. Without reducing the out-of-phase, vertical information, this sound cannot be feasibly cut.

The simplest method of reducing the vertical information is blending the two channels, cancelling some out-of-phase information. Some preamps have a blend control which, in its extreme position, reduces the sound to monophonic. Instead of the sound becoming larger in the forte passages, it simply becomes louder. Much of the sense of largeness of a crescendo comes from the additional reverberant hall sound. With two spaced omni's, much hall reverberance is out-of-phase information and is cancelled when blended. Blending also cancels, with a random comb-filter effect, sound information throughout the frequency spectrum. Though the losses are most obvious at low frequencies, whose wavelengths are large compared to microphone spacing, the relative confusion overall, because of differing arrival times in the remainder of the spectrum, is apparent when comparing to the unblended original. Often either channel alone will sound better than either the blended stereo signal or a mono total blend.

A more common method of reducing out-of-phase information takes into account that low frequencies require more groove space and present the worst vertical tracking problems for average cartridges. The louder passages in music are generally weighted in the direction of more bass information (with the exception of occasional "unnecessary" cymbal crashes, which can be sharply limited by the cutting amplifier's vertical limiter -- unnecessary because the "average" listener will not know the difference, because he can't hear the original). Passing the bass information through an "elliptical equalizer" (a sophisticated term for a high-pass filter or bass separation control, depending on the cutting system) is claimed to solve all stereo cutting problems without audible ill effects. Cancelling out-of-phase bass solves lots of cutting problems, but "audible ill effects" are judged subjectively, often on monitor speakers with poor imaging in environments with high noise floors. Unfortunately, with a spaced-omni-made tape, this results in cancellation of selected octaves, fourths, and fifths. With most microphone techniques other than "multiple-mono," the blending of bass information produces a bass-shy sound. Even if bass is boosted so that overall frequency balance is subjectively restored, another serious problem remains.

In a large hall, with a two-omni pickup, much of the sense of space and depth is conveyed by low-frequency out-of-phase information. This information will be lost with any method of cancellation during the cutting process. Additionally, there are possible problems on playback of even the tape made with two omni's. If bass information is bridged to feed a common woofer or sub-woofer, the same cancellations occur as those which can be introduced by the cutting engineer. Though discrete bass channels are not absolutely necessary to localize bass information (McCoy, JAES, Vol. 9, 1961), they are necessary to accurately portray low frequency ambience. Localization information is assumed to be midrange sound components of music having low-frequency fundamentals. However, experiments with surround sound techniques by Gerzon (JAES, Vol. 25, 1977) seem to question any practice that alters bass phase relationships or arrival times. On the positive side, playback systems using a common bass unit would not portray the degradation of a bass-blended omni pair.

In short, the non-critical and natural sounding technique of the spaced-omni pair, though useable for recording a tape, is less suitable for cutting a record. Fortunately, there are other types of microphones with differing directional patterns and several techniques of vertical compression and limiting that can help produce a record bearing some resemblance to the original live sound.

L1SZT - Annees de Pelerinage, Lazar Berman, piano, DG 2709076 (3 record set), AVPL - 12, 14 (5)

- Annees de Pelerinage, Gyorgy Cziffra, piano, Connoisseur 2141, 42, 43 (3 separate records), AVPL - 13 (4)

Both these recently released sets of Liszt's Annees have several unfortunate problems. Musically, Berman's rendering bounces from quiet and delicate to loud and violent, without the necessary range of gradation between. Many of the quieter pieces lose all sense of tempo; they suffer from over-Romantic rubato. Unlike his concert performances, these records fail to capture the impact of the violent playing and the enormous volume of sound Berman can elicit from a piano. One also misses his exceptional range from pianissimo through fortissimo. Though the true peak levels involved would certainly intimidate a cutting engineer, judicious microphone technique, limiting, and compression can help create a very convincing illusion of reality (for example, the Liszt Hungarian Rhapsodies, DG 2709044, reviewed last month).

This DG release features harsh, close-miked sound with a small amount of unnatural sounding ambience. The piano's width changes continually, and the piano moves around laterally to such a degree that one wonders how Berman managed to keep up with it. Fortes are shrill and hard-sounding and, despite fairly high cutting levels, do not have the impact of the Hungarian Rhapsodies. All sense of depth and low frequency solidarity disappears in passages forte and louder, creating the illusion of both bass compression and blend. There is very little out-of-phase information, and the scope pattern seems to indicate some inappropriate form of cardioid miking. The pressings are reasonably quiet, although the beginning of side 4 has shatter on the louder chords (3 cm/sec) of the relatively quiet "Gondoliera." Much louder information toward the label is clean. Sides 5 and 6 are warped, creating audible wobble of sustained notes.

Gyorgy Cziffra's performances, on the other hand, are more secure rhythmically and have a continuous ebb and flow of dynamics and expression. These were recorded in Europe for Pathe and mastered by Sterling in New York for Connoisseur. The quality of the records is variable, but they do not have the excessive echo of many Connoisseur piano recordings. The original master tapes of these probably sounded excellent compared to those of the Berman set, but more seems to have been lost either in the tape transfers or the cutting. The microphone technique used for most of the set gives a stable and not overly wide image, good detail, and a natural ambience, not unlike the DG Hungarian Rhapsodies. And forte passages become large as the ambience reinforces the direct sound, not shrill or limited sounding. Out-of-phase information is controlled (with a few exceptions); perhaps ORTF or Blumlein microphone technique was used. There is a problem with wobble and flutter in much of this set, and it does not relate to record warp or off-center pressings. Perhaps Dolby calibration is at fault for the wobble that is triggered by staccato notes modulating the held ones. The flutter may have been a master tape duplication problem. Surface noise of the Connoisseur set is higher than that of the DG, and though similar dynamic range was on the tape, breakup often occurs at AVPL's greater than 12 cm/sec.

Volume 1, CQS 2141, has occasional bursts of hiss, mostly on side 2. Volume 2, CQS 2142, contains no surprises until band 2, side 2, where the microphone technique turns into omni-pair, and the hiss in the left channel jumps up by 12 dB, in the right channel somewhat less. This should have been corrected with a new tape, if necessary. Volume 3, CQS 2143, is visibly off-center and sounds it. Band 4 on side 2 shows either a change in microphone pattern or substantial vertical limiting, as the out-of-phase information becomes minimal (though without serious degradation of the sound). If limiting was used, it should have been continued through the final band, which is not cleanly trackable with an XLM at 1.25 grams and looks overcut under the microscope.

Each of these sets is a mixed bag of technical problems, with the Cziffra set having the edge musically. Perhaps it would be worth saving one's money for a future (?) set of Annees by Arrau on Phillips, hoping they don't compress and limit it, as they did 802906, which includes a few selections from the Annees.

The following two records are examples of ungimmicked, predominantly two-omni recording, with both the problems and the benefits. As well, both records have one exceptionally long side. Side 1 of the Hammar is 28:19; side 2 of the Nonesuch is 27:12. Though both are somewhat distantly recorded, every detail is preserved as if heard from forty feet away. Neither record contains excessive dynamic range or large amounts of low-frequency information.

RUGGLES - "Men and Mountains," MOZART - Symphony No. 35 "Haffner, TELEMANN - Concerto in D, VIVALDI - Piccolo Concerto, New Hampshire Music Festival Orchestra, Thomas Nee, Conductor, Hammar SD 150, $5. 98 from Sonar Records, Box 455, Kingsbridge Station, Bronx, NY 10463, AVPL 7-8 (3)

Though these are professional musicians, this is a summer orchestra, and the lack of hall ambience does not help the somewhat uneven string ensemble, particularly in the Mozart. Overall, the best work is the Ruggles, where the playing is reasonably precise and the tympani fundamentals, and bass information in general, are reproduced with very high accuracy. The other works have sound equally as accurate, but are on a smaller scale dynamically, with AVPL's of 5 cm/sec. The quality of the direct sound is as accurate and detailed as that of the Telarc/Clevelant disc, but there is very little ambient sound, as if recorded in a small, acoustically dead high school auditorium. The Hammar disc was mastered by Sterling and pressed by Windsor, who are known for extremely quiet pressings if given good stampers. The highest levels occur in the Ruggles, which is the first band of side 1. The low cutting levels enforced by the 7-10 cm/sec vertical limit result in low cutting distortion overall. Sonar is also in the pre-recorded tape business, and tapes are available in various formats to check the accuracy of this and other Sonar releases. (7 1/2 ips, ,1/4-track, $20.)

SCHUBERT - Mass No. 5 in A flat, Carleton College Choir, St. Paul Chamber Orchestra, Dennis Davies, Conductor, Nonesuch 71335, AVPL - 9 (3)

This release is the only current listing of Schubert's Mass No. 5 in Schwann and the first stereo version. The performance, befitting a Schubert mass, is relaxed, yet accurate, with moderate dynamics. Engineered by Marc Aubort, the recording appears to have been a two or four-omni proposition, when judged by its sound and the oscilloscope display. Somewhat distant, but capturing the large ambience and adequate reverberation time of the House of Hope Presbyterian Church, the detail of individual voices is retained. Again, vertical cutting limit, side length considerations, and the somewhat distant pickup insure low cutting distortion, particularly necessary in choral works to retain vocal clarity. The cutting was done at Masterdisc (a new firm) by Robert

Ludwig (formerly with Sterling). If this disc is an indication, this firm is one of few who come close to duplicating a master tape on a lacquer. The pressing, however, has a cyclic 60 Hz swish on side 1 and some thumps caused by dimples on side 2. There are a few pops caused by non-fill, but other forms of surface noise are reasonably low and do not intrude on the moderate level of groove information. Despite the pressing defects, this is audiophile quality sound at a budget price.

-- Scott Kent (Massachusetts)

More on Sheffield's Discovered Again

You may recall that I sent in my opinion of the Sheffield direct-recorded disc, Dave Grusin, "Discovered Again. " Well, between my review and Alvin Foster's I wouldn't blame anyone for being confused; I am myself. After reading the two reviews, I got my copy out and played it again. I would guess that Sheffield did use multiple miking for this recording, which I do not think will reproduce as accurate a sound image as a simple crossed pair would. I agree that the imaging could be better. The bass is heavy in one or two places, but I think this is a function of the music, not the recording. I could not fault the record for being "excessively bright and occasionally too hard and piercing;" I don't hear it that way at all. Reversing the phase of one channel does reduce the bass, but it ruins the stereo imaging, too. I could not hear any difference simultaneously reversing the phase of both channels with a switching box I built for that purpose, nor could anyone else tell the difference with any consistency when I tried at an Audio Forum meeting.

I primarily like the record for the program content and the fine performance of the musicians, and I still think the recording is a very good one, technically speaking. My main concern is with playing it too much; I'm afraid of wearing it out.

-- Damon Hill (Georgia)

Two Excellent Records

The best discs from tape masters seem to be just as good as direct-to-disc recordings. Two outstanding examples are the recent "Indirect Disc" releases made by Dick Burwen: Misty, the Petty Trio, Celia Records BL-3, and This is the One, pianist Dick Wellstood, Audiophile Records AP-120. They are available by mail for $15 each (postpaid) from Decibel Records, P.O. Box 631, Lexington, MA 02173. Some of the stores that handle the KLH/Burwen Labs components also sell these records over-the-counter.

The dynamic range of these records is just about as great as that of the Sheffield discs. There is some very faint tape noise between bands, but you have to strain to hear it. By the way, a good test for master tape noise in a disc recording is to listen carefully right after the music ends on the last band; residual hiss that is audible for a few seconds and then disappears is from the tape master. If you turn up the volume, you can detect it on practically all discs.

On Misty the snares, bass drum, and cymbals are very loud at some moments and are quite realistic throughout. The girl's singing voice offers unusual fidelity to the mental image of a nightclub performance, though not to the image of a singer in a household living room. In general, the sound is what you would expect to hear in a smallish public room with a low ceiling -- not surprising in view of the appearance of Dick Burwen's studio (see the April 1976 issue of Audio). The music on Misty is what I would call mild pop-rock. I use it for relaxing, or as background for small parties; it's good, but nothing fantastic.

This is the One offers among the most realistic solo piano reproduction I have heard, again with wide dynamic range. The music is "supper club" 1940s-style jazz. The sound contains a good deal of wooden-cabinet bass and stringy treble, and your tone controls might have to be freely adjusted to get the maximum realism during playback.

Both recordings were made with the Schoeps condenser microphones, highly flexible equalizers, and noise-reducing companders described in Dick Burwen's AES papers. Some of the mastering was done on ordinary 1/4-inch tape. The Burwen 2000 compander employs 3:1 compression with both bass and treble pre-emphasis. Playback to the disc cutter is, of course, done with 1:3 expansion and mirror-image de-emphasis. (By comparison, the dbx II process uses a 2:1 with no pre-emphasis.) Both recordings were made with genuine stereo miking; in some bands some left-plus-right blending of the extreme bass and treble was used to cancel noise and to adjust the stereo perspective. The disc mastering for Misty was done in Germany. This is the One was cut by Bob Ludwig at Masterdisk studio in New York, described in db, September 1977, p. 42. Ludwig actually monitors on Quad electrostatics rather than on the usual studio JBLs or Altecs. All that care really pays off in the quality of the sound.

-- Dan Shanefield (New Jersey)

And a Third

The other day I went out and bought a new record titled simply, "Norman Blake/Tut Taylor/Sam Bush/Butch Robins/Vassar Clements/David Holland/Jethro Burns." Being a guitar player of sorts, I try to learn techniques from records of skilled guitarists, and since Norman Blake is one of my favorites, I bought the album when I spotted his name on it. When I played the first cut, however, I quickly realized I had found something special. Sonically, this record is outstanding. The surface is as quiet as a winter's morning, and the miking is excellent. There are subtle nuances, such as the actual sound of the plectrum (pick) going over the strings which is something I usually hear only when I'm playing the guitar myself, or when listening to someone else. The dimensionality is also excellent, and the group spreads out nicely in my living room.

As for the music, the record has, in my opinion, some of the very best talent to be heard. The style could best be described, I think, as bluegrass-jazz. David Holland provides solid and skillful bass playing for the rest of the group. Vassar Clements is superb on the violin, and Tut Taylor's mandolin is really something to hear. The guitar playing of Norman Blake is nothing short of amazing. He has a way of making incredibly complex pieces sound easy, with the dexterity of a high-speed computer and the accuracy of radar. (I saw him once in a concert, and he looked completely relaxed while playing songs with such speed that it was almost impossible to follow his fingers.)

As I said before, the first cut, "Sweet Georgia Brown" told me I had just found a special record. The next song, which lasts for 7 1/2 minutes, is a delightful group effort called "Sauerkraut 'N Solar Energy." (Just try to refrain from tapping your foot as this song gathers players and momentum. Just try!) Norman's solo, "The Old Brown Case" caused me to glance over at my Yamaha FG-160 flat-top and wish I could play like that. It's an incredible example of virtuoso guitar-playing. Over on the second side, Tut Taylor has his mandolin in fine form for his composition"Oconee," which is a lovely, somehow hypnotic piece (if that's the right word) that is repeated over and over, building on the tune a bit as it's played (somewhat the way Ravel's "Bolero" is).

Anyhow, you get the picture. Sonically, producer Hank Deane and recording engineer Claude J. Hill have given us, I think, a record that is easily comparable to the expensive direct-disc things. (I' m not convinced that direct-disc recording is the only way to perfection.) Musically, the musicians have given us a record that leaves many of the direct-discs in the proverbial dust.

The record number is HDS 701. It is distributed by Flying Fish Records, 3320 North Halsted, Chicago, IL 60657. It is available in record stores; in the Boston area, you can get it at the Coop.

Enjoy. -- Bob Graham (Massachusetts)

I haven't heard this particular record, but I have seen and heard Norman Blake play many times and can testify to his remarkable skill. The only other record I know of that features him (He's been session man on many) is "Home in Sulphur Springs," Rounder 0012, which may or may not still be available. But you'll probably want to find out after you've heard the record Bob recommends, so here's the address: Rounder Records, 96 Winchester, Medford, MA.

-- Michael Riggs (Massachusetts)

Good Gershwin

Gershwin on Broadway, Buffalo Philharmonic/Michael Tilson Thomas, Columbia M 34542

This recording could very well have been another "Symphony Orchestra Turned Pops" record in Grand Old Boston Pops tradition. Fortunately for us latter-day Lehman Engle zealots, this is a genuine Broadway overtures disc, without the horrendous excesses of Boston Pops brass, bass, and percussion (not always in that order, but always excessive). Thomas always has had a flair for show biz, and here he gets an entire album to let loose. He leads us through six merry Gershwin overtures, all arranged in perfect "pit orchestra" style by Don Rose. The playing is precise and well-balanced throughout. The sound is clean and well above Columbia's usual mediocre standard, although the brass are a shade too distant for my likes (the French horns seem to be playing in the corridor of Kleinhans Music Hall). The miking technique is that of a pop orchestra session (i.e., Andre Kostelanetz, et al), rather than typical symphonic miking, providing the listener with a far greater appreciation of this type of music.

If you're like me, you'll recognize only a handful of the tunes from the six musicals represented on this album, but you are sure to enjoy each and every one. You may even find yourself struggling to resist the urge to get up and conduct your speakers.

-- Warren Schroeger (Massachusetts)

Great Organ at Methuen on Columbia

The Great Organ in the Methuen Memorial Music Hall, Methuen, Massachusetts, has recently been recorded by CBS Records for an album of the Camille Saint-Saens Third Symphony, featuring organist Leonard Raver.

Through multi-track recording techniques, the recording of the organ will be electronically merged with a performance by the New York Philharmonic orchestra conducted by Leonard Bernstein as recorded at Manhattan Center in New York City in December of 1976.

The record will be released under the Columbia Masterworks label this winter.

-- Edward J. Sampson (Massachusetts)

(This should make an interesting contrast with the direct-disc recording announced a few months ago. -- MR)

Master Tape Copies of Outstanding Quality

Have you seen this tape advertised in Hi-Fi News (p. 84, August) and wondered how it might sound? Well, very good indeed. I wrote to Cresent Records and received back a two-page letter telling me all. The tape is available in many configurations, the norm seeming to be 15 ips with Dolby A, but 7 1/2 ips is also available, either with NAB or CC1R equalization, less Dolby (Dolby B or dbx not being available at this time), all recorded on Ampex 407 (I believe). I was a little concerned about playing a 1/2-track tape on my 1/4-track Revox, but in their letter they say that as long as the head is aligned properly there should be no problems. This is because the tape is dubbed with a narrow (.75 mm) guard band, which seems to alleviate any imbalance in playback, except for a bass boost. Test tones at the beginning of the tape play back at the same levels in both channels. Operating level and test tones occupy the first three minutes of the tape. The tones are: 700 Hz (reference), 10 kHz (azimuth), 40 Hz, 63 Hz, 125 Hz, 250 Hz, 500 Hz, 1 kHz, 5 kHz, 10 kHz, 14 kHz, and 16 kHz. The operating level is ~= 3 dB above my Dolby B level. All the music is excerpts from Cresent Records, albums being available at the same address, I assume. All selections are recorded using coincident hypercardioid microphones, except one, where they use a pair of figure eights. The tape includes:

Handel: Suite in D Major. Very good "you are there" ambient trumpet and organ sound.

Holcombe: Three Aires. Very open sounding harpsichord with a little annoying pedal treading noise.
Beethoven: Piano Trio in C Minor, Op. 1, No. 3. All three instruments sound very natural.
Handel: Trio Sonata in G Minor, Op. 2, No. 8. Again, fine sound with plenty of ambience.
D'Aquin: Noel X Grand Jeu et Duo. A very pleasant organ piece recorded at Hexham Abbey. Nice, deep organ notes recorded up to operating level.
A Child of God (Traditional arrangement by Salli Terri). Beautiful solo singing by Jane Robbins. A very ambient sound with choir stretched out across in front of you at some distance. Sound (on my system) gets a little blurred with choir at full strength, perhaps too much reflected sound recorded.
The Girl who Broke my Heart (Traditional arrangement by Harrison and Gordon). With guitar and Hardanger style fiddle. Again, very natural sounding instruments. This is the shortest piece at 2.27 minutes.
Over the Cliff (Stewart)/Dever the Dancer (Traditional arrangement by Furey). This and the finale cut are played on the Uilleann Pipes. This cut also has three other instruments on it that I've never heard of before. Interesting, closer, folky sound.
Tomorrow We Part (Furey). This piece is well titled. The pipes are very moving. Do I detect a little tape modulation noise here?

Well, that's it -- not a cheap tape, by any means, but worth every penny to me at £19.50, or about $33.50 at current exchange rates. This includes airmail postage to your listening room. If you have a good reel machine and can afford this tape, 1 don't think you will be disappointed. The sound alone is worth the cost. I should very much like to hear it at 15 ips with Dolby A.

Cresent Records, Avon Recording Services, 29 Belvedere, Bath, BA1 5HR, England. If you write please mention the BAS. -- John M. Tooley (Delaware)

A Musician's View of Digital Recording

I thought BAS members might be interested in another view of digital tape recording, a view from someone who has had direct experience with it. In August 1976, I played five recording sessions with the Santa Fe Opera orchestra, chorus, and soloists. I play first bassoon in the orchestra. The product of those sessions was New World Records' album of Virgil Thompson's "The Mother of Us All." The discs were made from analog master tapes. But the sessions were also recorded by Soundstream, Inc., of Salt Lake City, using their digital recorder which is shown on page 36 of the October issue of db magazine. During all the session breaks, the Soundstream tapes were available for us to hear the playback. What they recorded was the two-channel mix fed from recording engineer Jerry Bruck's console. Jerry and crew also recorded that two-channel mix -- not for the disc, but for other purposes.

After the fourth session, Jerry and Soundstream (Tom Stockham, president, and Dick Warnock, chief engineer) gave an interested group of the performers a demonstration of the difference between analog and digital tape fed from the same source (the two-channel mix). We listened in the hall used for the sessions, a rather dead one suitable for making intelligible the multitudinous words in the opera. The loudspeakers were British IMFs -- the next to largest, I think.

Differences ranged from audible, on soft strings, to dramatic on a snare drum roll. The digital tape appeared to me at all times to have no noise, distortion, veiling, muddiness, or what have you.

Two other indications of the difference the digital tape made: (1) Jerry Bruck indicated that digital tape was clearly what would be used when it becomes commercially feasible, because it is much better than analog; (2) before I discovered that Tom and Dick were doing digital tape stuff, I was continually amazed at the quality of the little speakers they were using for playback. Questions elicited the information that they cost $80 each, and that Tom did not know what kind they were. The sound coming out of them was fantastic, all right, but because of the recorded sound.

The album is New World Records NW 288/289, and it lists for $15.96. It is good, but you should have heard the digital tape. -- Crawford Best (Louisiana)

Using Electronic Crossovers

Why is it that most audiophiles will swap one component for another (at ten times the price), argue for hours over the tiniest subjective improvement in sound, and in general fret about the smallest details that may improve the sound of their systems only a slight amount, and ignore the single item that will make a really large improvement? I am referring to converting to low-level active crossovers ("electronic crossovers"). Little need be said about the large improvements such crossovers make in an audio system (anybody will notice it instantly), they cost very little, and if you know the crossover points in your present system, you simply cannot go wrong by adding them.

A number of additional advantages are also realized. It is not necessary to use a "super amplifier," because two or three low-power ones will do the job better. IM distortion is reduced, because each amplifier covers a small bandwidth. Each driver is driven directly and therefore is much better controlled than when being driven through chokes and capacitors and resistors. There is no electrical interaction between the drivers, and the total price is usually less than what you will end up paying for a huge amplifier or possibly entirely different components. Of course, one has to get into one's speakers, to connect each driver independently, but that is hardly a difficult task even in "sealed" systems.

There are many different manufacturers of low-level active crossovers. In my opinion, the best is DeCoursey Engineering. They use metal-film resistors, glass/epoxy P.C. boards, and completely modular design. All types of slopes are available, from 6 dB/octave to 18 dB/octave at any crossover frequency. You can get just boards, completed systems, or anything in between. If you can put together kits, then Old Colony has 12 dB/octave kits for less than $10/channel, as advertised in The Audio Amateur. All the above work well.

As always, there is great controversy concerning the ideal slopes and crossover points to be used. I have tried all types of different combinations. In general, if you use the same slopes that the manufacturer does, you will not be far off. However, I have found that cutting the woofer off at a lower frequency usually improves the lower midrange clarity. Slopes on the order of 6 dB/octave offer the best phase and power response, but are generally not steep enough to handle most drivers. 12 dB/octave are less perfect, but are used by most manufacturers because they offer steep enough slopes to keep the various drivers out of their non-linear regions. They do not have good phase and power/voltage characteristics. 18 dB/octave filters find their voltage and power again in agreement with each other. Of course, their phase shifts are more severe, because the slopes are steeper. Sonically, I have found the 6 dB/octave slopes to be very good, if you have suitable drivers (very rare). 12 dB/octave crossovers usually sound better because there seems to be less driver overlap, and the sound is just cleaner. However, I cannot detect when a woofer has its phase switched with respect to the tweeter/midrange in the same speaker. Clearly this should be audible. 18 dB/octave crossovers are very clear, and the phase difference between the woofer and tweeter is now quite obvious. I have not used 18 dB/octave crossovers above 500 Hz. Because phase shift is essentially inaudible below 500 Hz, I have no objections to using 18 dB/octave crossovers at such frequencies. It may be that at middle and high frequencies the phase shift introduced by 18 dB/octave crossovers will be objectionable. Of course, no speaker system should have crossovers above 500 Hz, but since most commercial designs do ...

Even if you are convinced that IC's have no place in audio (most electronic crossovers use IC's), you will still be amazed at how much better the sound is. If you cared to do some really critical testing, you would find that the IC's used in electronic crossovers do not change the sound at unity gain in any case. If you care to, you may order "fast" IC's in a unit you buy from DeCoursey.

In summary, you can be virtually assured of improving your sound with low-level active crossovers, and there is essentially no risk involved with trying them.

-- Roger Sanders (California)

Equipment Evaluations

Audio/Pulse Caveats

I finally had an opportunity to audition an Audio/Pulse One, digital delay line. I found that while it does offer an added dimension to recorded sound generally, it is clearly unacceptable to a purist. Specifically, it is just too noisy. On naturally recorded master tapes processed with a dbx 157, the reproduced sound is free of noise on material with 103 dB peaks. The tapes are uncompressed and have dynamic range well in excess of 70 dB. Under these conditions, the Audio/Pulse adds audible hiss to the sound, even when adjusted so that its level is just below that at which its effect can be detected. To add insult to injury, the levels on the Audio/Pulse must be matched to the rest of the sound each time the main system's level controls are adjusted. This is just not satisfactory in a $600 unit. The concept is sound, but the execution must be improved for the unit to be acceptable.
-- Roger Sanders (California)

Several months ago, I reviewed the Audio/Pulse Model One in the Speaker. At that time, I commented negatively about the lack of output signal suppression during power-on/power-off operations and indicated that this caused thumping and transients in my amplifier. Continued experience with the Audio/Pulse has not made this problem any easier to live with, and I wish Audio/Pulse would do something about it. In all other respects, the unit is still marvelous.

A few weeks ago, I had reason to suspect that there was something wrong with the Audio/Pulse. (Further investigations proved this assumption erroneous; the Audio/Pulse works fine.) In the interim, however, I called Audio/Pulse to find out who in Los Angeles could handle warranty repairs. The Audio/Pulse people were very nice, but they said no one could handle repairs but themselves; I would have to ship my unit back to the East Coast. That meant finding boxes, incurring delays, and enduring inconvenience beyond that which is reasonable. I would caution other BAS members, before buying any piece of electronic equipment, to be sure that they can get the unit locally repaired by authorized, competent people, or be prepared for the consequences. (And by the way, rent an extra house to save the cardboard packing boxes.)
-- Gerald H. Larsen (California)

Mr. Larsen may be glad to hear that Audio/Pulse, formerly a subsidiary of Hybrid Systems Corporation, has been sold to Hoffman Electronics and is slated to be moved to California early in 1978. But his letter might usefully serve as a springboard for a general discussion of servicing-related questions in the Speaker. What products are conspicuously well-designed for serviceability, and which are conspicuously bad in this respect? Does it really matter what the warranty policy of a manufacturer is, and if so, which are the best and worst? (Perhaps this has to be related to product category; maybe amplifiers are so reliable that warranties are irrelevant, while for tape recorders -- especially the video machines -- the availability of skilled service people is crucial.) What are the pros and cons of factory service versus local service? Comments on these and other questions are solicited from members.

Incidentally, as the author of the Audio/Pulse warranty, I will note that it covers all shipping costs to and from the factory, including the provision of a new carton and filler if you didn't save the original one. Does such a policy matter to the audiophile?

Some background might be useful, to put these questions in perspective. Local service has always been a major headache of the audio and TV industries. Long delays, poor technician training, and lack of diagnostic skill are traditional scandals, even where no dishonesty is suspected. Furthermore, the high performance levels of many current audio components, together with the complexity of their circuitry, are placing demands on the technical skill of service people (and on the cost of the advanced test equipment required) that even the most dedicated shops can hardly cope with. It is not hard to find a service department capable of replacing the idler in a Dual changer, the power transistors in a Crown amplifier, or the stock Peerless tweeter in an Avid loudspeaker. But how many shops have either the skill or the $4000 test gear required to align a new tuner to its specified 0.08% distortion? And how many service technicians really understand the workings of the IC logic circuitry in a Tandberg tape deck, much less the far more complex circuitry in a time-delay system? The economies of scale also affect servicing policies. If Pioneer sells 80, 000 cassette decks each year, they can afford to train and equip

200 service departments around the country. But many of the most respected audiophile products sell in small quantities, typically under 1,000 per year; in such cases it simply is not economically feasible to train and equip a network of service stations. Do we really expect to find local servicing for Audio Research and DB Systems preamps, Denon cartridges, electrostatic speakers, and Sequerra tuners?

Many audio manufacturers are between these extremes in size and have adopted a service arrangement that is quite practical, if slightly devious: "local" servicing which is really factory service in disguise. Many dealer service departments and independent "authorized" service shops really do not service all of the components brought to them. Instead, for the more complex or sophisticated products they simply serve as a trans-shipment office to send the product back to the factory (or to the importer). This is certainly no quicker than having the audiophile ship the unit directly from home to factory, but it may be more convenient if you cannot easily arrange for UPS pickup and delivery at your home. There is also a financial cost: on out-of-warranty work the local shopwill charge a profit margin in addition to the factory's service fee, while on in-warranty work the factory usually reimburses the local shop for the labor involved in packing and order-processing (as well as for the shipping cost) and builds these added costs into the retail price of the product. Is this approach preferable to direct factory shipment by the audiophile?

Comments are welcome. Incidentally, in case the above is taken as a defense of Audio/Pulse (which it is not intended to be), I will add that my employment at Audio/Pulse ended on April 1, 1977.

-- Peter Mitchell (Massachusetts)

<u>More on the Fons CQ-30</u>

I read with interest Bob Borden's comments on the Fons CQ-30 turntable (<u>Speaker</u>, August 1977). Having used a Fons since February 1977 without incident, and without noticing any of the quality-control problems he describes for his unit, I want to point out that there are at least two versions of the CQ-30 in the marketplace: one with cylindrical black speed-selector buttons and a glued-on rubber platter mat, like Mr. Borden's, the other with speed selection via a four-gang interlocking pushbutton array with different colored square buttons for 33 1/3, 45, 78 rpm, and Stop functions and an unglued felt platter cover. When I bought my CQ-30, both versions were available for inspection, and I selected the latter, because quality control, fit, finish, etc., seemed better, while price and claimed performance were identical.

The salesman assured me that the unit like Mr. Borden's was the newer (and "improved") model. However, there did not appear to be any differences aside from the type of pushbuttons and platter mat used, and the overall finish of the two units. The sample I was shown as the "newer" model, with the features of Mr. Borden's, did not have a strobe pattern engraved or painted onto the platter, but its speed buttons were askew. Mine are not, being parts of an integral four-button unit mounted on the PC.

Using a Decca arm, with a unipivot, I have not had to worry about arm leveling with the Fons, but did encounter quite a bit of trouble trying to get the suspended mounting board located so it did not rub against the inside of the hole cut for it in the baseboard. The tolerance here is just too close. My first sample of the turntable came without a tone arm mounting board altogether, but I suspect this was the dealer's fault. When I asked for another in a sealed box, all parts were there, and assembly went smoothly.

The Fons's base is rather thin, and to use the Decca arm, I had to cut a hole in the bottom panel (which, incidentally, does not sag or fit improperly on my CQ-30). Through the hole, the arm base with connector extends about 3 cm. Setting the table on pieces of the foam in which it came packed both raised it high enough to clear the connector and provided additional damping.

Because there are plenty of good turntables without 78 rpm speed, and with variable speed at 33 and 45 rpm, I think the Fons might be of primary interest to collectors of old records. Though many so-called "78's" were recorded at speeds as low as 72 or as high as 85 rpm, there is a definite shortage of good tables that offer 78 rpm with variable speed. The Thorens and Lenco models come to mind, but Thorens is considerably more costly than the Fons, and the low-priced Lenco L-75, though equipped with a not-bad tone arm, achieves its infinitely-variable speed control via a rim-idler set-up, with the idler operating on a conical drive shaft. This gives rise to rumble problems, of course, which are not a factor with the belt-driven Fons. Too, only the Fons gives its purchaser a choice of arms, if they will fit into the narrow base.

I would be interested to learn how Mr. Borden managed to remove the platter from his CQ-30. So far, I have not been able to do this from topside, and my installation does not allow convenient access to the underside of the unit. I am concerned over how I might replace the drive belt, should this become necessary. So far, however, after seven months of heavy (six to seven hours per day) service, including disco music taping involving lots of slip cueing, torque remains unchanged, and slippage does not seem to be increasing. Overall, the CQ-30 seems to be one of the nicest pieces of hi-fi equipment I have owned, and one of the most reasonably priced.

-- George Androvette (Massachusetts)

<u>Nakamichi/ADS Car Stereo Saga</u>

Cassette reproduction in an automobile is a sometime thing of wonder. In my previous automobile (a 1975 Porsche 911), I focused my attention on an in-dash unit that would combine cassette capabilities with AM/FM stereo reproduction. I tried several different units, including those made by Blaupunkt, Craig, Pioneer, etc., only to be intensely disappointed with the results. Either the radio portion of the unit was so poor that you were better off without it, or the cassette mechanism suffered from such severe wow and flutter that the resulting sound was just plain unpleasant. I finally settled on a Panasonic CQ724 unit with a Craig power booster and Craig Power-Play speakers. This combination gave reasonable performance, and the unit had built-in automatic reverse capability. All in all, it seemed the best of a very poor lot.

A few months ago, when I got a new Porsche 911, I decided to go for the big gun -- the Nakamichi 250/ADS 2002 combination. I made this decision based on several different analyses:

(1) I saw no other unit that was "in-dash" and that combined a radio of decent quality with a cassette handler of decent quality and had a Dolby capability. Dolby makes a substantial difference in sound quality, even in a car.

(2) I have been extremely pleased with my Nakamichi 700 cassette unit and with Nakamichi's enthusiasm and responsiveness as a company. In general, I have found the Nakamichi 700 to be superb.

(3) I was pleased with a pair of ADS 200 speakers being used for remote sound in my home system and felt that the added built-in amplifier available in the ADS 2002 unit would eliminate the need for a separate power booster. In a Porsche, every ounce of space counts.

It's been a couple of months now, and I can report to you that the added cost involved in the Nakamichi/ADS combination was well worth it. The unit is incredibly good. The sound quality is excellent, with virtually no audible wow or flutter. In short, if you really appreciate good sound in an automobile, this is the unit to get. The following are several assorted comments I hope will be useful to other BAS members if they decide to take a Nakamichi/ADS (or similar) trip:

(1) Space requirements dictated that I mount the Nakamichi cassette unit on the floor rather than under the dash. This has proved workable, but unfortunately the original installer fastened the unit so rigidly to the floor of the car that it picks up vibration. I am now in the process of remounting the unit in a manner which will provide isolation from the vibrations common to the Porsche and other sports cars.

(2) The Nakamichi 250 simply cannot properly play signals recorded at the levels easily handled by the Nakamichi 700. For those of you with *very* expensive home units, it's important that you back off on the recording level, or you'll drive a unit like the Nakamichi 250 into distortion. When I make cassettes for use in the car, I decrease the recording level and use the limiter switch. Too big a decrease produces too much noise, even with the Dolby. There is a magic recording level at which you can get maximum signal to noise ratio from your home recorder which will be acceptable to the Nakamichi. Therefore, if you test a Nakamichi 250 and get distortion, it may well be that you've recorded at too high a level.

(3) Stay away from traditional car stereo installers, and deal only with a good high-end stereo equipment dealer. Most important, buy a unit only from a dealer who has had extensive experience in installing the Nakamichi/ADS combination and who has respect for high quality equipment. The initial installation of my unit was arranged by my automobile dealer and was performed by an idiot at a car stereo emporium (they all appear to be idiots) who would have had difficulty learning to shovel snow. Be very wary of the typical automobile stereo shop. Most people in that business have little knowledge, and the most complex tool they seem capable of manipulating is electrical tape.

Further, when you go to get a unit installed, be sure that you are very picky and fussy and exact concerning the kind of cabling the dealer will use, how and where this cabling will be routed, and the precise manner and mechanism in which holes will be drilled in your automobile (unless, of course, you are having a unit installed in a beat-up '61 Chevy).

(4) In some respects, the great Nakamichi/ADS team seems to have had some small difficulty getting their act together. They have made several different revisions to the way in which the cassette unit is cabled to the speakers, resulting in a variety of different cables and sockets, none of which are compatible with the others. Be sure your dealer has fresh stock and sells you only the latest model. The new cables are all prefabricated and have built-in molded connectors on either end. If the wiring is to go through holes in the automobile, this will require fair-size holes. Under no circumstances should the installer attempt to force the connectors through undersize holes. My stupid installer did, and I am now in the process of replacing cables that have been shorted because of mishandling. The cable set costs $36 and reinstalling cables is not an easy task, so be careful.

(5) Finally, be sure the dealer from whom you purchase your unit is capable of repairing it. Because the Nakamichi/ADS combination is built for automobile use, the dealer will need special power supply equipment to work on the unit in his shop. A good dealer should also be able to check the adjustments of the playback heads on the Nakamichi 250 using your own cassette tape to be sure that you are getting the best signals possible and that the head has been properly set by the factory. I noticed that a short article in the September 1977 Speaker commented about a Nakamichi 350 whose heads were misaligned. Nakamichi has told me this is possible, so it's best to have the unit checked.

Despite those warnings, the Nakamichi/ADS combination is an outstanding unit -- you'll love it.

-- Gerald H. Larsen (California)

Paragon 12 vs. Trevor Lees Preamp

I would like to briefly express to you my experience with the Trevor Lees and the Paragon 12 preamps. I lived with both units for eight months, and it was quite an enlightening experience.

It all started one year ago when Trevor was visiting the home of BAS member Dow Williams. After seeing the Paragon, he exclaimed that he could modify a Dynaco PAS-3X to sound better. I challenged him by presenting a brand new PAS-3X a week later. About two months later I received my Lees-modified PAS-3X and was able to put it to test against the Paragon 12. My reaction after a short time was that the Paragon was significantly superior in every way (especially definition, imaging, and high-end response). The gain of the Trevor preamp at the 4 o'clock position was equal to the Paragon's at the 12 o'clock position. The clicks and popping transients were the same on both units when turning the mode selector.

My system at that time consisted of Dahlquist DO 10A's (mirror-imaged, Mylar caps), Son of Ampzilla, Technics 110A with Formula 4 arm, AKG E8PS, Denon 103S, Levenson JC-1.

I therefore put the Trevor Lees on the shelf and returned joyfully to my Paragon. I invited Trevor to Monterey to listen, but he had to return to Australia.

Periodically, I went back and forth between my two preamps. I took my Trevor over to Dow Williams' home and A-Bed it with his (Trevor unit). There were no significant differences between the two, except that mine may have had a slightly better defined low end.

A few months later, I added a Janis W-1 subwoofer (you have not heard Dahlquists until you have heard them with a good subwoofer) driven by a Son of Ampzilla, and put an Ampzilla II on the top. I used the DeCoursey crossover, and it is here that the story becomes fascinating.

There has been much talk about the importance of system compatibility, and it could not be more apparent than with the Trevor unit. Running the Trevor Lees into the DeCoursey crossover changed its sound entirely. The Paragon's sound was not affected.

Transients through the Trevor unit became fast and clean; it made the Paragon's transients seem blurred. The low end was terrific, every bit as good as the Paragon's, maybe even a bit tighter with slightly better sock. The high end, though, was not so good. It seemed brighter -- even a bit harsh -- compared to the Paragon's. The gain of the Trevor unit was now equal to the gain of the Paragon. Definition, coherence, and inner detail were about equal.

After the initial shock, I settled in with the two units, and after further listening, it became apparent that the Paragon was much more listenable over an extended period and was more accurate (natural) in reproducing instrumental timbre. The word "liquid" literally applied to the sound of the Paragon on female voices in the A-B comparison -- a most pleasurable experience. The Trevor unit, by comparison, sounded hard and analytical.

The Trevor unit is a very good attempt to give good sound at a low price, and system compatibility may be a factor in my conclusions. Many may prefer the spectacular transient response and the excellent low end of the Trevor Lees, but I prefer the Paragon 12 for its non-fatiguing, fluid (unstrained) sound. It images well. Voices on massed choral works maintain their definition and coherency.

I have listened to other preamps, such as the Rappaport, but once you live with the three-dimensionality, depth, and air (not neglecting coherence, definition, imaging, and inner detail) that the Paragon recreates from the disc, other preamps seem bland. The Paragon is an exciting preamp -- a totally musical experience on a good system. Maybe not totally true to a frequency response graph, but musical and listenable.

Incidentally, after listening to various components in various systems over the last eighteen months, I have come to the very strong conclusion that system compatibility and component variability are in themselves major factors in evaluating components, not to mention the listener's ear and what he is listening for. -- Eugene Constant (California)

(The more I learn about audio, the more I become convinced that "other things" are almost never "equal," so really meaningful and objective comparisons between units are virtually impossible. This problem applies to all components, but to speakers and phono cartridges in spades. -- DFT)

(What we seem to have here is a case of mismatched impedances. Apparently the output impedance of the Trevor Lees preamp (which has no high-level stage) is too high to drive Son of Ampzilla's 75 kOhm input impedance without rolloffs at the frequency extremes (causing mushiness and loss of detail) and reduced output. When a DeCoursey crossover, with its 100 kOhm input impedance, is inserted, the problem disappears. -- MR)

Acoustat Speaker: Brief Review

The dream of many designers has been to develop a reliable, sonically coherent full-range electrostatic loudspeaker. The Acoustat is Florida-based engineer James Strickland's attempt to better others in this endeavor. The speaker uses three separate but full-range electrostatic panels per speaker (no crossover), with the diaphragms driven directly from the plates of the 6HB5 output tubes. The electrostatic field runs at 5000 Volts, but the panels have none of the usual arcing problems and seem immune to climate, humidity, and dust.

Each speaker has 8.5 square feet of radiating surface. The Mylar diaphragms are . 0065 inch thick. The frames are high-grade heavy plastic; the grids are mil-spec wire. The integral servo-loop amplifiers are of hybrid design, with solid state circuitry in low-level stages and high-voltage G. E. output tubes.

I have used a set of these speakers in my home for twelve months. There have been no service nor breakdown problems, except one power transformer failure, quickly replaced by the manufacturer.

The Acoustats are very room sensitive, as befits their bipolar nature. The perforated fiberboard backs should be left off for best sound. There is an equalization network in the input circuits of the integral amps which compensates for standing waves and nulls when the speaker is placed exactly two feet from a reflective wall, a position suggested by the manufacturer. This network can easily be defeated, however, when using the speaker further out in the room in a free-standing position.

There is definite high-frequency beaming (which is less than with the Quad ESL or a single pair of KLH-9's). This is not so noticeable at a distance of twelve to fifteen feet. The amplifier's high-frequency balance control allows a ± 5 dB adjustment, which is useful in some rooms. There is also an input level control on the amp's back panel. In my listening room (approximately 13 x 28 x 10 feet), an SPL of 96 dB can be obtained ten feet from the speakers. Inasmuch as the room is somewhat overdamped, I suspect one could obtain 98-100 dB in more reverberant rooms.

The overall sound of these speakers is typically electrostatic: quick, precise, articulate, and defined, with fantastic transient response. The overtone structure is superb. The midrange is neutral, neither forward nor withdrawn, while the top end is smooth, extended, rather spacious, and somewhat "romantic." In my room, the bass extends to 30 Hz easily. There is a rather large hump at 36 Hz and a smaller one at 60 Hz. (A similar low end rise has been measured in the same room with other systems, suggesting that there may be a large room component in the lower octave.) With suitable placement, stereo imaging is excellent, and there is adequate depth with good recordings. There is a noticeable vertical beaming effect, so that the high frequencies are markedly diminished when one listens while standing.

The Acoustats easily outpoint a mirror-imaged pair of Dahlquist DQ-10A's with subwoofer and are better in all respects than single Quads or Magnepan MG-2's. I prefer them to the Fulton J or double KLH-9's.

In summary, then, the Acoustat X is a reliable full-range electrostatic loudspeaker, requiring no additional amplifier. It is minimally colored, having only some low-end aberrations, but room-sensitive, requiring careful positioning of both speaker and listener. It can, and should be, compared with the very best loudspeakers extant, using the best preamplifier and program source available. -- Charles W. Phillips (North Carolina)

<u>Horizontal Positioning Recommended for ADS 810</u>

I listened to four ADS 810 loudspeakers in a quadraphonic and Audio/Pulse system for over eighteen months. They have many strengths, including extreme accuracy above about 200 Hz, excellent transparency and clarity in the midrange and highs, excellent power-handling characteristics, fairly high efficiency (75 Watts will drive them very well), and excellent dispersion. But I gradually became disenchanted with these speakers, however, because of their mid-bass boominess.

Loren Bishop of Sound Ideas, Albuquerque, an expert on speaker design and construction, performed a series of tests using a locally-designed real-time spectrum analyzer with LED display. During these tests in different rooms and positions, it became clear that these speakers manifest the mid-bass boominess primarily in the vertical position, whether on the floor, on the recommended ADS stands, or two to three feet above the floor.

A remarkable change occurs, however, when the 810s are turned to the horizontal position with the tweeter always above the midrange dome and the speaker placed twenty to thirty inches above the floor on a small pedestal. The mid-bass boom is eliminated, with the frequency readout on the analyzer flattening out beautifully.

Listening comparisons of the 810 with the ADS 910 revealed some interesting differences. The 910 puts out deeper bass but I would characterize it as less well defined than the bass in the

810. The 910, with apparently the same midrange and tweeter, seems to lack the unique quickness, remarkable transparency, and definition of the 810 in the midrange and highs, although the 910 is certainly a very smooth and basically flat speaker system. I assume that this difference, if it is found in other samples of the 910, results from the different crossovers or cabinet designs. I have not heard the 910 driven by the new ADS electronics or triamplified.

-- Britton Ruebush (New Mexico)

Un-fan Mail for Fried

Irving M. Fried writes quite well on audio matters in his newsletter and elsewhere. He is also an entertaining hi-fi pioneer. However, these talents haven't helped make Fried speakers competitive with others on the market. I have listened to most of his speakers, have heard the H at two separate dealers, and had an opportunity at one of them to compare directly his speakers against the IMF line.

Indeed if he has had a hand in designing, not merely marketing the IMF's, it is hard to understand how he could have so regressed in speaker design. Surprisingly, the worst of his present line is the H. The speaker honks unbearably, and it did so at both places I've heard it. A $200 Cizek speaker, not to mention a $330 Paradox TA-12, is just vastly less colored.

The rest of his speakers at least do not honk; the M seems to be the best. They are nowhere as open as the comparable IMF units, however, and their bass is not as tight. The fact that Frieds are somewhat less expensive than IMF's does not make them a good buy when there are the likes of Paradoxes and Cizeks on the market.

The only shortcoming of the IMF's I've been able to find, aside from price, is imaging, especially front-to-back, which is not up with the best (which, in my experience, is the Acoustat X). The bass of the big IMF's is truly an achievement, significantly bettered only by the Infinity QLS, which overall is a less neutral speaker (a string of midrange drivers sounds like a string).

-- Vytenis Babrauskas (Maryland)

Variations on a Theme by Holt

I've seen the light. I only hope I can undo the damage already done by the gurus of grain at the altars of the air. The breakthrough came while changing the strings on my guitar. I noticed a marked contrast in the sound of each string as I replaced it with a new but otherwise identical one. The new strings had an "alive" quality lacking in their well-worn counterparts. The notes projected better, sounded less muddy, and were fuller and more pleasing to the ear.

Aha, how similar these changes to the improvements I noted in an earlier missive comparing the Rogers LS3/5a to the Fried B (September 1977 Speaker). In fact, my bumbling attempt to decipher the make and model of a recorded guitar (on the Sheffield album by Dave Grusin) was based on just such differences. Listening once, briefly, on equipment all of which was new and strange to my ears, I foolishly attempted to distinguish sounds more variable in real life than were the speakers being auditioned. To complete the picture, I might add that I scraped up the money and bought my own copy of the record, and the guitar in question (on side 1, band 3) sounds like it might be a twelve-string on my own system.

The point, then, is that live music is an infinitely variable thing. Isaac Stern's violin sounds completely different on a rainy night than on a dry one. Wooden instruments change dimensions in humid weather, and strings sound different when brought into a hot studio from a cold car than when they have been setting onstage for four hours. Little imperceptibles like the exact state of tune of the piano, the relative humidity, the size of the audience, and even the digestion of the soloist, all influence the overall sound quality of a performance. A new brand of string or reed can change the character of an instrument markedly. And the melding of all these factors, along with the state of "harmony" among the different artists making up a performing group, can result in a totally different sound every time they play together.

To explore this idea some more, I recorded a piece of guitar music twice, using the same instrument but different strings for each of the two recordings. Knowledgeable audiophiles will listen to these two recordings under the guise of "before" and "after" changes in some system parameter, and hypothesize widely varying explanations for the differences in sound. Some explanations offered included alterations in tone control setting, replacement of all the tubes in the preamp, change in the vertical tracking angle of the cartridge, and changes in load on the phono input.

Carrying this one step further, I discovered that recorded performances at which the musicians were having a good time are often perceived as being of superior technical quality to somber studio sessions. I can offer several explanations for this last phenomenon, but two seem particularly appropriate. First, the perception of transients is "enhanced" by improved accuracy in the musicians' timing. In other words, if a group swings better together, the simultaneous production of individual notes at a given beat actually is better production of transients. The dynamic range of the passage is also widened considerably by the instantaneous summation of multiple note attacks and the subsequent decay patterns. Secondly, the more you swing, the louder you tend to play. I'm sure there are measurable differences in average volume of dull and exciting performances. Even with peak limiting and other electronic manipulation, these differences in volume of sound reproduction are often perceived as differences in quality of reproduction.

What it all boils down to is this: variability in the sound quality of live music is far greater than the variability among components designed for the reproduction of live music. I would be willing to wager a small sum on the fact that even those reviewers who are familiar with live music through inactive participation at live performances, as part of the audience, would readily be fooled by my "same system/different strings" test. Only through intimate familiarity with the performing characteristics of musical instruments under different environmental and emotional conditions can we come to realize that string gauge and heartburn affect what we hear far more than do "air" and "grain."

Please don't get me wrong. I am emphatically not denying that such characteristics exist, nor do I deny that fact that some people can hear very subtle differences relating to such qualities. I am simply saying that it doesn't make any difference to me whether my top end has the grain of Kodachrome 25 or of high-speed Ektachrome, because I don't know whether the guitar player was using heavy gauge (mellow) or light gauge (brighter) strings. It really matters much more that I like the sound of heavier strings under more tension than that I like the high frequency reproduction of a Bryston better than that of a Crown. In fact, I notice that my tastes in musical instruments run parallel to my tastes in amplifiers and speakers, in that the same characteristics that make a D-28 Martin sound better to me than a J-200 Gibson made the Rogers speakers sound better than the Frieds.

In surveying my friends, I notice the same kind of thinking playing a strong part in their selection of system components. The electric guitarists even have two camps. The Fender players own JBL speakers and brighter solid-state electronics, while the hollow-body people tend toward KLH, Advent, and the like. I don't know any reed or French horn players with more than twenty-five Watts per channel, while no drummer in my acquaintance has less than fifty Watts per side. A customer at the shop with which I do most of my retail component business likes two things in the whole world most of all: harpsichords and his JBL 100's.

Another part of the problem is a phenomenon not, to the best of my knowledge, heretofore discussed in this context. Intermodulation is an integral part of the sound of music. Each pair of fundamentals and, to a lesser degree, their harmonics, create additional tones through intermodulation. This is not distortion, but the natural interplay of tones. The contribution to orchestral sound from intermodulation must be significant. When recording an orchestral performance, then, the microphones are capturing both the instrumental outputs and the resultant intermodulation. A two-mike setup will capture much more of this "non-orchestral" output than will a multi-miked setup with isolation between instruments. Upon reproducing this recording, the intermodulation is recreated and added to that already on record. So it is literally impossible, with routine recording methods and equipment, to eliminate this problem. I suspect that the "richness" described by reviewers in many components is in actuality a result of this expanded sound content. In addition, the "spaciousness" and, yes, even the "air" in certain equipment may be a result of the same phenomenon. Improved electronic capability and better electromechanical transducers may be bringing us more of this kind of sound, with enchanted audiophiles describing as ambience what is, in actuality, more sound.

I suspect that, through rational contemplation, we may uncover more phenomena subliminally influencing our judgment of sonic accuracy. It would please me greatly if, rather than offending owners of expensive equipment, I have created an inquisitive horde of music-loving technofreaks.

-- David Reiter (Pennsylvania)

Another Approach to Damping the Black Widow

For those BAS members who own Infinity Black Widow tone arms, I have a suggestion or two that will improve its performance -- at least with my setup (Kenwood KD500, Sonus Blue cartridge), I found a rather dramatic difference.

Those who have purchased this arm know that it is not supplied with any instructions for installation. Aside from the obvious factors of correct overhang distance and offset angle, one should be careful not to tighten too tightly the four small Philips screws that hold the base to the turntable. In fact, they should be loose enough that if you grip the arm base firmly, you should be able to wiggle it rather easily. In other words, there should be a little play. (Why? -- Ed.)

My second suggestion I've borrowed from The Audio Critic, with a slight variation. They suggest using a plumbing product called Duxseal to damp resonances, placing a small dab on the headshell between the cartridge mounting screws and a small amount around the bearing housing. I didn't have any Duxseal handy, but I did have some plumber's putty, so I tried that. I put a small dab on the headshell, but I also applied a small amount around the circumference of the tone arm itself at its proximal end, where the black carbon-fiber rod inserts into the steel part. (Be careful not to let any of the putty rub against the cueing lift pad.) Also, of course, one will have to readjust the tracking force and the antiskate adjustment.

These few tips have resulted in very audible improvement, particularly in the bass, which is tighter and better focused. I really haven't had all that much time to listen as yet for the more subtle changes (I just received the last issue of The Audio Critic a few days ago), but the improvement was dramatic enough that I felt I would like to share this with you.

-- Jerome Tanous (Iowa)

In the Literature

Audio, January 1978

*Audio ETC: Canby muses on compression and expansion processes (p. 8).
*Letters: Errors and modifications in the Leach preamp, and a disavowal of any BAS endorsements (p. 16).
*Behind the Scenes: Bert Whyte surveys the latest from Revox and Technics (p. 20).
*FM Antennas: A good basic introduction to roof antennas, with some valuable performance measurements. The Wineguard 6060 looks like a best-buy in medium-directivity designs (p. 30).
*Demonstration Records: Dan Shanefield's list of current discs with super sonics (p. 40).
*Time Delay Spectrometry: Discussing a technique for measuring the spectrum of the direct sound and that of the room-reflected sounds caused by a loudspeaker, concluding that in PA systems the best performance with least feedback is obtained by mounting a time-aligned speaker flush in an acoustically absorbing wall (p. 46).
*Audio Phase Detector: A construction project, mainly useful in wiring bi-amped and tri-amped systems (p. 54).
*The Maestro: Bert Whyte's memories of Stokowski (p. 58).
*Equipment Profiles: Enthusiastic reviews of the Onkyo A-7 amp and Burwen 1201A dynamic noise filter, plus a review of the Neumann KM-83, KM-84, and KM-85 condenser mikes. Figure 5 on p. 82 shows why the violin sound in BSO broadcasts is zippy (pp. 68-82).
*Tape & Turntable: Reviews of assorted direct-disc, PCM, and other super-fi recordings (p. 97).
*Note the Marantz ad on the inside back cover: Marantz jumps onto the T1M bandwagon, complete with Curl/Otala test scope photos.

The Audio Amateur, 1977 No. 4

*From the publisher: Announcing an increase in subscription price to $12; even so, TAA remains one of the best buys in audio (p. 2).
*The Decline and Fall of FM: Sad stories by a broadcast engineer (p. 3).
*The Audio Research/Dyna Stereo 70: How to convert your rusty old Stereo 70 to an Audio Research cream puff, by improving supply regulation and making the circuit a balanced push-pull design throughout (p. 4).
*A precision attenuator with 100 dB range (p. 12).
*Super Buff: Walt Morrey of DB Systems designs a high-performance impedance-matching buffer for under $5; adding output transistors makes it a headphone amp (p. 18).
*Slewing Induced Distortion, Part 4: Walt Jung has been making a persuasive case for the importance of SID, and in this part he describes the set-up for his A/B test showing correlations between audible distortion and slew rate. But there is a disturbing fly in the ointment: if SID really is a problem, why did he have to use a special test jig to raise the audio signals 6 to 20 dB above normal in order to make the slewing distortion audible? Does this mean that slewing distortion really is not audible at normal levels ? (p. 18).
*Kit Report: A mostly favorable review of the Heath 5248 IM distortion analyzer (p. 38).
*Audio Aids: Including Bob Sellman's method for making Technics direct-drive tables run at 78 rpm (p. 44).

Audio Times, December 1, 1977

*News Items: A fifth technique for AM stereo has been added to the proposals being considered by the FCC. Prices of Japanese audio components will rise another 5 to 10 percent in the coming months, because of the dollar/yen exchange rate. The new owners of Bozak plan more aggressive marketing. BES (Bertagni), which had gone bankrupt, has been revived by new investors. Fuji is planning a high-bias cassette tape with a new oxide, and may discontinue making CrO2. Ampex, Memorex, and 3M are increasing blank tape prices.

Gramophone (England), November 1977

*Shall We Damp? A commentary on tone-arm damping, with reviews of three dampers: the SME FD200 paddle-and-fluid accessory, the Disctraker, and the similar Audiomaster Viscotrack, concluding that all three effectively suppress the subsonic resonance. There is no mention of the fact that the SME appears to be an imitation of the Bob Graham paddle damper (p. 930).

*Modern Record Reproduction: Basic information on the groove/stylus interface (p. 938).
*Hi-Fi in London: Reporting on new products seen at recent hi-fi shows, notably the new SME 3009/II1 arm with damped titanium shaft (p. 947).
*Equipment Reviews: Tandberg 2055 receiver (excellent), Empire 2000 and 2000E cartridges (okay at the price) (p. 959).

Hi-Fi News and Record Review (England), November 1977

*Show Report: On new products shown at hi-fi shows in England lately. Sample excerpt: "The Phase Linear speaker system looked extremely expensive and sounded quite unpleasant." (p. 102). Also reports from shows in Berlin and Milan.
*Subjective Sounds: On using stacked pairs of Quads, plus an enthusiastic note on the ADC ZLM (p. 111).
*Slewing Induced Distortion: Walt Jung summarizes his story on SID. In the process, he says more clearly than he has elsewhere that the 301A op amp has much less SID in the inverting than in the non-inverting mode (p. 115).
*Magnificent Illusions: "The way in which hi-fi has developed indicates a nearly total neglect of some of the rudiments of auditory perception. " A stimulating survey (by a psycho-physicist) of auditory realities, illusions, and the analogies between sonic perception and color vision (p. 127).
*FM Radio: Angus McKenzie reports that his early enthusiasm for BBC matrix H quad is waning rapidly (p. 143).
*Five Amplifiers: A comparative assessment of medium-cost integrated amps, judged to rank as follows: Pioneer 6500/11 (best), A&R A60, Leak 3900A, Setton 1100, Cambridge P80 (p. 195).

Hi-Fi Stereophonie (Germany), October 1977

*Piano-Players Contra Chopin: Not all are kind to him.
*Contemplation Over Chopin's Letters, Work, and Interpretation.
*Beethoven, 3 Times All 9: Three complete recordings compared: Karajan/Berlin Philharmonic on DG SKL 101/108 of 1962 and on DG 2563 795/802, and Solti/Chicago Symphony on London CSP-9.
*Test Reports: The Canton Gamma 800 receiver (unconventional design); the Dual CV 1600, Marantz 3200/140, Onkyo A-7, Sansui AU-3900, Setton AS 3300,, and Yamaha CA-610 integrated amplifiers (all good, none exceptional); two preamp/equalizer combos: Spectro Acoustic Models 217/210 and Dynaco PAT-5/SE-10 (best results obtained when combining the PAT-5 with the 210); Audio/Pulse One digital time-delay system (outstanding versatility, the review is an enthusiastic rave); Burwen 1201A dynamic noise filter (noise modulation and filtered-out highs apparent); KM-System is two small loudspeakers and an ambience processor (interesting, fine results); the Avid 101 loudspeaker (neutral, transparent, full bass, capable of high output levels, flat down to 40 Hz).

Hi-Fi Stereophonie (Germany), November 1977

*Music in Broadcasting: Several articles on program philosophy, contemporary music, and economic aspects.
*Randy Newman: A portrait and biography.
*Report from the German CES, Internationale Funkausstellung, Berlin 1977: Highlights were Pioneer rack-mount system, the Revox B790 direct-drive turntable with a tangential tone arm, Sennheiser electret open system headphones, Koss electrostatic loudspeakers, Wega (owned by Sony) Acoustic Dimension Compiler adc-2, Arcus 2x160W and Sony PWM power amplifiers, and Fisher MT 6225 linear motor direct-drive turntable.
*Test Reports: Three turntables compared: the Braun PS 550, the Marantz 6200, and Thorens TD 126 Mk. II (all good, the 6200's tone arm geometry unbelievably misaligned); the Dual CV 1400 integrated amplifier and CT 1640 digital AM/FM tuner (both fair); the Sansui 5050 receiver (FM alignment mediocre); the Yamaha CR-820 receiver (mediocre); the Philips N 2521 cassette tape deck (rather disappointing); the KLH Models 317 and 345 Little Baron two-way speakers (relatively better than Baron).
*100 Years of Sound Recording: First part devoted to the development of the tape recorder.

Popular Electronics, January 1978

*Note the ad on p. 2 for a bucket-brigade time-delay kit from Southwest Technical Products for only $250.
*Stereo Scene: Some pros and cons of tape mastering versus direct-disc recording (p. 16).
*Julian Hirsch Audio Reports: On why loudness compensation controls never sound right (p. 24). Incidentally, Holman and Kampmann did a more thorough analysis of the same problem at the recent AES convention; a preprint is available, including a description of a loudness compensation circuit that really works, subjectively.
*Reviews: Yamaha 2020 receiver (amplifier excellent, with especially good loudness compensation; FM tuner good, but evidently a little misaligned); Optonica 3535 cassette deck (good); dbx 128 dynamic processor (superb performance, formidable to master) (p. 26).
*The Spectrum Analyzer in Hi-Fi Measurements: Hirsch describes the many excellent uses of the HP 3580A analyzer (p. 49).
*FM Tuner Directory: A compilation of tuner specs (p. 54).
*How FM Tuners Work, Part 2: On demodulation and stereo decoding circuits (p. 58).

Radio Electronics, January 1978

*Editorial: Announcing the Supreme Court decision that we all have the right to hook up our own extension phones, phone answering machines, etc. , as long as they are FCC approved (Ma Bell can no longer claim exclusive rights or require that special interfacing modules be rented). However, Bell can reserve the right to install all wall jacks (p. 14).
*Frequency Display: Construction project for a digital frequency display to add to ordinary FM tuners (you have to mount a sensing module on the front end of the tuner to pick up its local oscillator stray radiation) (p. 21).
*Pink-Noise Generator: Discussion of the pink-noise generator from National Semiconductor's Audio Handbook, now available as a kit for $14 or fully assembled for $20 (p. 43).
*Test Phono Cartridges: On the use of CBS test records and the Shure TTR-103, plus a meter or scope, to evaluate cartridges (p. 51).
*Test Reports: Enthusiastic reviews of the Dual 939 bi-directional cassette deck and the Sansui 717 integrated amplifier (p. 58).

The Stereophile, Vol. 4, No. 1

*Who's Right? A commentary on why audiophile magazines produce discordant conclusions (p. 1).
*Definitive Preamplifier Testing: Comparative preamp reviews, based especially on bypass A/B tests employing inverse RIAA networks to play tapes through the phono input. Preamps include the Analogue 520 (good, but others are better); Bravura (excellent when used with Shure V15/II1G in a Grace 707, otherwise doubtful); Audio Research SP-4 (early and recent samples unsatisfactory in different ways); DB Systems Model 1 (very neutral and detailed, but ...); Advent 300 preamp-receiver (not perfect, but a great bargain); Audionics BT-2 (may be the best solid-state preamp); Dyna PAT-5 Bi-FET (very good). More promised for next issue.
*Full-length test reports: Formula 4 tone arm (hairy to set up, but sonically excellent); B&W DM-6 linear-phase speaker (pretty good when used with solid-state amps); Infinity Monitor Jr. Speaker (excellent imaging, neutral sound except for peak at top end).
*The FMI cult: How it started and why the honeymoon is over (Fulton's too-frequent design changes went too far when the last change made the sound worse instead of better).
*The Death of Direct: A speculation that the direct-disc fad may quickly die when digitally-recorded master tapes become widely available (p. 55).

Stereo Review, January 1978

*Audio Basics: On learning to be an analytical listener (p. 30).
*Audio Q&A: Larry Klein bluntly points out that the 5 to 7 percent "IM distortion" commonly reported in High Fidelity tests of cassette decks "is quite simply a misinterpretation of their instrument readings" -- actually representing modulation noise, as reported in the Speaker two years ago (p. 24).
*Tape Talk: Including the embarrassing fact that many so-called "rack mountable" components do not have EIA-standard vertical dimensions (p. 32).
*Technical Talk: Julian Hirsch asks "Are Measurements Valid?" and discusses some of the

limitations of objective data (p. 36).

*Equipment Reports: Sherwood Micro/CPU 100 digital tuner (a super-enthusiastic rave review); ADS 810 speaker (excellent except in deep bass); Yamaha CR-620 receiver (excellent); Sanyo 2000 cassette deck (pretty good at the price, but not designed properly to exploit CrO2); dbx 128 dynamic processor (excellent) (p. 37).

*Audio Breakthroughs: Peter Sutheim surveys and comments on the various current high-technology devices -- Class D/G/H amplifiers, carbon-fiber cones, HPM tweeters, ferro-magnetic fluid, linear-phase speakers, SID, time-delay, crystal-controlled turntables, VFETs and power MOSFETs (p. 70).

*Note that the classical record review section is longer than it has been lately, and has been restored to the front of the Music portion of the magazine, instead of being buried at the end of the issue. Now if the reviewers could just be instructed to pay more attention to disc quality ..

-- Peter Mitchell (Massachusetts) and J. Burdych (Czechoslovakia)

December BAS Meeting

The December 18th BAS meeting featured a panel discussion on the use of ferrofluids in loudspeaker manufacture and design. The arrival of one of the panelists was delayed, and during the interim several interesting subjects were discussed.

Two reports on Grado cartridges were delivered, one negative, one mixed. The first example was a model F3e+ mounted in a Dual 1229, which has groove-skipping problems in response to external vibration. An Ortofon VMS20E and a Micro-Acoustics 2002e track satisfactorily in the same arm at 1 gram, but even at 1.5 grams the Grado tends to jump grooves. (The sound, skipping problems aside, is good.)

The second report described a Grado F2+ in a Connoisseur SAE-2 arm with paddle-damping added. The owner's previous cartridge was an ADC XLM, vintage unspecified. The Grado, for which the recommended tracking force was 1 gram for stereo, 1. 5 grams for CD-4, required 1.6 grams of VTF to track properly, but then did better than the XLM at 1. 3 grams. Sound was airier and thinner than the XLM's, with sibilants more prominent. A frequency response test showed the Grado to be up 2 dB in the 4-7 kHz region compared to the XLM.

Ira Leonard mentioned once again the General Electric MOV (metal-oxide varistor), a device for removing short, high-voltage spikes from the AC line feeding hi-fi or tv equipment. The problem seems to be the occasional pulse in the <u>kilovolt</u> range; a peak-holding voltmeter placed across the line in Ira's office sometimes pins on the 3000 V scale. This problem is especially bad in the Back Bay area, and WBUR has spent several thousand dollars coping with it. The GE device costs well under $5, however, and should protect several pieces of equipment at once. It is placed across the AC line near where the gear to be protected is plugged in, and shorts out anything over 130 V (this presumably means 130 Vrms, since 117 Vrms has a peak voltage of 165). According to Ira, all Dunlap-Clarke power amplifiers have one of these across the power transformer primary, which should protect other equipment plugged in with the amp as well.

The question of a multi-hundred dollar BAS contribution to Victor Campos' BSO microphone fund was raised. President Jim Brinton felt it improper to use Society dues money for things that benefit only local members (a fourth to a third of total membership) and that the best way to handle the issue would be for individual members to contribute. For those who are unfamiliar with the fund, the idea is to buy a pair of Studer/Schoeps microphones and hang them in Symphony Hall in place of the Neumann KM-83s now in use. If the microphones are bought, they will be hung, it is said. Background on this issue appears elsewhere in this month's <u>Speaker</u>.

Those interested in contributing to this project should make out a check to WGBH and send it to the station at 125 Western Avenue, Boston 02134. IMPORTANT: Be sure to address the envelope to Victor Campos and mark it "PERSONAL." Otherwise, your contribution will go into the WGBH general fund and will not reach the microphone fund. Also, to identify yourself and the Society, write "BAS member" on your check. Those who have used both these microphones, including Campos, Dick Burwen, and David Satz, all report the Schoeps unit sounds smoother and better overall, although the Schoeps may have its own problems. We will hear more on this from these men in a future issue of the Speaker. (We're trying for February. -- Ed.)

In addition, if any members have comments on the use of Society funds for projects such as this one, please send them to Box 7. It is, after all, your money, as Brinton noted.

The final item in the business meeting was a report from member Dr. John Ouzts on the BIC Beam Box FM antenna. Dr. Ouzts lives in a location haunted by many ghosts, and he says the Beam Box is the best multipath cancelling device of the many he has tried. It should be available as of the time you receive this issue at Atlantis Sound for around $60. Be advised also, however, that John feels the Channel Master Super Chroma powered rabbit-ear antenna to be 85-90% as good as the beam box, for around $25.

Meeting Feature: Ferrofluids in Loudspeakers

The panel discussion on ferrofluids was prompted originally by Roy Allison's response to an article in the Boston Phoenix. Roy's letter was reprinted in last month's Speaker, and seems to point to a difference of opinion among knowledgeable local people about the properties of ferrofluids. Several of these people were, accordingly, assembled to address the Society. What followed was not a debate in the tradition of the TIM fracas, but rather a discussion of the subject from several points of view, with much interesting detail about the manufacture of loudspeakers thrown in.

The first to speak was Dr. William Bottonberg, Product Manager, Fluids, of Ferrofluidics Corporation, the principal commercial developer of the material. Ferrofluid was developed originally because NASA wanted a liquid that could be circulated by magnets, to make a spacecraft cooling system with no moving parts. The idea was to use magnetic forces to circulate the liquid from the hot, sunny side of the module to the cool, shady side, where it could radiate its heat into space. NASA later came up with another cooling system, whereupon the contractor (Avco) lost interest in ferrofluids, and a small company was born, possessed of (in Bottenberg's words) a solution looking for a problem. Their first use for the new substance was in vacuum seals for rotating shafts. In 1974 EPI began using ferrofluid in their loudspeakers.

The fluid consists of three parts. The base, or carrier, is diester oil, as used in some synthetic lubricants for cars. The iron particles are magnetite (Fe_3O_4), about .01 micron in diameter. They are kept from clumping together by a stabilizer or "surfactant, " which has molecules that are polar at one end and non-polar at the other. The polar ends cling to the magnetite particles, and the non-polar "tails" project outward into the carrier fluid, and prevent the particles from getting close enough together to clump. The particles arrange themselves according to gradients in magnetic field strength, and so tend to collect in the voice coil gap of a loudspeaker magnet when placed in its vicinity. Furthermore, because the gap in a loudspeaker is circular, there are more lines of force per unit area nearer the center pole piece, and the magnetite is therefore more strongly attracted to the inner side of the gap.

Two important parameters are used to specify ferrofluid. The first is the saturation magnetization, in gauss. This is the field strength at which the magnetite saturates, and commonly runs around 100-200 gauss for fluid used in speakers. This means that the fluid contains 1-2% magnetite. The second parameter is the viscosity of the fluid. This is hard to measure under actual operating conditions, because the viscosity changes in a magnetic field. This property is critical, and must be carefully chosen for each driver design. All but the very lightest fluids will damp the voice coil motion at the lowest operating frequencies of the driver (but not at the high end). Ideally, the fluid is chosen during the design of the driver to give the desired degree of damping at resonance.

The other panelists present were all from speaker manufacturers who use ferrofluids in their products: Roy Allison, president of Allison Acoustics, who uses ferrofluid in all his midrange drivers and in the Model Four tweeter; John Draper and Dan Hathaway of EPI, the first company to use it in any driver, in 1974; Tim Holl, director of engineering of AR, which uses it in most or all of their newer tweeters and midranges; and Andy Petite of Advent, who use it in the tweeter of their new "large" speaker.

The discussion from here on will be presented by categories, and the comments of all participants in each category summarized.

Cooling Properties of Ferrofluid

Everyone agrees that filling the magnetic gap of a loudspeaker with ferrofluid helps cool the voice coil. The fluid has about six times the heat conductivity of air. An AES paper given by John King in 1977 describing tests done on a 5 1/4" speaker shows voice coil temperature dropping from almost 180°C to 103°C for 5 Watts steady-state input at 400 Hz when the fluid was added to the gap. In this case, the maximum operating temperature also falls from 180° (the highest safe temperature for the voice coil adhesive) to 105° (the maximum operating temperature of the ferrofluid), so for this steady-state condition, no additional power can be used, but a constant 5-Watt input does not occur in real music reproduction. Besides, there are benefits in extended life of the driver and long-term consistency of frequency response from the lower operating temperature. Roy Allison pointed out on this subject that the alternative gap filler, silicone grease, has a thermal conductivity two to three times greater than that of ferrofluid, which is why he uses it in the tweeters of his three-way systems. Silicone grease will not stay in the gap, however, if this tweeter is used, lower in frequency, as it must be in a two-way system, because the longer voice-coil excursions tend to throw it out of the gap. Under these conditions, Allison uses ferrofluid also. Everyone reports greatly reduced failure rates in transducers that have a gap-filler. Advent has had essentially zero voice-coil burnouts in fluid-filled tweeters; what failures have occurred have been caused by wire fatigue. On average, levels of reproduced music are going up, creating more problems for all manufacturers. EPI has seen speakers in which the ferrofluid had been vaporized. (They also hinted darkly at instability problems with some newer, high-powered receivers, which were designed primarily to drive resistive loads.)

Ferrofluid as a Damping Material

The second most important use of ferrofluid seems to be in tailoring the frequency response near the resonance of a driver. By proper choice of viscosity, it seems to be possible to Control the Q of this resonance very well, without the need. for electrical elements to do the same job. Petite says the fluid in the Advent tweeter eliminates the need for a choke in the crossover network, so that although the fluid is actually more expensive than the choke, the impedance curve of the system is improved. AR uses fluid to get proper damping. in some situations, and uses very light fluid to avoid additional damping in others. Holl notes that in drivers with very high flux density and very low moving mass, ferrofluid adds significantly to the mass of the system, thus lowering the overall efficiency by 1-2 dB. Bottenberg added that this effect is increased because the fluid causes a slight reduction of the strength of the magnetic field "perceived" by the voice coil. EPI also uses the fluid's gap-filling ability to seal the back of their tweeter, as part of the process of tuning the resonance. Because the fluid usually does affect frequency response somewhat, its use after-the-fact on speakers not designed for it is not a good idea. Bottenberg said, nevertheless, that some renters of sound reinforcement gear use it to prolong the life of their high-frequency drivers.

Does Ferrofluid Help Center the Voice Coil?

This question produced the only real difference of viewpoint in the entire discussion. Allison's position in his letter seemed to be that centering action did not exist. Actually, his principal point on the subject is that, once a driver is assembled, the position of its voice coil is determined by the mechanical pieces of the suspension, not by any gap-filling fluid. Now, in the absence of any other forces upon it, the voice coil in a ferrofluid-filled gap will center itself; the fluid is drawn towards the center pole-piece, so inward becomes "down" for the fluid, and the voice coil "floats" to a centered position. EPI uses this effect to assemble their tweeters, by shaking the detached voice coil in a fluid-filled gap and cementing it in place when the proper position is reached. But these forces are feeble when compared to the mechanical forces holding the voice coil in an assembled driver, and it is obviously incorrect to suppose that ferrofluid has enough centering action to eliminate the need for an entire mechanical element, such as a spider. In addition, what centering force there is is weakened still further if the fluid is on both sides of the voice coil, as opposed to being just on the inside, as it is in some AR designs. (The one AR driver that won't work without the fluid is one of these.) It is likely that much of the alleged centering action of ferrofluid really comes under the next category.

Ferrofluid as a Lubricant

One of the primary signs of incorrect assembly in a driver is what is known as "voice-coil rub." If the coil actually contacts the side of the gap, it makes a nasty sound, which is clearly audible when the speaker is playing music. On the basis of the evening's discussion, it seems likely that the sterling lubricating ability of the diester base allows some voice-coil rubs to occur without objectionable effects. (This theory comes partly from the statement by one of the participants that it is actually difficult to get a fluid-treated voice coil to rub, even by pushing it sideways.) This "oil" is the same stuff used in Mobil One, advertised as a miraculous lubricant. Tim Holl said that AR first used ferrofluid as a lubricant and only later came to appreciate its properties as a coolant.

Miscellaneous Observations

Roy Allison's preference for silicone grease where possible as a gap-filler was echoed in a statement by Bottenberg that a silicone-based ferrofluid would be nice, but has so far proved an impossibility because surfactants don't work in it. The surfactant is the most expensive ingredient in the mixture; it costs $157 a pound, and its density is 2, so a pound isn't very much of it. Some potential problems in using ferrofluid (or in retrofitting it to older drivers): it dissolves some epoxies, and some enamel insulation, at high temperatures. It is gelled by the activator used with Loc-tite compounds. (Its long-term stability seems very good; this was said by both AR and EPI.) Ferrofluid has not been successfully used below around 500 Hz, because excursions become high enough to throw even it out of the gap. This type of accident is very messy, sufficiently so that Ferrofluidics has a standing offer to exchange clean shirts for spattered ones for engineers looking at new applications.

Finally, there were a few tidbits of advance information from the individual companies. EPI is going into the car speaker business. A system consisting of a tweeter and a 6" woofer in a standard 6"x9" cutout for around $150/pair should be out by the time you read this. A three-way car system is planned later. Advent is bringing out an amplified car speaker for a target price of $150/pair. Allison Acoustics is planning to market an "electronic subwoofer," which will extend the bass of Allison speakers (and maybe some others) down to 20 Hz with absolute flatness. Roy reports subtle differences on many records, dramatic differences on a few. Cost should be about $250. AR's new turntable, the 77XB, has an arm with lower mass and slightly better bearings. The arm is not retrofittable to older tables.

-- E. B. Meyer

A Publication of the BAS

Modifying the Rabco SL-8: The Designers View

Jacob Rabinow

I had an opportunity recently to read several issues of The BAS Speaker, and I was flattered and amused by several articles that had to do with various modifications of my SL-8 phonograph arm. Apparently some sort of cult has grown up over this device, and I am amazed that people are willing to spend several hundred dollars to modify an arm that I used to sell for $150. In this article, I will suggest the modifications that should be made and outline test procedures to eliminate some of the hazy theory that surrounds work on phonograph arms.

Figures 1 and 2 are of devices by which some of the characteristics of any phonograph arm can be measured. I strongly recommend that anyone seriously interested in work in this field build these two gadgets and use them to measure the effects of what he is doing.

Figure 1 shows a method of determining the equivalent mass of a phonograph arm at any point. The device consists of a small loudspeaker, in the center of which is glued a small paper cone, such as shown in Figure 2. A small metal hook is glued to the apex of this cone. Suspended from the hook is a small steel spring, which can be seen in Figure 1. Either a weight or the phonograph arm hangs from this spring by a thread. Normally, the thread is located in line with the stylus. The loudspeaker can be driven by any good oscillator that can provide output between 2 and 30 Hz. No amplification is needed, because the power required to drive the speaker is small, and, in practice, the speaker cone barely moves.

If a weight is hung onto the assembly, it is easy to determine the frequency at which the weight and spring oscillate rather violently. In my setup, the weight moves up and down on the order of a quarter of an inch. A single weight is all that is necessary to calibrate the spring, because the resonant frequency of such a system varies inversely as the square root of the weight. If one doesn't wish to do calculations, it is easy to determine the frequency for different masses and plot a curve of frequency versus mass. This should produce a smooth curve, which can then be used in checking phonograph arms and cartridges. I would suggest that the range of weights actually used be between 5 and 40 grams. Four such weights can be seen in Figure 1.

When the setup of Figure 1 is used, one can easily determine the effects of different masses added, or subtracted, at various points along the arm or in the gimbal system. One can even make measurements of the effects of friction in the pivots or the effects of damping relative to the base of the machine -- damping of which, incidentally, I do not approve. To measure the effect of friction, one can plot the amplitude of the mechanical oscillation against the frequency, thereby drawing the usual resonance curve. One can also determine the effect of damping by noting the change in amplitude, at a constant frequency, when the friction is increased or decreased.

Figure 2 shows a method of measuring the stylus/arm resonance in the horizontal, vertical, or 45° direction. The device consists of two small loudspeakers mounted on a base. Each of the speakers has a small paper cone glued in its center, and between these paper cones is glued a rod of balsa wood about 1/4" thick. At the center of the balsa rod is a right triangle of balsa wood, also 1/4" thick. The arm is mounted so that its stylus is placed on the balsa wood with the desired vertical force. One or both of the speakers are then driven through a low-power amplifier

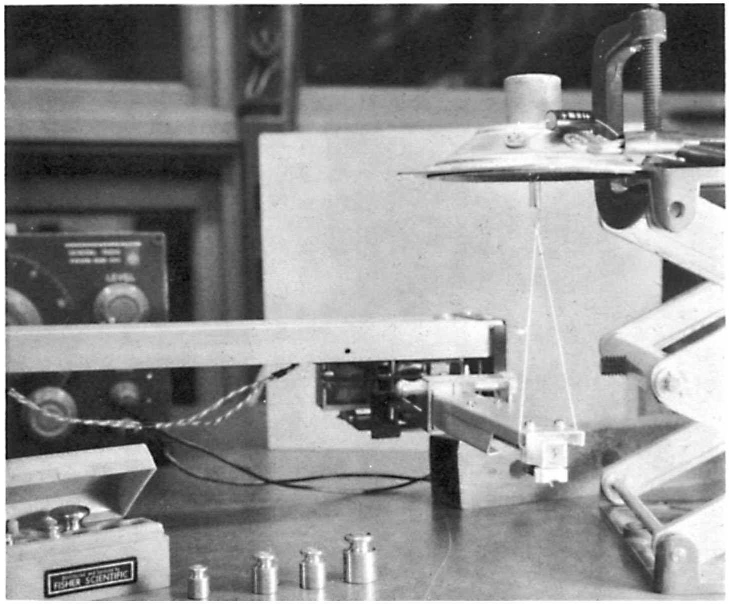


FIG. 1

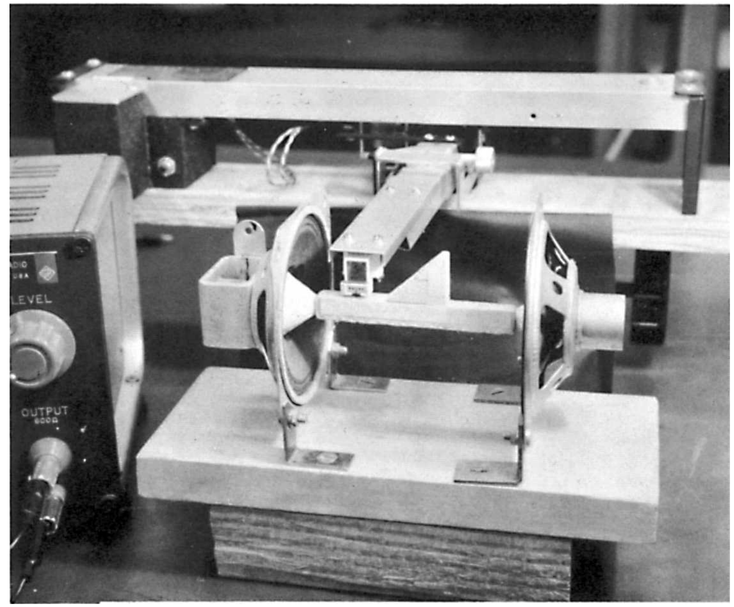


FIG. 2

FIG. 3

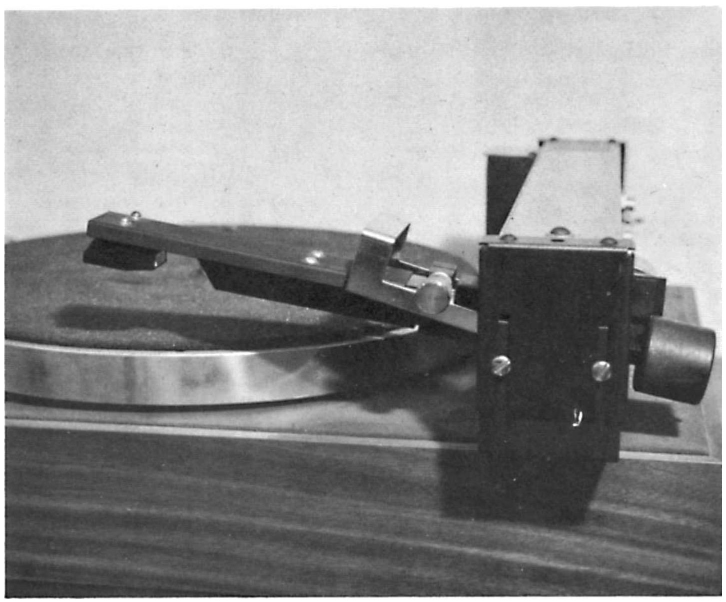
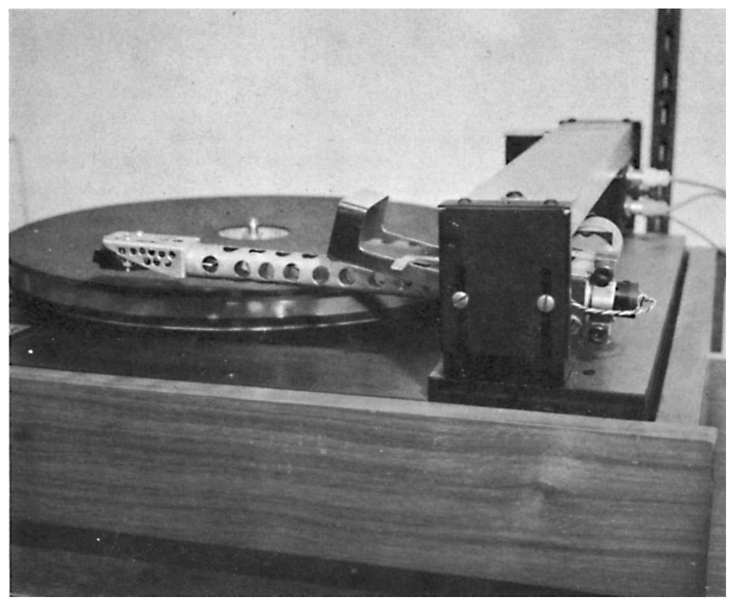


FIG. 4



(if necessary), so as to obtain some small motion of the balsa rod. At low frequencies, the cartridge will follow the wood. When resonance is reached, the cartridge will go into violent oscillations that will no longer synchronize with the motion of the piece of wood. As the frequency is raised further, the cartridge will stop, and only the wood (and the stylus) will move. The motions are easily observable by eye, and if one wants to do some precise work, one can measure the motion of the wood by using one of the speakers as a driver and the other as a pickup device. A microscope can be used to monitor the motion of the balsa wood and calibrate the pickup speaker. At these low frequencies, the speakers act as true velocity transducers.

To measure the vertical resonances of the cartridge and arm, the speaker assembly has to be mounted vertically, and the stylus has to rest on the right-angle surface of the wood triangle. For a 45° test, the assembly has to be mounted, of course, at 45°, and the stylus has to rest on the diagonal of the triangle.

So much for this fancy test equipment, which people who play around with phonograph arms should certainly build. Let me get to the business of modifying the Rabco arm. Figure 3 shows one of the standard arms, of a rather large mass, with a particularly heavy cartridge. When this arm was designed, I knew perfectly well that a lighter arm could be built. I had to design an arm that would be rugged enough to be used by all sorts of non-technical people, that could be easily pulled in and out of its socket, and that, could be changed in length to fit different turntables. Ed Myer of Myer-Emco tells me there is a thing called the "lunch cocktail effect," which he has observed in his stores. People have several cocktails at lunch and then come in to buy a hi-fi system. They think nothing of moving the arm across the record regardless of whether it is up or down, and if the arm doesn't move they make it move. They wreck records and styli. My SL-8 could handle most of such problems, but even people who are not drunk do amazing things to sensitive mechanical equipment.

Quite a few years ago, I gave my daughter (who was then living in Boston) one of my arms. We found that the floor in her apartment was so flimsy that any time anyone walked on it, the table on which the machine stood shook violently enough to kick the stylus off the record. So, I made a special model for her, shown in Figure 4. I took the breach block out of the gimbal and cut it away so that almost none of it was in front of its pivots, but some of it extended back. The new block is made from the front end of the original SL-8E block. The tube clamp is now behind the pivots and below the tube. I made a new arm of a thin aluminum tubing that extended right through this breach block. I mounted the connector at its rear end. The tubing was of the same type as used at the rear of the commercial SL-8. To lighten the arm further, I bored a series of large holes throughout the length of the arm and epoxied a light weight aluminum head to the front end. This can be seen clearly in Figure 4. Drilling these holes was no small problem. I did it by sliding a piece of wooden dowel into the arm, which stayed there during the drilling operation and was removed later.

I needed some additional counterweight material, so I used two strips of metal clamped together by two small bolts at the rear end of the arm. Thus, almost all of the breach block mass, the connector, and the additional counterweight mass together acted as a single counterweight. This arm had an equivalent mass at the front of about 5 or 6 grams. To the top of the breach block is cemented a thin strip of aluminum, which has a right-angle finger that cooperates with the lifting and lowering mechanism of the SL-8. This finger can be seen in Figure 4. Replacing the breach block with metal other than aluminum makes no sense to me, because if one makes the modifications discussed, the mass immediately behind the pivots serves a useful function, being a part of the counterweight.

And while I'm on counterweights, let me note that the closer the counterweight is to the pivots, the heavier it must be (at the pivots, it becomes infinite). Let me illustrate. Assume a perfect, weightless arm, 8 inches long from cartridge to pivots. Assume that the cartridge is very small and weighs 10 grams. Assume that the counterweight is placed 2 inches from the bearings and that it is also physically very small. It would have to have a mass of 40 grams to balance the cartridge. Because its equivalent mass varies inversely as the square of its distance from the pivots, its equivalent mass at the cartridge will be 2 1/2 grams. The total mass at the cartridge will then be 12 1/2 grams. The total weight on the pivots will be 50 grams. Now move the counterweight to 1 inch from the pivots. It would have to be 80 grams, and its equivalent mass at the cartridge will be 1 1/4 grams. The total mass at the cartridge would then be 11 1/4 grams, but the total weight on the pivots would be 90 grams. This means that for a 10% reduction of the equivalent mass at. the cartridge, we have raised the load on the pivots by 80%. The change in the stylus/arm resonance is even smaller -- only about 5%. If we move the counterweight to 1/2 inch from the pivots, its mass will have to be 160 grams, and the additional benefits will be negligible. The moral of this story is: "Go easy."

I do not understand why anyone would use a Nylon screw for mounting the counterweight. Why not use steel screws of two or, at most, three lengths if one goes to the trouble of using more than one size of counterweight?

For those who really want to minimize the effect of a counterweight, why not abolish it altogether and use a spring to produce the counterforce? The spring should have a long equivalent distance of extension (or torsion) so that small vertical motions of the cartridge do not materially affect the force. The spring can be a tension spring or a torque spring and, normally, it has to be anchored to the gimbal ring. There are methods of anchoring such springs directly to the frame, but I leave that as a puzzle to the reader. Having experimented with many arms, of more types than I care to remember, I prefer to use conventional counterweights and to design the pivots to take a reasonable and convenient load.

I have tried an arm made of thin balsa strips (1/16 by 1/16 inch) glued together into a box-girder section. Its equivalent mass is only 2 grams, but it is so flimsy that I would not suggest it as a practical design.

Now, about pivots. I think it is nonsense to use ball bearings instead of the pivots of the SL-8, and I suggest that anyone who thinks that putting a different lubricant into the ball bearings will make a difference in the damping of the arm build the device shown in Figure 1 and see for himself. It should be remembered that even pivots that feel loose are not loose when subjected to the weight of the arm, cartridge, and counterweight. Only about 1 gram of this combined weight rests on the record. The rest is supported by the pivots, and they can hardly be called loose under these conditions. It should also be remembered that the arm is not perfectly still when it is reproducing music. Vibrations of all types do travel back and forth through the arm, and they tend to keep the pivots free of static friction. In any case, I have not observed any effects of the pivots as long as they have felt free, and if they don't fall free, the arm will not play records correctly at all.

The greatest weakness of the SL-8 is in its electrical servo contacts. Even though one of the contacts is a platinum wire and the other is gold, and even though there is a transistor between the contacts and the motor, they give trouble. This is because the atmospheres of many cities are dirty. An oil film develops on the contacts and they fail to "close" when they touch. The repair consists of cleaning the contacts with suitable chemicals. My company lost a lot of money doing this as part of its guarantee. My advice, therefore, is that if one wants to modify the arm in any significant way besides reducing its mass, he should replace the contacts with some photoelectric system. One can, for example, mount a small vane in place of the moving contact and interrupt the light between a small bulb and a photocell. The photocell can be located behind a narrow slit, so that the sensitivity of the mechanism can be made quite high. I made such an arm many years before I made the SL-8, but didn't put it into production because it did not lend itself well to battery operation. And I didn't foresee the difficulties with contacts in cities outside of Washington.

Perhaps a better way of using a photocell is to mount only a mirror on the arm (or on the gimbal) and to place the light source and the photocell at the extreme left end of the whole system. When the mirror swings, the image of the light would move a large distance over the photocell. The system could be made very sensitive and would have the advantage that the moving carriage would have no electrical control circuitry on it. My staff and I designed and made such an arm, which had no difficulty detecting a stylus motion of one groove width. As long as the servo moves in only one direction, such high-gain systems remain stable.

As regards speeding up the lift motors, I suggest that one go easy. Higher voltages and higher speeds cause these motors to produce signals that can be picked up by the audio circuits, and they are a nuisance. One can, of course, provide a relay operated by a trip circuit so that the output of the cartridge becomes short-circuited during the lifting or lowering cycle. Personally, I do not think the speed of lowering or raising the arm is that important.

I would like to make some comments about the arm that is shown in The BAS Speaker of February 1977. If one really wants a light, rigid arm, one should use either a completely closed box section or a tubular section. These give the greatest torsional and flexible rigidity for a given weight. The arm Shown is not a good one in respect to torsional stiffness. I would suggest that one use thinner material and close the bottom, and that the closure should extend as close to the cartridge as possible. The reason I like metal, rather than wood, is that it provides shielding for the wires. I also believe that one could make very light arms out of aluminum tubing too thin to be practical by itself, but which could be filled with polystyrene foam to stiffen it enough to stand ordinary handling.

Because the normal mass of a cartridge is somewhere between 5 and 10 grams, making the equivalent mass of the arm much smaller than 5 grams (at the stylus) has relatively little effect. It certainly would be nice to get it down to 2 or 3 grams, if possible, and it would be even nicer if the cartridge manufacturers made their cartridges as light as possible and eliminated the present mounting system, so that cartridges could be plugged directly into a very light socket.

One of the articles made fun of the eccentric weight of the knurled knob of the SL-8. I don't know why the eccentricity bothers anyone. I suggest that the setup of Figure 2 be used to test the effect of these weights. It is easy enough to lighten them.

In summary, I am flattered that someone is willing to pay $400 to improve the arm, but I believe only three points really need attention. First, the arm system should be made very light, with all the unnecessary weight at the back, where it can act as a counterweight. Second, the arm should be made very stiff both longitudinally and torsionally, which almost automatically means a box (square or triangular) or tubular section. Finally, both contacts should be replaced by a photocell or other non-contacting device. The trip contact, however, can be left as is, because at this point the cartridge motions are large, and there is considerable "wipe" of the contact. This mostly overcomes the oil problem. The servo contacts should certainly be eliminated.

It seems to me that the greatest improvement that can be made in cartridges, exclusive of the transducer function, is to make the cartridge lighter. I recently read that someone in England is going to make a cartridge that weighs only 1 gram. If this were possible, one could then try to design an arm with an effective mass of a fraction of a gram, and the arm would then not need counterweights. Under such conditions, the cartridge would not have any problem in following almost any warped record, for it could move down at an acceleration of about one g.

For those who really would like to play badly warped records, perhaps for transcribing them onto tapes, I suggest the following procedure. Make a ring of metal that could fit on the outside of the record and clamp it to the turntable, either by sheer weight or by suitable clamps. Clamp the center of the record with a 3-inch round plate fastened to the spindle of the turntable. I have seen very few warped records that wouldn't be flattened almost completely by such a procedure. I have even obtained a patent on a mechanism that would do this relatively easily. (The patent, by the way, is now assigned to Harman/Kardon.) I do not know whether this is a practical solution for general purposes, but for those of the readers who are really fussy (and apparently there are quite a few of them), this seems to be a much better approach to playing difficult records than worrying about the polish of the pivots, the oil in the ball bearings, and other such items. For those who are so pure in heart, I also suggest that the turntable be mounted on a very firm base and as far away from the loudspeakers as possible -- preferably in another room, in a basement, or in a neighbor's building. The vibrations induced into records by loudspeakers do some very interesting things to reproduced music. The effects are much more serious than those produced by the viscosity of the oil in the pivots.

I would also like to say something about damping. As I said in the talk I gave before your society last year, I do not like damping the arm by a dash pot mounted on the frame. That kind of damping "ties" the arm to the frame and is useful only if the whole frame is vibrated, as when it is on a flimsy table which, in turn, is on a flimsy floor. But this type of damping is highly undesirable for following warped records. Here one would like to tie the arm to the record. I have proposed to do this, as shown in one of my patents, by using an air bearing between the cartridge and the record, so that most of the weight of the cartridge is supported not by the counterweight but by this air bearing. Thus, as the surface of the record goes up, the arm goes up with it, and as the record surface falls, the arm falls with it. The difficulty with such a device is that it requires an air pump and rather careful design of the air bearing to be as light as possible and mounted as close to the stylus as possible. If one is going to use a pump in the living room, one might as well use a vacuum turntable and pull the record down, hard, so that it becomes almost perfectly flat. The question of arm mass and many other things then become academic. I have built such a turntable, but do not seriously recommend that anyone else do so unless he has a basement conveniently at hand, a large vacuum tank to filter the pulsations of the pump, and the money required to build this special turntable and the rest of the equipment. I can suggest other ways of flattening warped records, either before they are played or on the turntable, but I think I have confused the issue enough for now.

All I can wish for the experimentalists who work on my SL-8 is that they have as much fun as I did (at times) in designing and building it.