THE B.A.S. SPEAKER

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Finally, a preview of rekaepS ehT for you folks who look at the cover and then read from the back to the front. Starting off (or ending up depending on your point of view) we have an article having to do with the final element in your playback chain, the speaker. A1 Foster reports on that monument of Acoustic Research's technology, the AR9, and a full year of living with it and moving it around his listening room.

Another unique speaker is previewed in the February meeting summary. It's the egg-shaped ECD ("Equalized Controlled Diffraction") speaker by Leigh Instruments. And, if you missed the ads and still thought Revox was only a maker of tape recorders, you'll get a thorough insight into their other products in the same report.

Moving on (or back) we have some spring maintenance tips for your lowly phono plugs, a simple way of having your turntable improved, and a new perspective on aligning your cartridge for minimal distortion. And so, as you can see, there's a lot to look backwards to in this issue of The Speaker.

The Faster We Go, the Behinder We Get Department

Our February issue was ready in jig time and then spent two weeks in the BAS "mailroom" awaiting envelopes. So, on the one hand, we can't win for losing.

On the other hand, none of that has slowed up production on the March and April issues. This issue has a guest editor (see below) and while we (I) have been working on it, they (Brad) have started on April and the two issues should be about two weeks apart at the typesetter.

But on the other hand, it turns out to be difficult to produce a full-size Speaker every three weeks when it takes four weeks before we receive enough material to make a full issue. The alternatives seem to be (1) to engage a staff of metaphysicists to come up with a better system, (2) to live with slightly skinnier Speakers for a while, or (3) to plead with any of you out there with story ideas to write them up and send them in without delay. We have decided to (1) engage the metaphysicists, (2) live with skinnier Speakers for a while, and (3) plead for contributions. The address for said contributions appears below. Consider yourselves pleaded with.

-- Henry G. Belot

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Queries and Quandaries

Quad Crossover

I am about to add a subwoofer to my system, so I have made some measurements on my Quad loudspeakers that I think would be of interest to BAS members. I have single Quads, not stacked Quads. I measured phase and amplitude of the acoustic output with a flat electrical input to the speaker from 200 Hz down. I centered the microphone on the bass panel and altered the position of the speakers relative to the floor and the adjacent walls. The results do not appear to depend on room placement.

From 200 Hz to 100 Hz the signals are more or less in phase and the amplitude is flat. Below that, the data show a phase lag of the acoustic output from the electrical signal at the speaker terminals that reaches 180 degrees out of phase at the -6 dB point of 50 Hz. Below 50 Hz the output drops at about 18 dB per octave. There is a 6 dB amplitude peak at about 80 Hz.

I cannot decide whether to cross over to the subwoofer at 100 Hz and put up with the subwoofer distortion extending into the voice range -- I will use a 24 dB/octave crossover -- or to cross over at 50 Hz which eliminates the voice distortion, but introduces phase distortion. I could arrange signal polarity so that a positive input signal gives a positive pressure front at low frequencies only, but that might reduce realism at higher frequencies. Any ideas?

-- Peter V. K. Brown (Maryland)

Hafler Preamp

A1 Foster's report on four preamps was particularly interesting to me, since I own a Hafler preamp. The manner of his testing is very significant. More reports should be like this.

I suspect that the quality of the null at the phono inputs of current runs of the Hafler may be better than merely "good," and that Al's results may not completely apply to units with serial numbers 1919000 and higher. These units contain different phono transistors and phono resistors. They are available to owners of earlier units as kit DH-106 for $19.95.

I updated my unit, but cannot vouch for any sonic change since my speakers are undergoing some changes also. Would it be too difficult for A1 to look into the effect of the DH-106 update?

-- Carlos E. Bauza (Puerto Rico)

(At present I am investigating amplifiers around 200 watts. I won't have much time to retest the Hafler, but I am sure it is still an excellent value. -- AF)

Two Construction Projects

Popular Electronics has two construction projects in the September, 1979 issue which, if decent, look to be of considerable interest. One is a two-band stereo parametric equalizer built around the TL074CN IC (pp. 47-50, 57-8, and 60-1). The parts, including complete kit, are available from Phoenix Systems, 375 Springhill Road, Monroe, CT 06468). The other is a hand-held spectrum analyzer which looks like a poor man's Ivie (pp. 62 and 64-8). Parts including complete kit or assembled unit are available from Gold Line, Inc., P.O. Box 20, Redding, CT 06875.

I wrote to Gold Line about the second item to inquire about the calibration of the mike in it and was told, "If you buy the kit you will find the mike of excellent quality. Much testing and calibration was done before it was selected to be used in the unit." Their spec sheet says, "The internal electret microphone is amplified and compensated to give a flat response ±1 dB from 30 Hz to 16 kHz. Therefore any input to the 'line in' jack will be flat throughout." I was hoping that when "In the Literature" got around to that issue, there might be some comment on the probable virtues of these units, but it seems to have fallen between the cracks. The September, 1979 Speaker listed the August issue of Popular Electronics and the October-November Speaker listed the October and November issues.

Does anybody know anything about these two units? Has anybody tried them? The prices are very attractive ($99 for the equalizer kit; $139 for the spectrum analyzer kit, with a pseudo-random pink/white noise generator for $39.95 in a kit described in PE for February, 1980, pp. 67-8, 70, and 72-3). They are also so low as to raise questions about quality. If anybody has experience or other useful comments on these units, will they let us know?

-- R. N. Wisan (New York)

McIntosh Speaker Claims

The McIntosh brochure advertises their speakers as "the best in the world." They state that their woofer has perfect transient response as a direct result of their very low Q design. They acknowledge the disadvantage of very low Q: a prematurely drooping response curve (though they describe it as a rising response curve). Their solution to the bass slope is their magical "environmental equalizer," with a maximum boost of 15 dB at 20 Hz.

What bothers me about their implications is that, according to them, woofer systems with higher Q and flat response come out with necessarily imperfect transient response, that is, they ring. They imply that a flat-responding woofer is "... the traditional solution with its emphasis on cost reduction rather than reproduction accuracy."

My two questions are: Is this relationship between Q and ringing true? Is this significant? A yes answer to both questions would mean that the majority of woofers on the market are wrong. Somebody knowledgeable, please comment. -- Carlos E. Bauza (Puerto Rico)

Another Fanatic on Cartridge Alignment

(In our September, 1979 issue, we published a brief article from Ray Kilmanas entitled "The Fanatic's Guide to Cartridge Alignment." Using the Seagrave equations and a random selection of inner radii found on records in his collection, Kilmanas concluded that "published values for offset and overhang are not to be held sacred" because fairly small differences in the inner radius value (R_1) resulted in surprisingly large changes in the optimum values for offset angle and overhang. With tongue near cheek (we thought), Kilmanas asked, "Since (conventional arms) can never be state-of-the-art, why persist in this exercise in futility when, deep down, we all know that a good tangential tracking arm is the only way to go?" The article inspired Jiri Burdych to take a different approach to the question. -- HB)

While it is true that as a whole tone arm design is full of compromises, proper lateral geometry can be achieved without affecting other aspects of design. The math is simple and good design in this area involves no extra costs unless one strives for exceptionally low errors by choosing a very long arm. In this respect, I am not convinced by Ray Kilmanas' contribution in the September, 1979 Speaker, and in fact, do not understand what he's trying to recommend. Are we to be called fanatics unless we ignore Dr. Seagrave's article because the inner groove radius, R_1, on commercial records varies? To be sure, the proper value for R_1 is up for debate and this could involve other factors besides those mentioned. There is also the question of the audibility of tracking error distortion which is often used as an argument against worrying about tone arm geometry.

I cannot agree with Ray's way of interpreting arm misalignment. Comparing two arms or their setting by the mere differences in offset and overhang is rather confusing and useless, to say the least. It is the values of the tracking errors, or better yet of distortion factors, which are pertinent. (By the way, since Ray stopped his calculations too early at the D_1 value, the given overhang values are not correct and should be 0.72" and 0.95" respectively. Kilmanas' overhang values for D_1s of 2.4 and 3.0" were 0.69" and 0.90".)

I have tried to make the difference between Ray's method and the distortion factor method clear in Figure 1. I have used the distortion factor, m, for examples given in Ray's article; i.e., R_1 = 2.4" and 3.0" and also the extreme, 2.2".

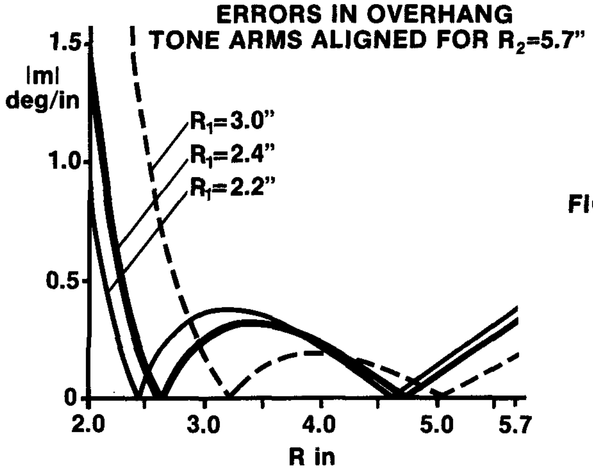


FIG. 1

It is evident that in the groove range between about 3.0" and 4.2" and again between 4.9" and 5.7" (the outer groove radius), the arm aligned for R_1 = 3.0" yields better m values than that aligned for R_1 = 2.4". The worst point is at R = 3.2", where the two arms have their approximate zero and maximum m values respectively. Here, the m value for the arm set for R_1 = 2.4" is 0.32 degrees/inch. This is still quite an acceptable value. It can therefore be said that the 2.4" arm behaves fairly well even in the range between 3.0" and 5.7" where by Ray's terms, it is misaligned.

What happens below 3.0" is another story. Here, the 3.0" arm becomes "misaligned" almost immediately. The distortion rises rapidly and quickly exceeds the m values for the 2.4" arm. At the 2.4" radius its m value is already 1.4 degrees/inch which cannot be overlooked.

This example may demonstrate that it is more reasonable to use lower R_1 values when designing or aligning an arm even if this involves higher minimized distortion factors. Note that even the third case of an arm set for the extreme R_1 = 2.2" can be considered as more convenient than that set for R_1 = 3.0".

However, each of these arms was, in fact, set properly for specific premises, and it is not proper to label any of them "misaligned," even if one set of premises may be found more convenient in general use. As for the most published and recommended value for R_1, 2.4", this need not be held sacred by those who know the problems involved. A different alignment may be used if considered more desirable. It's like Christmas gifts -- unless you know otherwise, you have to hold sacred that they are from Santa Claus.

Since the actual R_1 values are of a statistical nature, the specific value for the arm alignment should be chosen accordingly. Assuming Ray's selection of thirty record sides is representative, then with an arm aligned for R_1 of 2.4", about ten percent of records will be tracked with a higher than minimized distortion in the inner grooves. But on the other hand, only ten percent will fully benefit by the lower value of minimized distortion. Ten percent will be tracked with the same distortion as if the arm were set for R_1 = 2.4", while fully eighty percent of the records will suffer from substantially higher distortion.

A more representative analysis would require more data, but try to find somebody prepared to undertake the pains of measuring R_1 on several hundred records. Even from the limited data, it can be seen that the "sacred" value of R_1 = 2.4" is not groundless, but is, in fact, statistically sound. A similar situation exists in "aligning" the tracking ability of cartridges or the power of amplifiers to levels existing on records. From the statistical point of view, even this cannot be 100% perfect and, in addition, involves substantial expense.

In spite of all I've said, there are arms which are misaligned in the true sense that offset and overhang values are not fitted to each other. Such arms do not yield equal distortion factors at the three critical points R_1, R_2 = 5.7" and $R_p = (L^2-S^2)/L\sin\beta_0$, where L is the effective arm length, β_0 is its offset angle and S is the pivot to spindle-center distance L-D, D being the overhang. Unless the cartridge is twisted in the headshell, the most probable error in setting is that of the overhang.

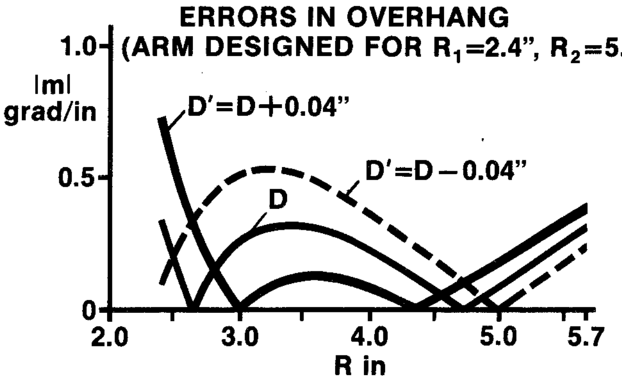


FIG. 2

Figure 2 shows how the m values change when the overhang is set with an error of ± 0.04". In an arm aligned for R_1 = 2.4" and R_2 = 5.7", the proper setting gives m_{max} = 0.32 degree/inch at the critical points (L = 9"). The +0.04" error in D will increase the m value to 0.72 degree/inch at R_1 and to 0.39 degree/inch at R_2, while at R_p, m = 0.12 degree/inch. The -0.04" error has an opposite effect, increasing the m value to 0.53 degree/inch at R_p, while at R_1 and R_2, m will be decreased to 0.09 and 0.25 degree/inch respectively. The Rp radii will not remain the same in these cases, being 3.38", 3.56", and 3.20".

Similar changes in the m values would be caused by an error of ±1 degree in the offset angle, but here the plus error has an effect comparable to that of the minus error in overhang and vice versa. It is evident that an unfavorable combination of both errors can increase the distortion factor considerably.

It can be argued though, that some other relative values of m at R_1, R_p, and R_2 rather than equal may be preferable, allowing for the sensitivity of the record's musical content to the tracking error. It is apparent, however, that such an alignment would hardly be feasible due to the inherent "mechanism" of the changes in m: striving for $R_1 < R_p < R_2$, for instance, the alignment causes m at R_2 to rise beyond that minimalized for R_1 and R_2. Besides, there is no analysis available of how these three values should be distributed.

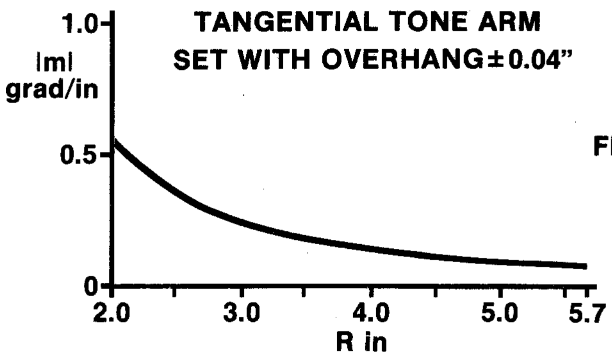


FIG. 3

While I agree with Ray as regards the tracking advantage of tangential tone arms, it has to be realized that even these arms are not error-proof. To insure true tangency the overhang has to be zero, of course. A misaligned arm will cause tracking errors which increase from the outer to the inner grooves. The tracking error angle α is given by $\sin\alpha = D/R$, where D is the unwanted overhang from misalignment and R is the groove radius. For small values of D, the error can be approximated as $\alpha \sim = D/R$, so that the distortion factor $m \sim= D/R^2$ (Figure 3). It is evident that this error can totally spoil the theoretical tracking advantage of the tangential arm in the inner grooves. Another error can occur in the basic arm setting as regards the servo action: in order to activate the servo, the arm must depart from true tangency, in some designs by a significant amount. In this respect, the tangential tone arm makes no difference for alignment fanatics.

To summarize, the name of the tone arm alignment game is "be aware of the principles." Once these are understood, the necessary degree of precision in alignment will be apparent. As the practical values of the errors involved indicate, the potential distortion need not be great. While the published values for offset and overhang need not be held sacred, the chosen values should be congruent to one another. Since the alignment procedure must be undertaken in any case, why not do it as carefully as possible? -- Jiri Burdych (Czechoslovakia)

SoundAids' Rejuvenated Turntables

Hi-fi manufacturers spend a considerable amount of advertising money to convince you that the turntable, tape recorder, or amplifier you bought last year is now obsolete. SoundAids, a three-year-old company, has decided to run counter-clockwise to fashion. Their answer to ennui is to recycle turntables. Their method is to replace or substantially modify the massive arms found on older turntables and touch up other elements of the turntable itself when practical. The old arm is removed and a modern arm designed by Clement Meadmore, author of the recently published book All Sound and No Frills (Pantheon), is installed.

Figure 1. The SoundAids-modified Pioneer PL-112D

If you don't have a turntable to modify, for $200 they will supply you with a redesigned Rotel RP2300. It includes their complete package of modifications: additional chassis damping, low capacitance cables, low-mass arm, headshell, and counterweight. The existing bearing assembly, antiskating and cueing in the RP2300 are retained.

My introduction to SoundAids was through a modified Pioneer PL-112D. It was forwarded by a SoundAids representative about a year ago for testing. The first unit I received had problems. The turntable suspension was too stiff, it ran 1.5 percent fast, and the antiskating mechanism was totally inaccurate and interfered with the arm's performance. At my request the company sent another modified PL-112D. The second unit also ran fast and the suspension was still too stiff, effective only for damping vibrations above 50 Hz, but the arm was dramatically improved. I swapped the Shure IV cartridge between my SME III and the SoundAids unit and I was not disappointed.

The unit handled itself credibly over my collection of warped and difficult-to-track records. The "open" nature of the headshell, only a bar with two holes, made mounting the cartridges a snap. However, as usual, I broke one of the fragile cartridge signal leads and had to resolder it.

SoundAids employs a disk-shaped counterweight which concentrates the weight very near the pivot and reduces effective mass. It also means a very small movement of the counterweight results in a large change in tracking force. For this reason, and because the counterweight is secured in position only by friction, adjusting it to within a quarter gram takes time and patience.

In the year or so since my tests there have been a couple of policy changes at SoundAids. They can no longer supply the PL-112D as an off-the-shelf package, though they can modify yours if you send it in. The Rotel RP2300 has replaced it.

Second, in response to a draft of this review, Clement Meadmore assures me that they are now testing for speed accuracy and shaving the drive pulley when necessary to correct any errors.

In general, the SoundAids modification can be performed on any "manual" record player which has a high mass arm. "Manual" is in quotes because the Pioneer is what the admen call an "automatic" turntable these days, that is, it indexes the arm automatically and shuts itself off at the end of the record. They also modify the Phase Linear straight-line tracker, so if you're unsatisfied with the performance of your turntable and/or arm of whatever type, you should check with SoundAids and see if they can accommodate you.

Should you choose to recycle your turntable, the procedure is simple. If you opt to recycle your AR for example, forward it along with a check for $200 to SoundAids, 800 West End Avenue, New York, NY 10025. The phone number is (212) 662-6652. It will be returned with their complete modification which includes a modified J. H. Audiolab arm with retensioned bearings.

Clement Meadmore's own tonearm appears to be of good design and probably the sonic equal of any low mass tonearm. If you have an old turntable going unused or a new one with an overly massive arm, pack it off to SoundAids. I think you'll be delighted.

-- Alvin Foster (Massachusetts)

Netronics Turntable: Up to Speed

Owners of the Netronics have a common problem when the local power company lowers the voltage supply -- the motor loses torque. In extreme cases, the turntable's servo system is taken out of its design limits and the motor starts to accelerate and decelerate continually. Since low voltage spells are a fact of life here in Puerto Rico, the Netronics problem <u>must</u> be solved. The solution is disarmingly simple -- change the power transformer. That's it!

The stock transformer supplied by Netronics is housed in a neat little plastic box which is plugged into the wall outlet. It puts out a nominal 12 volt A. C. at .47 ampere. I replaced it with one supplying a nominal 12.6 volt A. C. at <u>2.5 amperes</u>. It solved the problem completely.

My unit is the Triad Filament transformer No. F-26X costing around $10 locally. Any unit with similar specifications will also solve the problem. A friend of mine installed a 4 ampere transformer, but that seems like a bit of overkill.

This simple change has extended the life expectancy of this turntable at my home indefinitely. Its virtues are: compatibility with almost any arm; extremely low rumble, wow, and flutter; low susceptibility to acoustic feedback (according to Julian Hirsch); 33, 45, <u>and</u> 78 rpm speeds. At close to $130 (plus transformer), it's an extremely attractive performer.

-- Carlos E. Bauza (Puerto Rico)

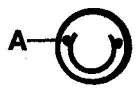
Connections

A much neglected part of every hi-fi system is the cables and connectors used to hook everything together. I think it would be worthwhile to discuss the most common element known as the phono connector or pin plug. People are always complaining about how unreliable it is, but with a little care it can be made almost trouble-free.

First, take all your cables and, while holding the outer shell of the phono plug in one hand, try to twist the center pin. If you can rotate it, then it's likely to become intermittent in the near future. The reason is that there is a wire soldered to the back of that pin and with repeated insertions and twisting the wire will eventually break. It's unfortunate but true that some of the most expensive connectors, such as the all-metal plugs, are the worst offenders in this regard. To forestall this problem, the center pin must be immobilized. In the cables we are making at db Systems, the back side of the center pin, actually a tube, is flattened and then epoxy is applied to hold it rigidly in place. When this is done, the center pin cannot be turned even with a pair of pliers.

There must be an adequate strain relief for the cable as it exits the back of the plug. Most plugs have some kind of clamp for the cable, but it doesn't hurt to add some silicone rubber or heat-shrink tubing.

The phono jack is not usually a source of problems. About the only trouble here results from the insertion of a phono plug with an overly fat pin. This spreads the spring contact in the jack so that normal pin plugs make poor connections with the contact. The phono connector is not a precision connector and we will just have to live with that fact. What can be done if the inside contact is too loose? If you examine the inside of a phono jack closely, you will see a tube with a spring contact inside:



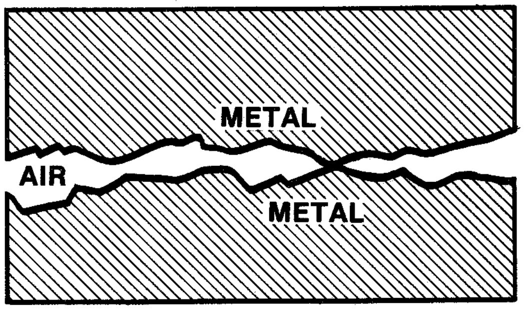
Open the instrument up so you have access to the back side of the connector. With a large safety pin and using the clasp as a handle, try to push the sharp point between the tube and the contact at point "A." If you can reach the back side and perform this simple repair, the result will be a noticeably improved connection with normal phono plugs.

The other part of the connection is between the shell of the plug and the outside of the jack. When inserting a plug, resistance should be felt as the pin meets the inner spring, then further resistance as the shell makes contact. It is fairly easy to bend the shell inward to make better contact, but be wary of the obvious technique of clamping ordinary pliers around the shell and squeezing. It's very easy to bend the leaves too much and break them. A better method is to use needle-noise pliers and bend the leaves individually.

The next issue is the quality of the contact itself and whether the ultimate contact is a gold one. A digression on the nature of contacts is necessary.

When two metals are placed in close contact, they are actually touching at a relatively small number of peaks, called asperities. The true contact area is a function of the hardness of the metals and the pressure forcing them together.

Even the areas in contact may not pass current, however. Except in a vacuum, metallic surfaces are never clean but are covered with contaminating films. If the film is thin enough, then current will flow, although there may be a nonlinear effect due to the film acting as a rectifier. If the voltage is great enough -- over 50 mV -- it can puncture the film. However, this cannot be relied upon in audio applications. Therefore, when inserting a plug, always use a twisting motion. This insures a wiping contact which will break through the oxide film for a reliable connection to the conductor underneath. If the center pin is loose, then this will exacerbate the situation, but we know you have already solved that problem.



A useful analogy is to consider a tin-plated connector as an ice-covered pond. The oxide (ice) is broken through when sufficient pressure is applied, and an essentially ideal connection is made to the conductor (water) beneath. When the connector is removed the oxide (ice) re-forms, but when the connection is re-established, the oxide is cut through again. Problems can occur only when the pressure is insufficient or when there is a continuous vibration causing new oxide layers to constantly re-form.

Gold does not form an oxide film, so it would seem ideal for reliable long-term connections. Solid gold contacts would be ideal, but for the expense. Plated contacts help, but are not a cure-all. Consider the worst of all situations: when both high and low signals are present as at the output of a power amplifier. If the circuit is interrupted while heavy currents are flowing, some of the surface metal will be vaporized and any precious metal plating will soon disappear. Therefore, it's a good idea to disconnect and reconnect one's speaker leads periodically to break the oxide layer and re-establish a good connection.

The materials used for contacts are a problem with relays, too. The only type of relay that is totally acceptable is a mercury-wetted one since its contact material is constantly being renewed. That's why db Systems has clamped turn-on and turn-off transients in its amplifier without using relays.

In five years of testing new and used equipment to a resolution of a few parts per million, we have not observed any distortion from metallic contacts unless they were so bad as to be intermittent, and so a little care is all that's required to assure excellent performance from these simple phono connectors. A firm connection is a good connection and, in fact, is "as good as gold." What gold buys in a given situation is increased reliability. That is certainly nice, but it is not essential. With proper care, you can achieve comparable performance with ordinary tin-plated phono plugs and jacks. -- David B. Hadaway (New Hampshire)

Ear Equality!

While we all know about Fletcher-Munson loudness curves, very little in the audio press is said about the effect of the ear on the testing of equipment. Our hearing has many variables, including dirty ears, the cut of our hair, the shape of our heads, and this is interrelated with our physical and emotional states.

A couple of years ago I decided to run a frequency response test on my ears at a hearing aid testing service. I discovered that my ears measured differently and were not flat. I would like to postulate the following and suggest a series of tests to include the "ear" in the testing procedure.

1. There is a great difference between the frequency response of different people's ears and this explains to some extent the differences they hear.

2. There exist absolute and relative differences in people's ability to distinguish distortion, differences in tonality, loudness and pitch. Because we are all not possessed with the same discriminatory abilities, some audiophiles do indeed hear things that others don't.

3. Our hearing is affected by whatever is going on inside and outside our heads so a stressful short hair person will hear differently when he is calm and his hair is long.

4. The fatigue threshold of ears is different. Some ears work better over the long term and some work better over the short and this affects the hearing experience.

5. There may be a correlation between the sonic personality of our ears and our preferences for equipment.

It seems incredible to me that audiophiles are arguing about what reviewers "hear" or don't "hear" when we know nothing about the reviewers' ears. I am sure there could be a series of tests that would show the differences in the hearing and discriminatory abilities of different reviewers.

Is it possible that Harry Pearson has a strident midrange in his left ear and a veiled lower midbass on his right? Could Peter Aczel have a liquid tubelike left ear and have an ultra-high-slew-rate-phase-coherent right ear ? Is it possible that we have been arguing not about the differences in equipment, but the differences in the design of the reviewer's ear?

I suggest the BAS take up the flag of pioneering (pionearing? -- HB) ear testing. Why not hold "Ear Test Clinics?" I think many of the members who have S-O-A equipment do not possess S-O-A ears!

-- Harvey Rosenberg (New York)

In the Literature

The Absolute Sound, No. 17

*Letters: Open season on digital, the Riggs-Shanefield articles in High Fidelity, and the B&O and Revox straight-line turntables; Kuby and Otala report audible veiling due to potentiometers.
*Cartridges (p. 19): A comparative assessment of 30 pickups, all mounted in the Yamaha PX-2 straight-line turntable, and concluding that: the optimum resonance frequency is 6 to 7 Hz; the only satisfactory MC step-up device is the Mitch Cotter Type L transformer; and impedance matching, ambient temperature, and VTA are critical. Pickups ranked in groups as follows: Category A - Mark Levinson, Koetsu, Grace F9E, Linn Asak, Dynavector Karat-Ruby, EMT, Van Alstine-Sonus Dimension Five, Grado Signature 3a Revised; Category B - JVC MC-1, Prestige, Koetsu Memnon, Dynavector Karat-Diamond; Category C - Grado FTE+1, Direct Sound, NAD 9000, Fulton Revised, B&O MMC 20CL; Category D - Supex SDX-1000, Yamaha MC-1S and 1X, Denon 303, Signet Mk IIIe, Empire EDR. 9; Category E - Sumo H, Sonus Blue-Gold H, Shure SC39ED, ADC XLM Mk III Improved; and Category F - Ortofon MC-30, ADC ZLM, Conrad Johnson Argent, Van Alstine-Grado F3E+1.
*Sneak Previews (p. 54): Spectral MS-One preamp (fantastic) and Sumo power amplifiers (terrific).
*Technocracy (p. 56): Interview with John Curl on the flaws of present digital circuits, the potential for improved analog recorders, and the problems with the relays in Spiegel's switchbox.
*The Music (p. 66): A report on Warner's decision to gut Nonesuch, plus reviews of lots of records.

Audio Engineering Society Journal, March 1980

*Detection Threshold of TIM (p. 98): Otala reports experiments showing that the audible threshold of slew-induced distortion is 3% instantaneous peak or 0. 003% averaged RMS.
*Bandwidth Necessary for Optimal Sound Transmission (p. 114): Reporting experiments which demonstrate that listening panels don't hear any difference between sharp-cut filters at 15 kHz vs. 20 kHz.

Audio, March 1980

*Borgia Column (p. 6): Parody of an "underground" review.
*Behind the Scenes (p. 18): Notes on products seen at last fall's AES Convention.
*Doug Sax (p. 30): Interview with the mastering engineer about the dynamic range of the disc medium, dbx encoding, digital, and speaker power-handling and efficiency.
*Another View of TIM (p. 39): Notes on realistic causes of slewing distortion, appropriate cures, and the benefits of high feedback.
*Selectavision (p. 44): Notes on videodiscs from RCA and others.
*Picking Capacitors (p. 50): Further data on why capacitors can yield audible sonic aberrations, with recommendations on making substitutions.
*Reviews (p. 64): Sony PCM-1 digital VCR converter (a rave review). Sansui BA-F1 power amp (an enthusiastic review with the usual stupid omission of any load impedances other than an 8 ohm resistor). Dennesen Soundtracktor phono cartridge alignment gauge (very accurate and reasonably easy to use). White 4220 octave-band, cut-only equalizer (well made, possibly useful for speaker EQ). B&O MMC 20CL phono cartridge (neutral sound, tracks pretty well).

DB, March 1980

*A special issue on cinema sound with articles on movie sound formats, Dolby playback, JBL's new theater speakers, the Comtrack quad system, and the synthesized special effects in Star Trek.

Gramophone (England), February 1980

*Quarterly Retrospect (p. 1247): Notes on some of the best recent discs.
*Better Disc Quality (p. 1308): Notes on direct-to-disc and digital releases.
*Reviews (p. 1314): Six magnetic cartridges (reviewer John Borwick damages the credibility of

the review by consciously using the wrong load capacitance for some of the pickups, notably the Ortofon Concorde 30). Sony TC-K65 cassette deck (quite good with ferrichrome and metal). Celestion Ditton 332 speaker (good at the price).

Gramophone, March 1980

*Sounds in Retrospect (p. 1454): Reassessing the sonics of recent discs.
*Reviews (p. 1460): Audio-Technica 1100 tonearm (low mass, damping in the lateral plane, well made, works well). Audio-Technica 6006 Safety Raiser tonearm lift (expensive, effective, reliable). Hafler DH-200 power amplifier (superlative performance, outstanding value for money). ADC ZLM-Improved cartridge (high compliance, smooth performance). Toshiba micro components (preamp and integrated amp okay, power amp excellent, tuner good except for its vulnerability to overload, cassette deck good).

Hi-Fi News & Record Review (England), February 1980

*Loudspeaker Research (p. 41): Two articles on independent uses of laser interferometry to study cone motion and cone breakup; e. g. , in the B&W 801.
*Equalizers in Doubt (p. 53): On why graphic equalizers really can't do many of the things claimed for them, such as room EQ.
*An Integrated Amplifier (p. 61): Part 2, the engineer's thoughts on preamp circuit design.
Reviews (p. 115): Optonica 7100, Teac TX500, an Philips AH180 tuners (each good). SAE 1800 parametric equalizer (well made, very flexible, performs well, but control labeling poor). dbx 100 subharmonic synthesizer (works as advertised, is effective with pop but of doubtful utility with classical, its value rests on individual judgment). Burwen TNE-7000 tick-and-pop suppressor (works quite well when correctly adjusted). Burwen DNF-1201 dynamic noise filter (measures well but judged to be of marginal value).

High Fidelity, March 1980

*Cross Talk (p. 22): Qs and As.
*Equipment Reports (p. 28): Apt 1 power amp (a uniquely thoughtful and original design, a truly useful overload indicator, lots of dynamic headroom, drives all load impedances cleanly, very potent in the bridged mode). Yamaha C-6 preamp (remarkably flexible taping controls, the tone controls are a parametric equalizer and are extremely useful, control labeling is poor, performance is excellent). Amber 70 power amp (clean and strong, especially in bridged mode). Hafler DH-200 power amp (clean, potent, no current-limiting, no frills, kit assembly easy). Cizek KA-1 mini speaker (superbly crafted cabinet, efficiency low, power-handling high, impedance low but uniform, forward imaging, crisp sound). Acoustic Research AR25 speaker (modest efficiency, high power-handling, smooth response, excellent value for money).
*How to Buy an Amplifier (p. 52): From Mike Riggs, a clear and thoughtful basic introduction which raises real issues and slays advertising myths with remarkable directness.
*Equalized Double-Blind Tests (p. 57): Dan Shanefield's controversial contention that most subjective differences among amplifiers are due to frequency response.
*Input-Output (p. 98): On the DeltaLab Acousticomputer, a very flexible and wide-range stereo digital delay, reverb, and special effects device.

Modern Recording, March 1980

*Reviews (p. 66): Toshiba PCM Mark II digital converter for VCRs (superb performance, audibly better dropout compensation than the older Sony PCM-1). Nikko Alpha VI power amp (expensive and powerful). Teac X-10R 10 1/2" open reel tape deck (dual capstan drive, bidirectional record/play, response very smooth and wide-range except for slight bass emphasis, high end is good to 20 kHz at 3 3/4 ips, accessory dbx unit available).

Popular Electronics, March 1980

*Stereo Scene (p. 14): On recorder-tape matching and Dolby HX.
*Reviews (p. 22): Pioneer SR303 reverberator (synthesizes reverb and mixes it into the stereo sound, works okay). Crown SA2 power amp (very powerful, 450 watts/channel at 4 ohms and 650 watts/channel at 2 ohms, IHF slew factor 2. 5, low-level crossover distortion noted at low

powers, sophisticated protection circuit combines indestructibility with high output current). Garrard GT350AP automatic turntable (clever scheme for adjusting overhang, controls operable with cover down, medium-light arm, good feedback isolation, good arm geometry, rumble only fairly low).
*Car Stereo Standards (p. 44): Details on the specification standards recently adopted by some autosound companies.
*Experimenter's Corner (p. 80): Some noise generators, pink and otherwise.

Recording Engineer/Producer, February 1980

*Tube Condenser Microphones (p. 82): The chronology of the Neumann line of mikes, with info on the physics of condenser mikes.
*Line-Level U47 (p. 94): Adding circuitry to produce a line-level output from the mike.
*Nonlinearity in Microphones (p. 102): Response errors, phase shifts, and high-level overload, leading to the idea of using B&K lab mikes for recording.
*Review (p. 110): Shure SM81 electret cardioid mike (splendid ruggedness and reliability, impressively neutral sound, unfortunately not available in omni form).

Speaker Builder, 1980 No. 1

*Ambience Reproduction System (p. 12): A clever and inexpensive approach to making speakers ideal for multichannel time-delay ambience system. Strongly recommended.
*Small Cheap Speaker (p. 18): On making an inefficient vented system.
*Diffraction (p. 28): How cabinet edges cause large irregularities in on-axis response, with sensible advice for speaker makers.

Stereo Review, March 1980

*Audio News (p. 20): Details of the new EIAJ standard for digital converters to be used with VCRs.
*Audio Basics (p. 24): Cassette vs. open-reel.
*Tape Talk (p. 26): Background on Connoisseur Society's superb InSync prerecorded cassettes.
*Tech Talk (p. 33): Why metal tape isn't necessarily dramatically better than cheaper formulations.
*Test Reports (p. 34): Phase Linear (by Pioneer) 7000 cassette deck (expensive, superb, microprocessor tape-matching works fine, ultralow flutter). JVC QL- F6 turntable (incorrect arm geometry, high arm mass, ineffectual damping, ambiguous instruction manual, otherwise fine). Sennheiser HD420 headphones (light, comfortable, unusually smooth response). Ohm M minispeaker and N woofer commode (satellites have 3 ohm impedance and 14 kHz peak, woofer has 70 Hz peak, system is otherwise fine). Sony V55 receiver (lightweight switching power supply, rather good digital tuner, decent preamp, peculiar treble control action, power amp good except for 3. 5 ampere current limit).
*Going on Record (p. 59): On Warner-Elektra's massacre of Nonesuch.
*Buying Cassette Decks (p. 61): Information and advice.
*Tape Recording Vocabulary (p. 68): Translating the jargon.
*Tape Matching (p. 74): Picking a tape purely by ear, one man's approach.

-- Peter Mitchell

February Meeting Summary

The business portion of the February meeting was kept short since two guest speakers were each expected to give full-length presentations. The one announcement of general interest was of the release of two locally-engineered pipe organ records, Volumes One and Two of the }look Documentary Series, recorded mostly by David Griesinger. These are available to BAS members through Scott Kent, who described them as having ample deep bass content, but as being unusually easy to track with normal phonograph cartridges. This is due to special processing of signals which correspond to vertical information on the disc.

Part One: Revox

The principal guest speaker was Renaud Delapraz, Technical Manager of the Revox Division, Studer Revox America, Incorporated. The Studer organization is certainly best known for its tape recorders such as the fully professional Studer machines and the "semipro" classic Revox A77. Willi Studer founded his company in 1948 in Switzerland for the manufacture of special purpose oscilloscopes. Their first recorder appeared in 1950, the "Dynavox" tape recorder intended for home use. About a year later, they marketed the first Studer professional recorder, the Model 27, and began to use the "Revox" trade name on their new home model, the T26. In 1954 Studer introduced the Revox "36" series. These included the Models A36 through C36 mono recorders and continued with stereo models through the G36, all with vacuum tube circuitry. These were superseded by the solid state Model A77 in 1967. By that time, additional production facilities had been opened in the Black Forest region of West Germany and the company had begun to make high fidelity amplifiers, mixer consoles for broadcast and studio use, and multitrack tape recorders. FM tuners and language laboratory equipment soon followed. In general, the same engineers have designed both the professional Studer and the consumer/semiprofessional Revox equipment.

The Revox product line has been completely revised over the past few years, so there are "up-to-date" products comprising a complete home stereo system minus a cassette recorder. A few items, notably accessories, are made for Studer by other manufacturers. The general standard is unmistakably "luxury class" and the equipment is priced accordingly.

Since Revox is known in the U.S. almost entirely for its A77 and B77 recorders, Mr. Delapraz's presentation emphasized the Revox turntables, FM tuner, and receiver.

The Phonograph. There are two models of the Revox phonograph: the B790 for a list of $899 and the somewhat simpler Model B795 at $599. Both have quartz-crystal controlled, direct drive motors and run at 33.33 and 45 rpm. The B790 features adjustable speed over a '7% range and a four-digit LED indicator which displays the measured rate of rotation, but otherwise the two models are the same. The tonearm design provides straight line (tangential) tracking which maintains a constant lateral tracking angle and thereby reduces certain forms of playback distortion, since record masters are always cut with tangential heads. Distortion due to skating is said to be eliminated by this design as well.

Another goal of the Revox design was for the moving system to have very low effective mass. This was achieved by hanging the cartridge from a horizontal rail system. The "arm" on which the cartridge is mounted is just four centimeters long and its dynamic mass is claimed to be only 3.75 grams.

The Revox system is sold at present with either an Ortofon or an AKG cartridge. Users can buy a kit which facilitates the installation of other cartridges. Tests by Revox have shown that the fundamental resonance of this arm with most cartridges lies between 10 and 11 hertz where it would not readily be stimulated either by record warps or recorded audio signals.

The arm and cartridge are guided along the rail system by a small motorized carriage. A block assembly which contains these rails and the carriage, arm, and cartridge is then pivoted in much the same way as a conventional tonearm. To put on or to remove a record, one sets the block assembly into its "at rest" position out of the way of the platter. And to play a record, one sets the assembly over the record's surface into a detent with a claimed repeatability of ± 0. 01 mm! This starts the turntable motor, but the stylus stays lifted above the record's edge until the user presses a button to lower it. While it is lifted, cuing can be accomplished with two other push buttons.

As a record is played, the carriage is propelled forward in response to signals from an ingenious optical and electronic system. Time constants of the system are chosen so that the arm moves smoothly, and so that an eccentric groove is tracked about in the center of its range of excursion. (The back-motion is absorbed by the free-swinging arm, not the servo.)

(Since the arm is so short, and since it follows eccentric grooves and record warps as a conventional arm would, it seemed logical to some that audible distortion induced by geometric errors would be exaggerated in this system.) Mr. Delapraz acknowledged that in severe cases of eccentricity this loss of ideal tangency can result in audible distortion but, he said, the inevitable wow would be far more disturbing and the wow would be roughly comparable in any conventional tonearm system. Likewise with the vertical tracking errors which occur when records are warped: it is claimed that a three-millimeter vertical warp (the thickness of two pennies) creates less vertical tracking error than is commonly encountered anyway due to the discrepancies among the 15° to 20° vertical angles of various cutter heads in use during the past half-decade; again, the resultant "warp wow," which Mr. Delapraz states is not much greater with the Revox design, would probably be the source of most of any audible disturbance. (But note that the vertical error due to the discrepancy between cutter heads and cartridges is constant throughout a record and is likely to be blocked out mentally; warp wow is not. It's sometimes possible to compensate for the former in conventional arms by adjusting the height of the arm or the mounting of the cartridge. -- HB)

The little tonearm has only one point of contact with the motorized carriage, a jeweled bearing which forms the lower pivot point. The vertical shaft of the tonearm is held captive by the field of a small, oblong, concave, permanent magnet. This magnet keeps the arm from rocking laterally, yet adds no friction to the system.

The mechanical suspension for the overall turntable system has its basic resonance at or below 3 Hz, rendering it quite insensitive to feedback even if it is set directly atop a subwoofer cabinet. This was demonstrated.

The dustcover, which is transparent and removable, is contoured such that it can be opened or closed even when the turntable is placed flush against a wall, and is sprung to "stay put" in any position. All operating controls are outside the locus of the dustcover.

Electronic components. The Revox B760 tuner, costing $1,649 in the U.S., is tuned with quartz-locking digital synthesizer circuitry. Its station roster is in 25 kHz increments, said to be necessary in Europe for accurate tuning of certain Armed Forces Radio and Television Service stations. It has a CMOS memory for storage of up to fifteen station frequencies. An antenna rotor, available as an accessory for $649, can also be controlled by this memory facility. Internal nicad batteries allow the stored information to be held for as long as a year when AC power is disconnected from the unit. An optional Dolby decoder module ($130) can be installed by the user. Deemphasis can be switched to 75, 50, or 25 microseconds.

The Model B780 receiver, at $2,699, incorporates basically the same circuitry as this tuner plus that of the B750 integrated amplifier which is $999 as a separate product. The amplifier (or amp section) delivers 75 watts per side into an 8-ohm load and is not specified for any other load. There are some extra refinements in this receiver: the tape outputs can be assigned to a source other than the one being fed to the main amplifier. The sources selected are displayed with front panel lights. There are some automatic scanning modes of FM tuning, or the station frequency can be entered on a small calculator-style keypad. The memory holds nineteen station frequencies and antenna headings and signal strength is shown directly in μV and dBf on a front panel meter.

Tape recorders. A few years ago, Revox unveiled a product which had long been the subject of speculation by many audiophiles -- the successor to the Model A77 tape deck. The Model A77 had been sold in many special versions. Some of those which are not available as versions of the B77 are still sold. Apart from the widespread use of integrated circuits, the Model B77 has audio circuitry rather like that of its predecessor. Amplifier headroom has been distinctly improved in the new model, although this could have been done easily enough in the old one years ago.

The reel motors are controlled by solid state switching devices rather than by mechanical relays, and reel motion sensing makes tape damage due to wanton button-pressing far less likely than on the older model. There is more space between the head block and the control panel which makes editing a little easier. The machine has a socket for a varispeed accessory as well as for remote control. (An optional fourth head is available to handle slide-sync or other control signals which are recorded on a third track separate from the audio. The fourth head is a combined erase-record-playback head, so the extra track cannot be precisely synchronized to the audio tracks. Its frequency response is also limited to that which is useful for its primary purposes.)

Studer intends to produce digital tape recorders -- the use of microprocessors in their newer professional machines is already quite advanced -- but they are waiting for the initial battles over standardization to be fought out, which may well take years. (See Stephen Temmer's talk at the March BAS meeting in the next issue of The Speaker.)

Some personal observations. The Revox tape decks are simply no longer a bargain on the American market. The basic selling price, currently a buck under $1500, is what the fancier model A700 used to cost. A700s cost $3000 these days. Not as popular among audiophiles, the A700 has three-speed electronics and transport, servo controlled tape tensioning in all modes including fast wind, tape length counter, and a built-in four input mixer. It was not shown at the meeting.

The manufacturer's price for these machines has not risen so much as the currency relationships have changed. A dollar today buys only half as many Swiss francs as it did in 1974.

In spite of the enormous increase in price, the B77's performance is not really much improved over the A77's, nor does it entirely live up to what could be expected from its manufacturer. On a deck with such high quality audio circuitry, heads, and capstan drive, it's unfortunate that there are not really accurate, peak-reading meters and some form of tape tension control more advanced than the big reels-little reels switch. The first is found in many cassette decks, and the last is available in a number of tape decks including the cheapest Sony. A good tape tensioning system reduces flutter, head wear, dropouts, and pitch variation in the play and record modes.

Mr. Delapraz was critical of dual-capstan drive, which stretches the tape to provide tension because the downstream capstan runs slightly faster than the upstream one. This argument, however, confuses the issue; tension and elongation are directly related, so any tape drive which provides tension also produces a corresponding amount of elongation. The question is, how much of each should there be? Tape transports fall into two general categories in this regard, because the requirements for 1 1/2 mil tape, used for most studio mastering, are different from those for 1 mil tape, the most popular thickness for amateur or semi-professional recording. The thinner tape conforms to the heads fairly well, and is more fragile than the thicker stuff, so it requires lower tension, and most home or semi-pro machines are designed for it. For the professional, the thicker formulation, with its higher maximum output level and increased physical ruggedness, will be the choice. Since the thicker tape is stiffer and does not conform to the head so well, and since a dropout can ruin an expensive take, most professional machines are designed to pull the tape down hard over the heads. If there are dual-capstan machines now being sold for home use that have as much tape tension as these professional machines, then they may have problems with 1 mil tape. Delapraz told a story of a dual-capstan machine on which repeated playings caused a loss of output of a 10 kHz tone. Unfortunately, this phenomenon is observable on all kinds of tape recorders; test tapes deteriorate at the highest frequencies after a few uses. Rewinding a test tape (instead of turning it over and playing it onto its original reel) will accelerate this process.

Revox doesn't seem to take part in the "features race" with their tape recorders to the same degree that they do with other components. Added cost is given as the reason, but when prices are rising so steeply anyway, one looks for improvements in value.

Revox sells a version of the B77 with built-in Dolby B circuitry for a $200 premium. This writer can testify from having owned a pair of A77s with Dolby B that the noise reduction is quite useful in live recording even at 15 ips. I do not understand Mr. Delapraz's claim that even Dolby A encoding "doesn't help the quality of the high speed B77." I have heard a similar claim made by a spokesperson for Ampex in reference to their ATR100. It was based on a confusion of reference levels and was not practically valid.

Revox specifications for the B77 are based on the use of Revox 621 tape, a special version of Scotch 250 on 1-mil backing and available only from Revox (at $36 per reel!). This tape has many favorable characteristics, particularly an ability to take very high midrange record levels, but even in the standard 1 1/2-mil version, it is notorious for print-through problems, particularly when used at 15 ips. This problem is too often overlooked, but what good is a 70 dB signal-to-noise ratio, for example, if the signal-to-print ratio of a tape is only 50 or 55 dB? One only slightly cynical answer: it makes the most-looked-at numbers on the specification sheet of both the tape and the tape recorder look better.

At any rate, suppression of print-through is one of the cardinal advantages of a good noise reduction circuit such as Dolby A. If standard speed (3 3/4 and 7 1/2 ips) consumer tape decks, especially quarter track models, are to be used for live recording, some form of noise reduction is needed for the sake of adequate dynamic range. The standards for "adequacy" have risen rapidly as more people have stereos that play much louder. Revox needn't endorse any particular circuit where so much useful development is going on: they could simply provide a break-point in the record circuitry after the preamps and level controls but before pre-emphasis and meter circuitry. This would have a function similar to the "preamp out/power amp in" jacks on many modern integrated amplifiers including Revox. It would permit correct and fairly easy connection to any desired external unit for noise reduction, equalization, limiting, or mixing. Admittedly, this feature (which appears on only two recorders that I know of, the German-built ASC 6000 which is roughly competitive to the Revox A700 at less than the price of the B77 and the Swiss Nagra IV-S which competes mainly with itself) is not idiot-proof. To use it one would have to know a little, and Revox seems increasingly to wish to remove such constraints, since that way they can sell more tape recorders -- not just to people who are willing to pay extra for a better machine, but also to those who want to pay extra for a more expensive machine.

Part Two: Leigh's Egg

The second presentation at the meeting was made by Paul Young, Manager of Engineering and Product Development for Leigh Instruments, Ltd. of Waterloo, Ontario (Canada). Paul was expected to demonstrate and describe the Leigh ECD ("Equalized Controlled Diffraction") loudspeaker of which he is the designer. However, the Studer people and their equipment left immediately following their presentation. Considering that the BAS meets in a very large, acoustically dead room, and that the rather diminutive ECD speakers were designed for use in home living rooms, this didn't hobble the meeting. From my own perspective, this may have been a blessing in disguise, since Paul's talk gave a refreshing, realistic view of what is considered when a product is designed and produced, and the fifty or so minutes left that night seemed barely enough for the enjoyable discussion.

Probably the first striking aspect of the ECD loudspeaker (for somebody who hasn't heard it) is its egg-like shape. This might appear to be nothing but a marketing stunt, but it actually resulted mainly from the designer's wish to suppress undue flexion of the cabinet wall(s), standing waves within the enclosure, and diffraction from cabinet edges. Paul pointed out that given the large relative area of cabinet surface to woofer piston, even slight flexing of the cabinet walls in conventional enclosures can generate appreciable distortion at certain frequencies. The ECD's molded plastic cabinet contains structural ribs to control the first problem and a predetermined amount of Dacron stuffing to help solve the second problem. The third problem, that of irregular reflection of sound waves from various parts of the speaker or of surrounding objects, is ameliorated by the smoothness of the "Egg's" outer surface, by its small size, by the use of a felt "blanket" around the drivers, and by an overall design which allows free-field, midroom placement of the speakers if desired. (It is suggested that such placement gives superior stereo imaging and cleaner transient response since significant early reflections from nearby walls do not occur.)

The "Egg" was designed to compete in quality and price with the better British "minispeakers" such as the Rogers LS3-5A. The Leigh is a small, two-way system, less than a foot in diameter and sixteen inches tall, with a system resonance at 60 Hz down roughly 3 dB at 50 Hz. Its price of $475/pair includes comprehensive mounting accessories: two weighted metal floor stands, a pair of sculptured plastic wall brackets, special internally prewired chains for hanging the 12-pound speakers from a ceiling, and all necessary connectors and instructions.

Separate sets of input contacts are provided at the top and bottom of the cabinet. (On the top, these are accessed via a "stereo" phone plug with a threaded collar which secures it to the speaker. On the bottom, there are color-coded banana jacks for the same purpose. -- HB) The three contacts in each set provide two electrically distinct ways of getting the drive signal into the crossover network. If the speakers are to be mounted near a wall where midbass acoustical energy would be reflected and -- ordinarily -- cause a hump in the response curve, a calculated rolloff can be selected to compensate. This rolloff can be "unselected" later by switching input contacts if free-field placement is chosen.

The crossover itself is accomplished by two 12 dB/octave Linkwitz-Riley filters in series, each tuned to 2.2 kHz. The resulting very steep crossover slope leaves little overlap between the ranges of the drivers, minimizing any possible interference between them around the crossover, and the phase characteristics of the filter are such that the drivers remain in phase at all frequencies. The exception is a slight overall retardation of signals to the woofer, which is compensated by the tweeter being set about 1 3/4 inches back into the cabinet face. Voice coils change their impedance when they get hot, and a multisection filter helps isolate the power amplifier from these changes in impedance. The crossover circuit contains self-resetting circuit breakers for each driver.

The system is designed to produce forward motion of the corresponding driver when fed a positive-going impulse. This is motivated by the view that correct "absolute" polarity gives an audible advantage in the reproduction of transient signals. (I remember the attempt to demonstrate this phenomenon with recorded applause at a past BAS meeting where Paul Young appeared with applied mathematician Stanley Lipshitz. But these people are also devotees of the Blumlein technique of stereo recording, with which most applause in a real concert situation will be picked up in reverse phase.)

Paul promised to make some early production samples available to A1 Foster for critical testing, so perhaps we shall soon be able to read a review of this fertile design concept.

-- David Satz (Massachusetts)

A BAS User's Report

The Acoustic Research AR9 Loudspeaker

An Evaluation

Alvin Foster

It is difficult to evaluate a loudspeaker that is too tall and unsightly to fit into any living room decor and so expensive that the mere mention of the $750-each price to a nonaudiophile is enough to convince him that you have lost your marbles.

I have had the Acoustic Research "flagship" in my home for over a year. What follows is my account of living with their top-of-the-line loudspeaker -- physical description, placement experience, measurements, and how they sound.

Physical Description

The cabinet dimensions of the AR9s, 53" x 15" x 16", and their weight, 130 pounds each, do not sound like much, but when you place a pair in a typical living room they dominate it. Located on the bottom third of the cabinet and protruding about an inch from each side are two 15" x 16" grill panels. The panels resemble legs or feet and suggest the units will walk out if they do not like your living room accommodations.

Cabinet Construction Quality

After one year my first pair of AR9s separated at the seams. Specifically, the glue in the top quarter of the cabinet let go. This led to the formation of tiny air gaps along the top and side seams. The separation occurred gradually, however; I was not particularly aware of the defect until I noticed air escaping from the seams while playing the cannon blast in Telarc's 1812 Overture (DG 10041).

To verify my misgivings, I fed a 30 Hz sine wave into the speakers and held a wet finger close to the seams. As expected, the gaps were real, and in one of the speakers the bass quality had deteriorated.

At times like this it's good to know you are dealing with a company that has been in business for twenty-five years. AR speakers are covered by a full, five-year warranty which even includes shipping costs both ways. AR will pay all costs associated with repairing or replacing a defective driver or the complete speaker system if it fails in normal use. After a phone call to the factory, the offending pair of speakers was removed and a new set was left in its place. An AR representative has informed me that my speaker was from a defective lot which has been recalled.

This unfortunate experience did give me a chance to compare the older AR9s, which I received about two months after their introduction, with a new 1980 pair. The earlier model had several deficiencies which have been improved upon to one degree or another in the newer version. It buzzed and rattled at several frequencies below 100 Hz when a sine wave was fed to the speaker. Its grill cloth mounting hardware was flimsy and eventually the large frame holding the front grill cloth began to distort. AR assured me that the minor warping that resulted would not interfere with the speaker's dispersion. The three frequency contouring switches in the older model did not feel firm although they never failed to operate properly. The original speakers also lacked the black foam which now surrounds the upper-midrange speaker and the tweeter. The quality of the wood veneer finish on the first speakers was no more than fair. The second pair is only slightly better. Both, of course, include the 3/4"-thick felt pad, or "Acoustic Blanket," around all the forward-facing drivers.

Inside the AR9s

Another slight defect in the original speakers gave me a chance to explore the interior structure and components of the system. A speaker terminal on one of the cabinets became loose after six month's use. To retighten, I removed one woofer which revealed two separate crossover networks. One section, presumably for the two twelve-inch woofers, covered the entire bottom of the cabinet. It included three huge capacitors and four hand-wound copper coils.

Located on the back panel, above the terminals, is the second crossover section. Again, capacitors and hand-wound copper coils abound, but of somewhat smaller size.

In the upper two-thirds of the cabinet are independent enclosures for each of the three drivers. The lower-midrange, an eight-inch unit, crosses over at 200 Hz and is housed in a long, hardboard-like tube stuffed with fiber glass. The enclosures of the dome upper-midrange unit -- 1200 Hz crossover -- and the 3/4" tweeter -- 7 kHz crossover -- are buried beneath the fiber glass which fills the top half of the cabinet.

Placement Experience

My basement listening room is approximately 24' x 13' x 7'. At first, as suggested by the manufacturer, I kept the backs of the speakers within two inches of a wall. After six months my opinion of the 9s began to take shape. In most locations the sound was only acceptable, but in some it was good. I settled on placing them against the short wall, 8. 3 feet apart, and seventeen feet from my listening chair. Both speakers were angled slightly toward the chair.

They excelled in their ability to extract "air," or ambience, from recordings and this contributed to an increased sense of the hall. The lateral, or left-to-right imaging was also exceptional. I credit this to the vertical alignment of the drivers. The low end, however, always seemed overpowering. I have always found this to be true in my room with speakers designed for playback in the real world, and eventually I discovered that a 48 Hz room resonance accounted for this low-end perturbation. Well-designed speakers with real output in that frequency range will excite that room mode. I solved this problem by constructing an inverse filter to remove the peak, but I could not seem to get a sense of front-to-back depth from the speakers; I couldn't discern the location of the recorded instruments in space, one in back of the other.

I remained disappointed until I received the Carver C-4000 Sonic Hologram/Autocorrelation Preamplifier. With the aid of the Carver, I discovered what was wrong with my room-speaker combination and how to cure it. The lack of depth and only acceptable stereo image stability were the result of too many early reflections. This became obvious with the Carver C-4000 because the same gremlins reduce the effectiveness of the Sonic Hologram Generator.

With conventional speaker placement, some of the sound waves from the mid- and high-range drivers move across the front of the speaker, and when they reach the rear and/or nearby side walls they are reflected back at the listener. The precise effect depends on the distance, and absorptive characteristics of the nearby walls and surfaces, but in general, these waves of sound which reach the listener at minute intervals "blurr" the sound.

The very earliest reflections, caused by speaker moldings, mounting screws in the drivers, and so on, are suppressed by AR's Acoustic Blanket. However, according to Robert C. Kral (Speaker Builder Magazine, February, 1980), very early reflections are not audible in a real listening environment. The more damaging secondary reflections from nearby walls are audible, but most audiophiles permit them to go unabated and thereby damage the stereo image and frequency response.

My experience indicates that for best stereo imaging all reflective surfaces should be at least two to three feet from the drivers. This produces a delay of 4 to 6 milliseconds, enough to convince the ear that the reflected sound is just an echo and not a part of the primary wave front.

Even after moving my speakers away from the nearest wall, the reflected wave front of the midrange drivers was still too loud. As a result, my attention shifted to the adjacent wall for certain percussive sounds. Because the time interval between the primary and secondary waves was now sufficiently long, the reflected sound needed only to be attenuated to restore proper stereo localization and stability.

To solve the problem I got two one-inch-thick, cloth-faced, acoustical ceiling boards, three-feet square, and installed them on the side walls. I placed the boards slightly to the front of each speaker against the offending wall. I positioned them to absorb early reflections from both the lower- and upper-midrange drivers equally. The woofer and tweeter are not affected by the panel, each for different reasons. The sound from the tweeter is reflected from the glass cloth, and the longer bass notes pass through the panel.

I used Owen/Corning Glass cloth-faced boards and am completely satisfied. However, a variety of panels will do, including Sonex Noise Baffles, manufactured by Illbruck/USA, and Soundsoak Wall Panels made by Armstrong Corporation, Coral Gables, Florida.

To further attenuate the reflections, I angled the speakers toward my listening chair and away from the nearby side walls. A good test for finding the "acoustic" center of your speakers-room combination is to switch your stereo system to mono and then listen to the character of the "stereo" image. The sound should appear to be coming not from the speakers, but entirely between them. Returning the system to stereo will increase the stereo spread, but most of the musical information should lie firmly between the speakers.

Through many trial-and-error adjustments, my listening chair was finally relocated to about eight feet from the speakers, intersecting precisely the acoustic center of the room. This is not to be confused with the visual center of the room; my listening position is actually closer to one speaker than the other. And this is typical of many rooms. Remember, you do not listen with your eyes. The room boundaries, absorption characteristics, and so on, alter the ideal listening location from what would be considered the visual center of the two speakers.

The AR9s were transformed when I (1) moved the speakers to the center of my room facing the short wall, (2) reduced the distance between the speakers to four feet six inches, and (3) placed my listening chair eighteen inches from the rear wall. This is the current arrangement and it seems to be ideal.

In summary, I recommend you ignore the manufacturer's suggestion to place the speaker against a rear wall, and put them at least two feet away from any reflective surfaces. Until this is done, you have not listened to your speakers free of image-interfering reflections. Incidentally, if you move your speakers away from a rear wall, AR recommends a minimum distance of three feet to avoid the 200 Hz dip which these speakers and the Allisons are designed to avoid.

<u>Fusing the AR9s</u>

The speakers should be protected if driven by an amplifier capable of putting out more than 100 continuous watts. I recommend placing a four- or five-amp fast-blow fuse in series with the speaker input. My experience, and that of others, indicates that the ferrofluid tweeter is the most likely to burn out. The upper-midrange driver, which also uses magnetic fluid, is not nearly as fragile. AR informs me that a one-amp fuse is sufficient to protect the tweeter if you are inclined to high-powered sine wave testing.

While completing the measurements which follow, I accidentally fed a loud burst of random noise to both speakers. The right channel four-amp fuse blew instantly. The five-amp fuse did not. Terrified, I figured all my drivers in the left speaker were ruined; however, the units were in perfect condition.

Measurement

Because very few of us listen to our hi-fi systems in anechoic, 4-pi environments, I made a frequency response plot of both speakers operating simultaneously in my room (Figure 1). This type of measurement gives you an idea of how well the speaker interfaces with the room, and it tells you immediately if the manufacturer forgot to design for typical real-world listening environments. A listening room completely alters the original anechoic balance of the system, particularly in the lowest and highest frequency ranges. The test also exposes room problems.

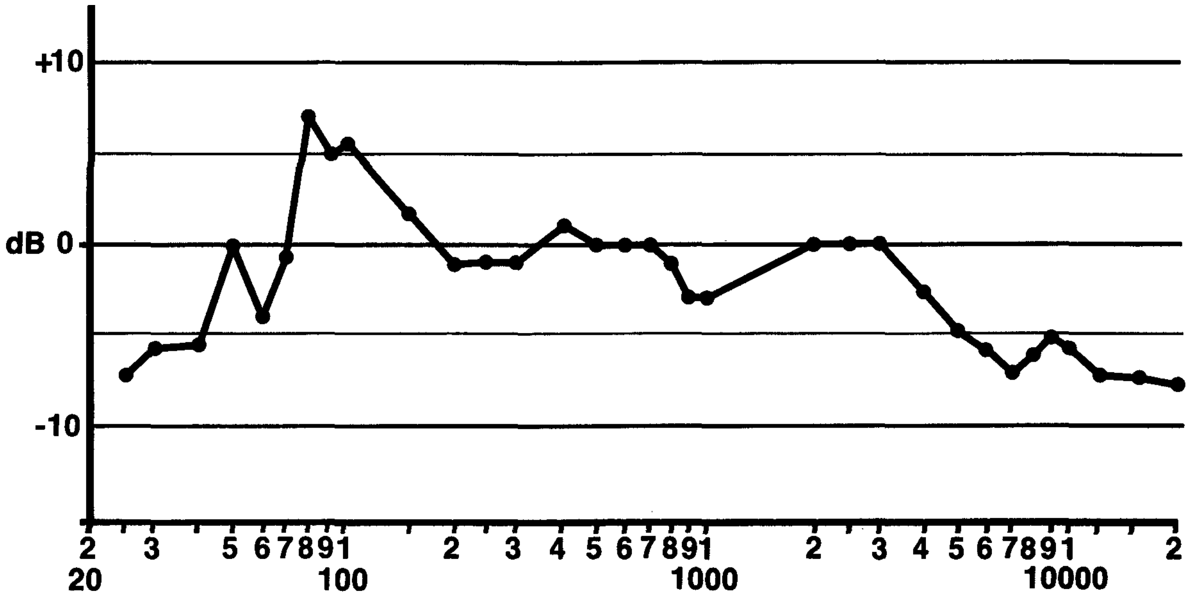


FIGURE 1

The frequency response measurement in Figure 1 was made with the B&K 4133 half-inch microphone which is flat on axis and down about 10 dB at 20 kHz for random incidence. I located the microphone above my listening chair and adjusted it to correspond to the height, and lateral and front-to-back position of my head. This technique, although realistic, produces frequency response plots that are typically not as smooth as those produced in an anechoic chamber.

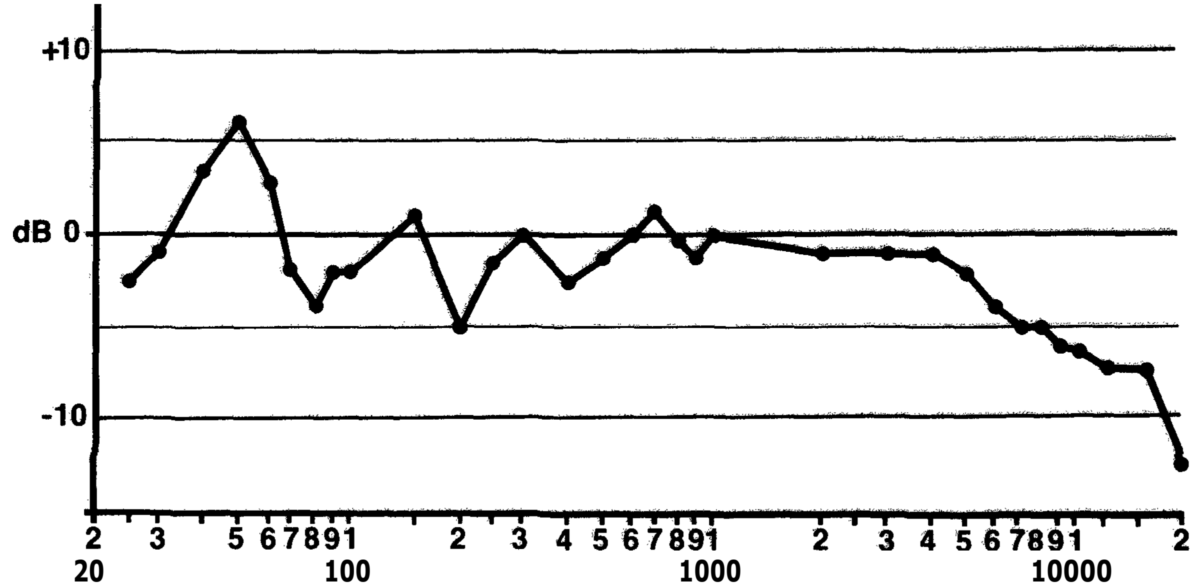
To complete the measurement, I used the General Radio 1554A third-octave analyzer. Each speaker was fed by a separate pink noise source. Because both speakers were already angled slightly toward my listening chair, the responses that are combined in Figure 1 are from off axis of both speakers. The high frequency roll-off in the graph is the result of room absorption and microphone characteristics. It is the response of the speakers in my room only and it would be different in any other room. The rear wall is located about twelve feet from the speakers. This produces the peak around 50 Hz since twelve feet is about half a wavelength for 50 Hz. The 200 Hz dip is caused by the distance of the microphone from the rear wall. (The latter is akin to the dip described by Roy F. Allison in the August, 1979, issue of Audio .) Overall, the room-speaker combination performed very well.

In Figure 2, the microphone has been raised to a height of about six feet. This simulates the frequency response heard by someone standing near the listening position. The results are mixed. The measurements indicate an overall dulling of the frequencies above 3000 Hz and a rear-wall-related bass boost around 90 Hz from the first vertical room mode.

In Figure 3, the microphone has been moved sideways to a listening position thirty degrees off axis. This produces a very slight dulling of the very high frequencies above 8 kHz.

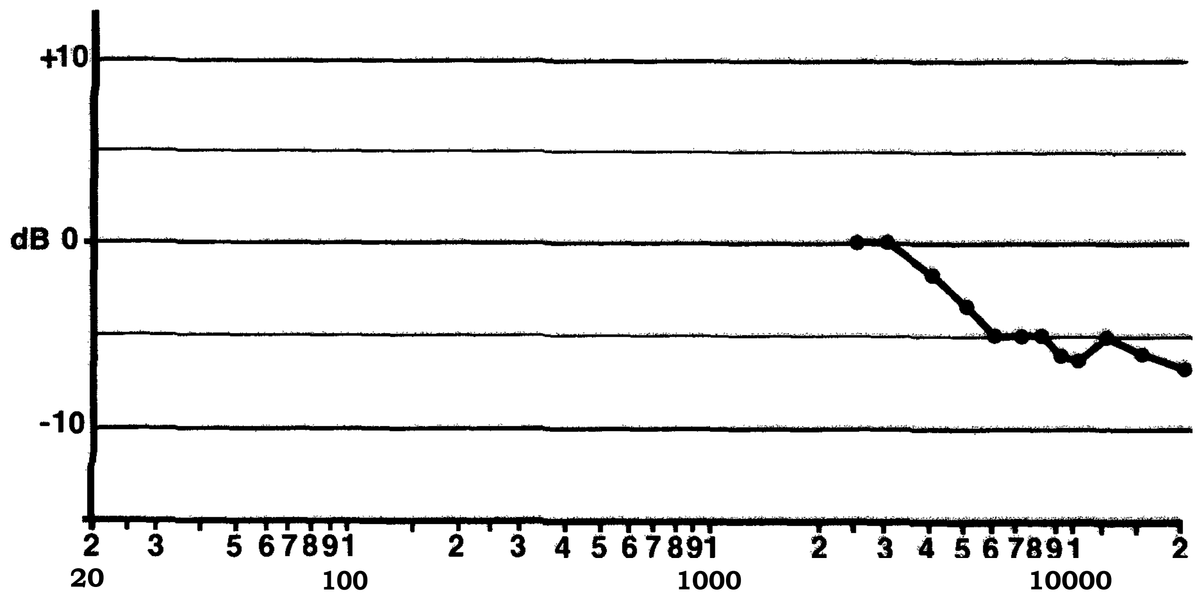
The AR9s are capable of producing loud and clean 124 dB peaks while reproducing average levels of 110 dB. The latter was measured with the built-in SPL meter of the Ivie Spectrum

Analyzer, The speakers also appear to be fairly efficient. With identical volume control settings the 9s sound noticeably louder than the 3As which have similar impedance.



FREQUENCY IN CYCLES PER SECOND

FIGURE 2



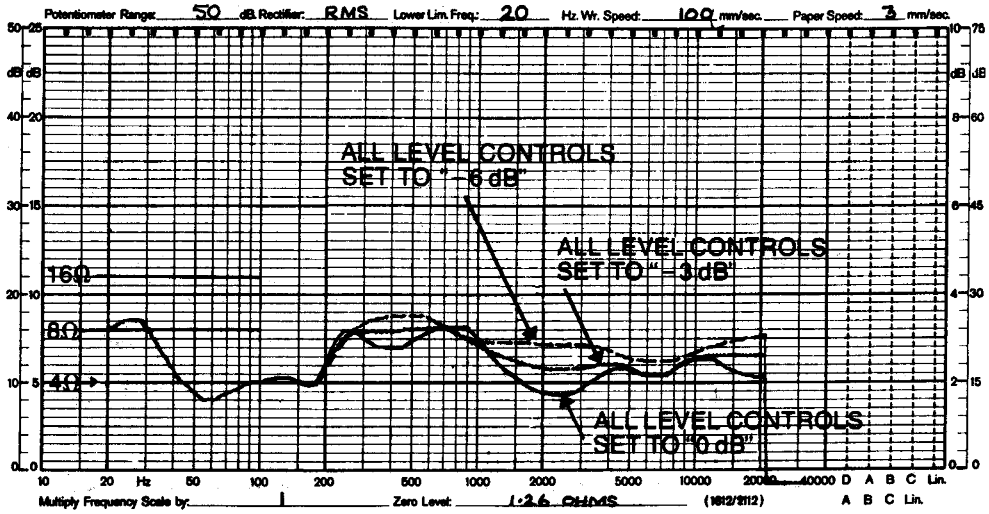
FREQUENCY IN CYCLES PER SECOND

FIGURE 3

Figure 4 is the impedance plot of the 9s. It was copied from the well-written owners manual. The booklet also describes how to identify a bad driver and so avoid returning the entire speaker to AR for service.

Twin-Tone Test

The two woofers were fed a 175- and a 100-Hz sine wave simultaneously. The distortion by-product was therefore 75 Hz. The wave analyzer revealed that this by-product was slightly less than 50 dB down or 0.3%. The exact SPL was difficult to determine because of the nature of low frequencies. Within two inches of one woofer, the SPL level registered 106 dB; however, with the microphone one meter in front of the speaker, about midway up the cabinet, the level was down by 13 dB to 93 dB.



Logarithmic impedance plots of AR-9 for the three indicated upper-range level-control settings.

FIGURE 4

The lower-midrange speaker was fed 1100 and 800 Hz at a level of 93 dB. The distortion was also 50dB down or 0.3%. Using the combination of five and three kiloHertz to test the upper-midrange unit produced 0.05% distortion; that is, the distortion by-product, 2000 Hz, was 65 dB down.

Because it is difficult to measure the tweeter with twin-tones and because the results are meaningless, I fed a pure ten kHz sine wave and measured the resulting second harmonic product of 20 kHz. The SPL was reduced to 73 dB one meter on axis. The distortion was a low three percent, inaudible.

(Editor's Note: The more usual method of measuring loudspeaker distortion, in which twin-tone IM tests are used at high frequencies and harmonic distortion components are measured at the low end, gives results that are both more relevant and considerably more accurate than the ones used here. Particularly suspect are the woofer test, which looks at only one of several possible frequency components and which was made in front of a cabinet with side-facing drivers, and the tweeter test, whose result is more typical of drive levels 15 dB higher than the one purportedly used.)

Subjective Impressions

Once you get tired of trying to blend the speakers into your room decor, you begin to notice that they have some sonic virtues. The speakers are among the best available and the only systems scoring higher are also much more expensive.

My first impression after installing them in my room, was that they sound like "large" speakers. This was contrary to my expectations after long experience with bookshelf speakers. The ability to recreate the sound pressure levels encountered in live concerts is immediately gratifying. They go louder and louder effortlessly. They can play louder than the Snell Type As and are significantly more efficient. Bass transients, which too often define the limit of typical bookshelf speakers, are reproduced effortlessly. High frequencies are neither harsh nor irritating.

My second impression is that the 9s extract more ambience from recordings. The differences are akin to my switching from an undamped to a damped tonearm. (See The Speaker, January, 1975.) The increased sense of air coupled with their fantastic image stability allows for more precise front-to-back instrument localization.

Hearing frequencies which are clean down to 25 Hz is another new experience. I agree with R. A. Greiner's statement that extension into the very low frequency territory may, in fact, result in a speaker that sounds slightly veiled or too heavy to inexperienced listeners.

Most speakers are not capable of performing as well as the 9s. In fact, only the best speakers designed with the living room listening experience in mind are successful at recreating the frequency balance inherent in a recording.

One disc which makes living with the 9s' four 12-inch woofers rewarding is the Telarc 1812. You have not heard the album until you experience it on a full-range system capable of reproducing realistic playback levels.

In summary, if the price of the AR9s doesn't choke you and the decor of your listening room can stand the imposition, give the speakers a chance. They certainly compete for the title of the best sounding full-range system available and they have the specifications to back up the claim.