THE B.A.S. SPEAKER

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The feature article this month is the first installment of the results of the BAS turntable/arm/cartridge clinic. We didn't include the whole thing because even after all this time there is still some uncertainty about the meaning of some of the data. We will resolve the ambiguities as best we can and finish the report next month. This time, you will learn some interesting and probably surprising things about frequency response, vertical tracking angle, flutter, tonearm geometry, and channel separation.

We also have a review of the Hafler DH-200 power amp, which by this and other reports looks like a real winner. Dan Shanefield returns after too long an absence from these pages with two brief notes, one on slow A-B comparisons and the other on equalizers. Dan has an article under a slightly pseudonymous name in the April issue of Stereo Review which deals with the same subject. And if any of you cheered when quadriphony faded from the scene, a careful and well-written piece by Massachusetts member Jerry Davis may change your perspective.

Considerable controversy has arisen over last month's review of the Carver C-4000 preamp. Along the way two questions of a more general nature have arisen. One is the possibility of a regular editor's column in the Speaker. The Speaker has traditionally been edited in a low-profile way to prevent its becoming the personal soap-box of the Society's officers or editors. This is a useful and proper tradition, and I intend to maintain it. As anyone who reads hi-fi publications knows, many people associate certain philosophical and technical opinions with the BAS itself instead of with its individual members, despite all our disclaimers. An editor's column would only worsen the problem by linking the opinions of the editor with the Society, which as you will see inside is the last thing the members want. So, for you lovers of paradox, here is the editorial opinion which states that there should be no editorial opinions.

The second, related problem is whether the editor can contribute to the magazine at all without being seen as representing the BAS. I have not decided this yet, although I hope the answer is yes. In the meantime, be reminded that the editor did not get his job by knowing more about audio than anyone else; it's just that only he was foolish enough to take it.

We round out this issue with a reprint from db magazine of an article on room equalization by Alan Fierstien, who is the scheduled guest for the April BAS meeting. One interesting implication of his piece is that the worse your listening room, the less appropriate it may be to use an equalizer to correct it. His talk should be an interesting one.

-- Brad Meyer

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For Sale

*Audionics PX3-2 power amp, 100 W/Ch, black, rack mount, without meters, recently factory-tweaked, mint, $325; Fisher MS- 100 walnut veneer speakers, brand new, $75; Automatic Radio solid walnut mini-speakers, new, $30; Dynaco OD-1 quadaptor, ambient sound for rear speakers without separate power amp, factory-wired, new, $25; antenna system consisting of Archer triple-driven six-element directional FM antenna, Winegard FM-340 rf amplifier (18 dB gain), Alliance Genie antenna rotor, with cable, used two summer months, mint, $75; dbx Model 128 tape noise reduction system plus adjustable compander, mint, $350; ADC Sound Shaper Two ten band/channel octave equalizer, mint, $200; Optonica RT 1515B cassette deck with APSS, black, mint, $210. L. Barry Tinkoff, P.O. Box 590, Fall River, MA 02722, (617) 673-6622 evenings.

*Audio Research D75A, silver, new output tubes, mint, $495 or trade for Hafler DH-200 or *Dyna* 416; Magnepan MG-2, white, with input jacks, mint, $435. P.O. Box 487, Gibsonville, NC 27249, (919) 449-6912.

*AR-10 speakers with matching AR metal stands, $500. (617) 485-2005 after 5 PM weekdays, anytime weekends, leave message.

*Shure V-15 III cartridge with both elliptical and spherical styli, $40, will ship prepaid UPS. Buford Reynolds, (615) 353-0994.

*Harman-Kardon Citation 11A preamp, $180; Citation 12 amp, $180; Sherwood S-3000III mono FM tuner, as is, $25; Sony ST 5150 AM-FM tuner, $115; Sony NR 115 Dolby adaptor, new, $60; DeCoursey electronic crossover, 50 Hz, 18 dB/octave, with summing circuit for center channel, $70; Linsley Hood harmonic distortion analyzer as described in Wireless World, commercially built by Teleradio Electronics in England. Don Konicoff, 120 West Palmetto Park Road, Boca Raton, FL 33432, (305) 392-6716 (7:30 to 10 PM EST).

*Audionics CC-2, immaculate, $350; Win Labs SDT 10-2C, newest model, brand new in original box, $375; Goldring G-900SE, $25. Bill Boswell, 3914 W. Robinson Street, Orlando, FL 32805, (305) 298-1640 after 5 PM EST.

*Revox A-77 Mk. III recorder, less than 100 hours total use, mint, $570; Dynaco 416 amp with C-100 power supply addition, almost new, $650 (will sell C-100 separately); Ampzilla, with meters, factory tweaked 1/80, $435; Sony Model 2251LA servo-controlled direct drive turntable with three new walnut tone arm mounting boards and Formula 4 silicone-damped tone arm in box, all for $240. Call Mike at (404) 321-4036 after 6 PM EST.

*Koss ESP-9 electrostatic headphones, excellent condition, $90. Hal Williams, 7305 S. W. 25th Avenue, Portland, OR 97219, (503) 246-1784.

Wanted

*Pair AR-3As, in good condition, will pay up to $250. King-wo Chiu, 32 Lyle Terrace, Malden, MA 02148, (617) 973-2080 (days).

*Sennheiser HDI 434 wireless headphones with SI 434 infrared transmitter; Niles Audio CPM-31 component patching matrix; Brazilian rosewood cabinet for Sequerra tuner; will buy, or trade items in my ad in "For Sale" section. L. Barry Tinkoff, P.O. Box 590, Fall River, MA 02722, (617) 673-6622 evenings.

*Norelco full-range drivers, 12-inch, 10-inch, and 8-inch, were commercially distributed in this country about seven years ago. Good condition, please. Bill Kalish, 565 Walnut Avenue, Redlands, CA 92373.

*Information regarding a small, decent-looking enclosure suitable for a preamp project. Damon Hill, 3261 Circle Oak Drive, NW, Atlanta, GA 30339.

Corrections

The Shure V-15 Type IV

In the December issue of the Speaker there are three short pieces on cartridge brushes and skating force. In the one by George Alexandrovich of Stanton Magentics, on p. 4, there is an editor's note which contains an error. The brush on the Shure V-15 Type IV is nearer the end of the arm than the stylus, not the other way around. This fact necessitates a modification of the method for checking the antiskating force outlined in the final piece. With the brush down, the stylus of the Type IV will be invisible from the front. The correctness of the antiskating force must therefore be checked not by sighting past the front of the cartridge to the stylus as the arm is lowered onto the record, but by aligning the side of the cartridge body with the edge of a band break or other identifiable feature of the record surface. If the antiskating is correctly set, the cartridge body will not move horizontally as the arm is lowered onto the record.

-- Brad Meyer (Massachusetts)

<u>The Daline Speaker System</u>

My notes on transmission line design in the October 1979 issue contained an error regarding the "Daline" design. The initial chamber behind the woofer is not supposed to approach the properties of an infinite baffle. Rather, it is a Helmholtz resonator. Its port exits into a damped, vented labyrinth which is tuned somewhat below the chamber's resonance. Details may be found in an article in <u>Hi Fi News,</u> November 1974. It should be required reading for speaker builders.

The benefits claimed for this type of loading are allowing the woofer to be effective to lower frequencies than would be possible otherwise, unique control of the woofer's resonance, and superior woofer transient response.

-- Carlos E. Bauza (Puerto Rico)

<u>The Holman Power Amp Tests</u>

I was disappointed by the editor's parenthetical comment about the audibility of some of the grosser forms of misbehavior during clipping in the meeting summary about power amplifier design (<u>Speaker,</u> October-November 1979, p. 40). The editor gave the impression that it was doubtful whether anyone had actually heard the effect that I was describing; that is, that the amplifier put out greater power as 120 Hz ripple than it was at clipping at 10 kHz with an 8 ohm resistive load.

In fact, the reason that I went looking for such a phenomenon was that its owner brought to my attention the fact that the amplifier went BLATT during a cymbal crash! In an amplifier clinic I conducted, such behavior was exhibited by a number of the amplifiers tested. One I measured produced 12 Vpp of 120 Hz when it was clipped, and as the power rating was 60 watts, and the owner used AR-3As, I believe it is likely that this phenomenon would be audible all too frequently.

-- Tom Holman (Massachusetts)

Sonic Hologram Follow-Up

<u>The Manufacturer Replies</u>

Regarding Brad Meyer's review of the C-4000: ouch! That was hurtful. You're wrong, Brad. A fully-functioning, properly set up Sonic Hologram is spectacular.

-- Bob Carver (Washington)

<u>A Possible Complication</u>

Because of a misunderstanding on my part, the sample of the Carver C-4000 preamp discussed last month by Brad Meyer was expected to be typical of production units. But in fact, it is a partially hand-assembled engineering prototype whose internal wiring is not entirely in final form, which could account for its reported hum level and possibly for other non-ideal behavior as well. Final conclusions must await both further measurements and extended listening to production-line samples of the preamp. Other questions which warrant consideration: how much is the holographic effect diminished by an acoustically live room, or by a phono pickup whose interchannel phasing varies with frequency? Stay tuned.

-- Peter W. Mitchell (Massachusetts)

A C-4000 Reviewer Replies to Brad Meyer

I take exception to several points you mentioned in your review of the Carver Sonic Hologram-Autocorrelation Preamplifier in the Speaker, Volume 8, Number 3.

1. The C-4000 you reported upon measured 13 to 15 dB more noise than your reference preamplifier. With your extensive audio background, you should have mentioned that the unit most likely was defective. In my years of testing, I have never measured any two separate preamplifiers that differed so widely in noise unless one was defective.

2. You stated that your review was "... the first one ever printed of the Carver Sonic Hologram." All other reviewers of the C-4000 had been subjected to the "spell weaving" power of Bob Carver, and by implication, our much more exhaustive reviews cannot be trusted.

I maintain that of all the reviews I have read, yours is perhaps the only one that may not have reviewed the Sonic Hologram. You spent a total of eight hours with the unit, scattered over one week. The other reviewers, instead, took the amount of time required to convince themselves that they had thoroughly listened to all the unit was capable of doing before they tackled a review that would reach thousands of readers. I know it is hard work, but this is the heart of a good review and reviewer.

Furthermore, I take exception to what might be considered an attack on my objectivity and the objectivity of other reviewers. I have lived with the unit for five months, long enough to have come to my own opinion. Other reviewers, Julian Hirsch, Larry Klein, and Robert Long, etc., have been in the hi-fi business much longer than you or I. Personally, I have faith in their objectivity surviving the onslaught of Bob Carver.

Although you stated that your review was "sketchy," this does not relieve someone with your reputation for thoroughness of the responsibility to insure that, at least, the unit you tested was operating to specifications. -- Alvin Foster (Massachusetts)

Another Reader's Reaction

Your review of the Carver Sonic Hologram in the -- uh -- "December" Speaker has prompted several thoughts. One is my long-standing opinion that The Speaker would benefit from the inclusion of a regular column of editorial opinion. The Carver "review" suggests a number of possible subjects including the possible interaction of publicity on supposedly objective reporting, room acoustics and the perceived quality of components, the relative merits of designs which seem to be innovative in a revolutionary sense and those which seem to represent the perfection or refinement of established products, and so on. Several of these are touched on in the Carver article and those who have heard you on Boston radio are likely to agree that you would have interesting and illuminating perspectives to offer on a regular basis. However, the so-called "first" "review" of the Carver is placing the wrong foot forward, to say the least. When I encountered its last two paragraphs I felt as though I had stumbled onto an experimental copy of Rona Barrett's Hollywood wherein they had included audio reviews. (I hope that comment doesn't slander Ms. Barrett.)

As I hope I can make clear, my complaint is with the tone of the writing more than with the perspectives -- prejudices, if you don't mind my saying so -- that are expressed in it. Since I, too, have prejudices about audio, let me set some of them out up front so the appropriate grain of salt may be applied to the comments which follow.

First, like you I am not a "genuine audiophile," but for different reasons. If I could buy "audio nirvana" for $49.95 I *might* make the purchase. On the other hand I might decide to buy a telephone, a movie, or some household appliance instead. Beyond a certain rather minimal point, my interest in audio must compete with other priorities. My $100-for-the-pair loudspeakers, the only ones I own, are satisfactory mostly in the sense that they do not offend me, nevertheless I have had them for four years or so now even though I could easily afford to replace them if I chose to do so.

Given such a "first" a logical "second" might be what I'm doing in an audiophile society in the first place, and the answer is that I appreciate and wish to support the systematic and rational approach to the hobby which the B.A. S. generally represents. This approach is also exemplified by designers like Tom Holman whose work I admire very much. I believe the cause of audio progress is well served by designers who take the trouble to identify and define real problems and then solve them systematically and imaginatively in their products. But that doesn't mean I would always choose such products for my own ideal system. The Apt preamp is not versatile by my definition, his tone controls are not adequate for my purposes, and I don't care for using a tone control for loudness compensation even if its curves are well chosen. Thus Holman's preamp, as much as I admire it, must compete in my priorities with other products which may reflect design philosophies I don't care for and which may be inferior in absolute terms if there are any absolute terms in this field. However, none of this is a reflection on Tom Holman or his products.

Third, about Bob Carver. He is a charming man and an effective advocate for his products. The meeting at which he described the Hologram was my first exposure to him and I was pleased to be able to join him at Al Foster's house that night as the process of setting up the Hologram was begun. If I ever return to the Northwest, I would hope our paths cross. I enjoyed his company.

At the same time when I see or hear or enjoy the company of a man with Carver's persuasive skills, little warnings sound in my innermost ear which say, "Yup, the Roeblings' bridge is a gorgeous, impressive, engineering feat, but is it appropriate to the Neponset River ?" And as it happens I feel it ought to remain where it is, spanning the East River. I don't think the Carver preamp is appropriate to my listening environment even though I judged its effect to be very worthwhile. Furthermore -- although my exposure to the Carver in Al's final setup is much too slight to venture this opinion -- I'm not at all sure that Al hasn't given up more in the sonic quality of his speakers than he has gained in spatial effects from the Sonic Hologram.

As for your noise figures, I am in no position to question them. Julian Hirsch's unweighted numbers (auto?) correlate well with your weighted numbers, but he drew a different conclusion. He regarded them as essentially unmeasurable and "impressive." For my own part, I can scarcely imagine owning a listening environment in which I could even begin to care about S/N figures like those you quote for the Apt.

As you can see, my own prejudices are in several instances similar to your own. It would appear that at least two of us have managed to survive the onslaught of the "spellweaver" as you have characterized him. And here we begin to approach the problems with your "review."

In the first two sentences of your penultimate paragraph you set yourself up as the final judge of the competence, taste, or psychological sturdiness of all the other reviewers and you cast Carver in the role of a cheap medicine man. The major U. S. reviews of the product in prototype or production form do not suggest to me that the reviewers were naive puppets predisposed to dance at Carver's beck and call. To the contrary, most auditioners seem to have approached the product with no small amount of skepticism and for that reason spent a great deal of time with the product before rendering their judgments. Yes, Larry Klein described himself as "mindblown" but what he describes in that context is his initial encounter with the Hologram in his home. With one channel unconnected at the input he heard spatial effects from a single speaker. Since he had not yet learned that the Hologram was feeding a signal to his other speaker, it's not surprising that he would be taken aback.

But what is the evidence that Bob Ajaye, Jerry Feder, Julian Hirsch, et al were bewitched by Carver beyond the fact that they drew a different conclusion from their experience than you did from yours? Must they be pilloried for liking what they heard?

And what does Carver claim for his product? That it produces an accurate image of the original sound field? No. He was quite clear that the Hologram only simulates a more convincing sound field and one that might very well differ materially from the actual recording situation. Beyond that he made no extravagant claims for his product and was quite open about his design considerations including his compromises. To my way of thinking Carver is much less deceptive than the many designers we've heard who have practically invented forms of distortion in order that they might reduce them to the vanishing point. It does not take a shiny pair of golden ears or a predisposition to hear or not to hear sonic anomalies to judge whether Carver's recent products meet their maker's product claims. And based on my listening experiences, I am not prepared to jump to the conclusion that Hirsch, or Al Foster, or Diversified Science Laboratories were incompetent to make reasonable judgments about what they heard.

"Carver is not a detail man." No, he is not. Next sentence: "The entire basic preamp is remarkably similar to the Apt. " Implication: Carver copied Holman. Good for him. Will you disparage the forthcoming "holograms" of other manufacturers and designers like Sound Concepts because the concept is not original with them as you seem to disparage Carver for following Holman's lead? Many monster power amps have improved on the Phase Linear, Joel Cohen may well "improve" the hologram idea, and numerous manufacturers are already expressing interest in the economical power supply circuitry of Carver's new amp. But Carver was there first. Why is it necessary to couple into one sentence that Carver is a "genuine genius" and -- referring back to the innuendo at the top of the paragraph -- a "brilliant and engaging salesman" as though the one canceled the other ? I think it's enough to point out the compromises in the products and to let others judge the man for themselves.

Finally, you draw the comparison between yourself and "genuine audiophile(s)." I don't buy it. Your standards of evaluation are eminently sensible and you have implied a nice distinction between yourself and many other audiophiles. But either way the standards are subjective ones and there's always legitimate room for disagreement.

In the end your opinions are no better than the intelligent, experienced, informed opinions of others in the field. There's no honor, no standard of fairness, no objectivity in packing all the opposition -- the defenders of Bob Carver's Hologram -- into a can and kicking it. I am happy to have your opinion of "sonic holography." I am sure there will be others equally provocative and useful soon in The Speaker, but let's not have the editor converting his opinions about products into rash statements about their designers and those who defend them. You have a higher responsibility.

-- Henry Belot (Massachusetts)

The Reviewer Responds

There are several issues in the previous reactions to my review which require an answer. The one that seems to have generated the most heat concerns my treatment of Carver himself and, by implication, other reviewers of the product. First of all, I obviously have no way of knowing to what extent, if any, other reviewers were influenced by Carver. I tried to treat the subject of Carver's effectiveness as a demonstrator and salesman without implicating the recipients of his presentations; this, I can now see, was naive, and to anyone who is offended by my implied assumptions, I apologize. I still maintain, though, that there is a reason for having brought the subject up in the first place. Details of what in drug research are called set and setting have genuine relevance to an equipment review. If, instead of buying a product at the store, taking it home, and listening to it, I have had the manufacturer himself in my house for many hours of listening sessions, experimentation, and demonstration of the product's virtues, then the nature of my encounter with the product has been altered. If I have become friends with the designer, or have entered into a business relationship with him, this too is relevant, and someone who reads my review deserves to know it. I will therefore try to include any such information in my own reviews, just as I tried to reveal other sources of my own prejudices when I wrote about the C-4000.

To those who feel that I have attacked Carver personally, let me clarify. Carver's persuasiveness and charm are in my opinion an asset. I certainly did not intend the phrase 'brilliant and engaging salesman" as an insult; quite the contrary. To sell one's product effectively is in this culture a time-honored tradition, not a form of offense. Although I too enjoy listening to Carver, it is part of my personal makeup to resist sales pitches, so I chose to leave A1 Foster's house after about two hours the night the C-4000 was being set up, and to listen to the device later at home. This may have something to do with the difference between my impression and that of others, so I offered it for consideration. This does not mean that I think there is anything wrong with such demonstrations, or that Carver is the only one who gives them. A surprising number of rave reviews (as well as, to be fair, some less enthusiastic ones) in magazines great and small, arise out of such personal encounters between manufacturer and reviewer. Everybody does it; I just wish they would say so more often.

As for the relevance, or lack thereof, of personal comments of any kind: most hi-fi equipment sold today is made in Japan and designed by committee. In the United States, Canada, and Great Britain there are still companies where one man is responsible for most or all of a design, and the products that result carry the imprint of the designer. I find this fascinating, and so I will probably continue to make occasional forays into personal description where I think it is appropriate. At least I am more aware now of the dangers.

There are questions about the hardware itself that have been raised which seem if anything more important than the foregoing. First, there is the matter of prototype versus production model. I reviewed a particular C-4000 under the firm impression that it was a production version. That it was not might be particularly relevant to the noise measurement I made, so I went to Natural Sound and checked a unit (#0589) which I was assured by the salesman was picked at random from a batch of production units. The production unit had, just as Peter Mitchell predicts, lower hum. At 180 Hz and in the 315 Hz 1/3-octave band (but not at 60 Hz), the hum was 4 dB lower than the one I originally measured, although the values were generally much higher than my Apt (#01332). The hiss, however, which was the audible problem with the C-4000 in my system, was identical in the two Carver units, and everywhere above 400 Hz the noise was from 9 to 13 dB higher than the Apt. For further comparison, I measured a Sony TA 2000 (#80418), which was 3 dB quieter than the Apt above 1 kHz.

There seems to be some misunderstanding about just what these measurements are, so let me elaborate. All of these figures are measured with the volume controls on the units all the way <u>down;</u> they represent the lowest noise that will ever come from the unit. All measurements were made <u>in situ,</u> with the units connected to a power amplifier of either 50k Ohms (a pair of Kenwood L-05Ms, in the case of the Carver) or 47k Ohms (Audionics CC-2, for all others) input impedance. Whether this noise is audible depends on several things: the voltage gain of the power amp, the efficiency of the speakers in the high frequencies, the size and reverberant characteristics of the listening room, and the background noise level in the room. I am using the Audionics amps in bridged configuration, which increases their voltage gain by 6 dB over stock; my speakers are quite prominent in the hiss range; my room, as documented in the original report, is quite live; and the background noise, when the forced-air heat is off, is very low. At Natural Sound, the Carver was feeding power amps with a voltage gain 4.5 dB lower than mine, driving Rogers LS3/5As which put out only 82 dB at one meter with one watt input, in a smaller but relatively dead room. In that situation, the noise was inaudible more than two feet from the speakers. Most systems, except those with very efficient horn-loaded speakers, will probably fall somewhere between these two examples, so you may have a problem or you may not. (Incidentally, taking more conventional IHF measurements from published reports, the Carver's two phono preamps seem to be 3 or 4 dB noisier than the Apt.)

There may be some reason to believe that the Sonus Blue cartridge, which has poor phase coherence between channels at high frequencies, is not suitable for use with the Sonic Hologram. For the record, I have also listened at Natural Sound, using an ADC ZLM, and to half-track, 15 ips tapes of the output of Foster's unit which I brought home. This should be an accurate enough method of transfer, as Foster says the effect can be reproduced through a cassette recorder. The results were not significantly different in either case.

Although what the C-4000 did to the stereo image in my listening room resembles what the preliminary instruction manual says it should do, it is still possible that I have not heard the Sonic Hologram perform to the limit of its capabilities. It may be that my room is just too reverberant to allow this, and there is some evidence that it is necessary to search for the exact physical arrangement of speakers and listening position more thoroughly than I did. But whether or not I have achieved the ultimate spatial effect, the frequency response problems remain. I am not persuaded by arguments that these will disappear in the exact position of maximum imaging; the problems occur over the entire frequency range, not just one part of it, and moving the listening position a few inches is just not going to make them go away. What will be interesting is to see whether the frequency response problems are inherent in the holography concept, or whether we can have one without the other. -- Brad Meyer (Massachusetts)

Another Image Enhancer

My wife and I have recently had the pleasure of listening to the Sound Concepts answer to the Carver Sonic Hologram Generator -- tentatively named the Image Restoration Control -- and were sufficiently impressed to want to write about it. Since the basic principles of these devices have been covered in detail in earlier issues of the Speaker, I shall confine my discussion to subjective impressions gained from an all-too-brief two-hour listening session with Joel himself, and to the special features found in his little box.

As those who have listened to such devices have mentioned, these devices, when properly used, broaden and deepen the sound stage, sometimes to an astonishing extent. For instance, the Prokofiev Fifth Symphony, second movement, as realized by Tomito on Bermuda Triangle, not only produced a sound stage about twelve feet wide and ten feet deep but also was able to place sound images behind the listener -- all this from two speakers placed about four feet apart and about six feet in front of the bewildered (but delighted) audiophile.

But what about normally recorded classical music ? Here the stage-expansion effect seems to be generally much more natural so that keeping one's speakers reasonably close together results in a natural sound stage. What comes through very nicely is a greatly increased sense of ambience, of being in the hall where the music was recorded. Joel's device makes at least as much difference to time-delayed material in this respect as time-delay does to straight stereo. However, with this device the ambience effect is primarily confined to the sound field to the front and sides of the listener unless, of course, one adds in time-delay. What is fascinating is that even if one is listening off-axis, there is still some of the ambience effect present -- something like a mild time-delay or Hafler effect. Furthermore, if two people are willing to listen one behind the other in close proximity, both can obtain essentially the full effect.

But, comes the question, even if the effect is very nice, what price must be paid? Monetarily, for Joel's device, under $200 and a brief wait until he has it on the market (May 1 ± two weeks). In terms of convenience of set-up and flexibility of listening position, there are some very definite restrictions. The speakers need to be at least one foot from all reflective surfaces, and the listening position must be equidistant from both speakers (and preferably reasonably close to them). Because of a special continuous control on Joel's box, unlike Carver's two-position switch, one is not confined to a specific distance from the speakers, but can simply dial in the angle one makes with the speakers. Signal-to-noise for Joel's device is, worst case, 75 dB, unweighted. Harmonic distortion is .2% worst case, while IM distortion is .15% worst case.

Bad recording techniques can cause problems. If there is no ambience to speak of in the recording, then the device will add little in that area. In fact, for mono signals Joel's box does nothing at all. Spot mikes whose gain is varied or soloists who move can cause an increased shift in instrumental location as compared to when the device is not in the circuit. In order to correct for this problem and for the varying degree of ambience in recordings and to prevent a kind of hole-in-the-middle, out-of-phase condition which could result if there is too much injected signal for a given recording, Joel has (in contrast to Carver's two-position switch) provided the listener with a continuous control which allows the listener to optimize the amount of injected signal. Based on my experience with the control, I find it to be a most valuable feature in the all-too-real world of modern recordings. This was especially evident in a Jean-Pierre Rampal recording of Mozart flute sonatas during which he was apparently dancing around while playing. With the control one was able to optimize the dance-to-space ratio according to one's own tastes.

One area as yet unmentioned is the important one of frequency response. With Joel's device there is no change in frequency response and no comb-filtering effect as far as the central image is concerned (he is hoping to patent the circuit which does this). In terms of overall frequency response, there is a slight change which can be best described as a slight high-end rolloff starting in the mid-treble region. The effect, however, seems perfectly appropriate in light of the greater sense of the original space which is produced. In Carver's device there is some deliberate contouring of the frequency response which was evident when Joel played us a tape comparing the two devices. This was a brief listen, and hence what follows is highly tentative. This is especially true in light of a subsequent conversation I have had with Al Foster, who asserted that everything has to be correct within a few inches when using Carver's device or it is possible to obtain a spatial effect but with frequency response errors. The two devices seemed to have essentially the same spatial effect on the signal source used -- Fennell's Macho Marches. Carver's device had a definite mid-bass hump and a peak somewhere in the 8 to 10 kHz region (and a low-bass rolloff, I've been told by several other members of the BAS). The result was that Joel's device, under those conditions and with all the caveats mentioned previously, sounded more natural both on and off axis.

In addition to the two continuous controls previously mentioned, Joel's box, in its present form, has two switches, one a bypass and the other a headphone switch. When the latter switch is engaged, the injected signal acts to reduce separation for headphone listening. Unfortunately, we did not have time to sample this feature. According to Joel, the effect is nice but not nearly so dramatic as the other mode is with speakers. In fact, he is debating whether to include this feature because of the added cost and complexity. As a teaser I can also say that he is thinking of adding another feature, which I am sworn to secrecy about, that people will find adds to convenience and ease in using his device.

All in all, land my wife found it a most fascinating afternoon and are eagerly awaiting Joel's marketing of his device. In the Sound Concepts tradition, it should be a natural.

-- Jim Richardson (Massachusetts)

Greiner on Speaker Distortion-A Query

In the October 1979 Speaker, Dr. Greiner states that woofer distortion seems to have little correlation with type of loading, except for transmission lines, which are much worse. What does he mean? Besides their cost, bulk, and inconvenience, can he show convincing evidence that translines perform inherently better or worse than other types of loading, and why?

Now that speaker building is an up-and-coming thing, enlightening comments on this will be surely appreciated by many hobbyists. -- Carlos E. Bauza (Puerto Rico)

More From the Caribbean Contingent

Re the Speaker, Vol. 7, No. 10, there are no "Sound Guard woes" if it's used right. (1) Use it sparingly. (2) It's best used on mint un-played discs. (3) After spraying lightly, buff the hell out of it! All my records, treated as above, are silent as a tomb -- no ticks, no pops.

My "PIF" Filters from Electronic Specialists work perfectly on my phono inputs. At the other end of the system, four . 03 uF caps, from all four speaker leads to the chassis ground of the amp, eliminate RFl on all channels, even though my rig is in the shadow of high-power interference.

Stereo Cost Cutters and Bob Heenan are absolute topsl Dependable, honest. I've eliminated all other suppliers on the basis of sad experience.

-- Oeveste Granducci (Virgin Islands)

The ADS Time Delay and Laments on the Passing of Quad

In spite of all the rationalizations I've heard, I don't think I will ever really understand why the audiophile community so emphatically turned its collective back on quadraphonic sound. Stereo also went through its teething days -- of noisy ping pong games, rumbly bowling balls, and locomotives, excruciatingly recorded and reproduced by less than sophisticated hardware -- and came through triumphant. Why wasn't the same benign tolerance exhibited for amorphous, gimmicky, distorted quad software reproduced on correspondingly poor hardware? Must be that on top of everything else you had to choose your matrix vs. matrix vs. matrix vs. discrete brand of stuff. Software and hardware maturity was just too slow in coming and when it finally

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arrived for several of the systems (yes -- it did finally arrive!) not only was there practically nobody listening, but the manufacturers and filers were pretty much tossing in the towel also.

While quad was slowly fading from the scene, interest in time delay systems for "ambience enhancement" was beginning to stir. At present, it seems to have gained a foothold among the more venturesome and/or well-heeled consumer audiophile community. While some of us feel that time delay can't do anything that quad (at least discrete quad) couldn't do equally well or better, an undeniable advantage of time delay systems is that they work on conventional stereo, or even mono, material without source encoding of any sort. Furthermore, control of the effect is adjustable by the user to suit his tastes. The fact that rather similar effects (psychoacoustically) relative to a feeling of expanded space for conventional stereo material can be obtained by simple A-kB type back channel hookup (left hot to right hot, Dynaquad, RM matrix, etc.) brought forth emotions ranging from polite yawns to black rage from most audiophiles (not all, mind you). To me, that seemed a shame. Having lived now for quite some time with Dynaquad, SQ, SQ logic, RM, CD-4, and much more recently with a time delay unit, I know they <u>all</u> work -- admittedly with varying degrees of effectiveness and variability.

A very few CD-4 records have been available where everything seems to come together right, which, reproduced on the very few really good CD-4 pickups and demodulators (so much CD-4 software and CD-4 hardware was absolutely abominable) creates a sonic image that is absolutely stunning! (Consider that statement as a eulogy to a technology which apparently died right in the process of recovering from a long and severe illness. Sad.)

Getting down to specifics, I have recently added the ADS 10-01 digital time delay unit to my system. That's the ADS 10 system stripped of amplifier and speakers for those of us who are already endowed with same. I assume most BAS readers are familiar with time delay units, pros and cons of analog vs. digital, etc. I will mostly describe what I think is important about the ADS approach.

It has a single channel memory bank, leaving no trace of original source left/right directionality in the generated ambient sound field. As far as I am concerned, this is a benefit rather than a drawback.

I am sold on the idea that left/right directionality in the ambient sound field is of no significant importance even for the "early arrival" delays. It may, in fact, be detrimental to a realistic impression. (Peter Mitchell presented a lucid summary of his views concerning this subject on a recent WBUR "Shoptalk" program with which, for the most part, I agree.) This is <u>not</u> to say that the ambience loudspeakers can be permitted to have identical phase-coherent signals presented to them. To repeat what has often been stated, the possibility would then exist for a monophonic-like localizable ambient sound field to be generated which would not be realistic (I say possibility because this may be defeatable by asymmetric location and/or aiming of the ambience speakers). By use of "non-coherence" networks, different order of presentation of delays to the ambience speakers, etc., non-coherence can be achieved for the time delay outputs, even from a mono source. Now, if directionality is not being sought, why go to the expense of separate left/right memory banks (plus the required separate analog/digital-digital/analog converters)? Why not reinvest the cost savings to improve S/N, distortion, frequency response (yes, 10 kHz <u>does</u> sound better than 5 kHz), etc. ? There is one problem, however, introduced by this approach. If a single memory is used, the input signals must be combined. Should L + R be used, possibly enhancing the monophonic elements of stereo sources while correspondingly diminishing an ambience signal deliberately introduced out-of-phase into the recorded left/right signals? Should L - R be used (wiping out mono sources entirely)? Some other scheme? ADS has made the following choices: (1) provide an L + R mode (labelled mono); (2) additionally, provide a mode (labelled stereo) which phase reverses <u>a portion</u> of the frequency spectrum (~250 Hz to 1 kHz) in one channel, then sum the two channels. It does an excellent job of suppressing excessive ambience in a center located voice -- so noticeable on FM broadcasts. One position relative to the other can, and often does, change the nature of the ambient sound in this frequency range for non-vocal signals as well. The end product is a strong function of many recording variables, a most important one being microphone technique.

The unit's memory bank is tapped at several points and yields three initial delays for each setting of its "stage distance" and "hall size" selectors. These delays are mixed and can be fed back for reverb. The mix ratio for reverb can also be changed by a "character" selector, increasing or decreasing the proportion of short delay signal for narrow or wide time spacing between subsequent echoes. Delay taps vary from 10 ms to 100 ms and max reverb decay time is 1.6 seconds.

I go along with the viewpoint that it is preferable to listen to the original ambience signature of the recording environment (providing it is a good one, obviously) rather than to create one electronically. However, I am not so much of a purist that it bothers me to do some "enhancement" if it subjectively sounds good. How can we best get the hall on the record? Set up a coincident microphone pair for the direct signals (primarily). Make sure these mikes have as flat a phase-frequency relation as is available. Set up a single mike back in the hall for the ambience (primarily). Mix the output of this mike into the two front channels, the signals going to left and right channels being <u>out-of-phase</u> with each other. Choose a level for the back mike resulting in a good stereo recording. For playback through a single-memory-channel time delay, electrically combine the left/right signals out-of-phase. Relatively little additional time delay or reverb should be necessary. You might think a time delay unit wasn't even necessary. All that would be needed is one or two differentially fed ambience channels, or its close brother, the commercial Dynaquad unit. Well, that would almost be correct. The big difference would be that relatively extreme left and right signals are not reduced in amplitude by the L - R connection, and combined with the vagaries of minor speaker frequency response aberrations, room mode effects, listening position, etc., they would, rather often, become localized sounds emanating from the ambience speakers -- to the disgust of some and the delight of others, depending primarily on their innate hatred of quad. The time delay unit gives you that 10+ ms of delay vital to keep the primary image up front. By the way, the ADS unit <u>does not</u> give you that L - R option across the entire frequency spectrum, so useful for the recordings produced as just described. I wish they did. (As an aside, summing the back channel outputs of an SQ decoder will essentially do just that.)

As you may have already deduced, any front microphoning technique which maintains left/right zero relative phase for each sound source, e.g., close multi-miking of individual sound sources (a pox on the puristsl) applies equally well to coincident miking. (In spite of nasty things which can be said about close multi-miking, good control of phase coherence in the end product can be achieved.) Wide spaced omnis, on the other hand, whatever their other virtues, are not particularly supportive of matrix augmentation of ambience coupled with time delay. Except for a very narrow region along the perpendicular to the line connecting the two mikes, the sound fields at each mike created from each single sound source are essentially random in phase relation, and simple matrix operations don't do very much to them. (One can argue, however, that the spaced omni approach often produces signals rich enough in ambient sound so that further augmentation for the delay channels is excessive.)

Getting back to the ADS unit, I feel that the extension of bandwidth to 13 kHz versus the 5 to 8 kHz bandwidth typically offered by other units, is a substantial advantage. I have never really been comfortable with the technical (and economically convenient) argument that concert halls naturally roll off the highs above 2 kHz anyway, so that one needn't have an extended high end for the ambience channels. Well, my living room's absorption characteristic, plus good placement of the ambience speakers (i.e., with the listening position well off-axis) also rolls off the high end. Furthermore, a gradual roll off above 2 kHz and running into a stone wall at 6 kHz are <u>not</u> the same thing. Although ADS provides for 5 kHz and 8 kHz choices in addition to 13 kHz, the full bandwidth sounds best to me.

Setting up the input gain so that a .5 volt pink noise input just flickers the peak signal LED, and adjusting the output gain for .5 volt out, I measure a S/N ratio (half rotation of reverb control, next to longest delay settings) on the delay channels of 83 dBA. Noise is just not a problem with this unit. Nor is distortion, subjectively at least; I made no measurements.

Provision is made for an additional delayed output for another set of ambience speakers using only the intermediate and longest delay taps (one to each channel) rather than the mix of short/intermediate/long taps used for the main delays. This is actually how I am using the unit with front, side, and rear speaker pairs. The unit is fed from my quad decoder, front outputs to main in and back outputs to tape in. Needless to say, plenty of combinations are available (maybe too many as a matter of fact). Most often I find the best effect for encoded program material is to use the back channel decoder output into the L + R (mono) blend of the time delay. For straight stereo material, the best sounding combinations are, as previously mentioned, a strong function of the original miking employed. Something compatible and good can always be found.

I haven't used the front channel provisions which exist for mixing some delay into the direct signal because I completely bypass the unit for input to the front speakers.

In summary, I am satisfied with the audible effects of time delay generally, and more than satisfied with the ADS 10-01 specifically. Nevertheless, or perhaps additionally, I would be delighted with the resurrection and advancement of quad technology and software. What can you get from quad that you can't get from time delay? Well, for one thing, there's the true ambient sound of Symphony Hall. For another thing, there's this doubles ping pong game ...

Jerry P. Davis (Massachusetts)

Slow A-B Comparisons

In James A. Mitchell's October-November 1979 Speaker article, a suggestion was made that A-B comparisons be made very slowly, so the ear/brain system can latch onto differences and not be confused by fast changes. The article points out that, while this might easily be proposed, it is not usually done by people in the "objective test" camp.

Actually, I have done this many times, and I reported the results of one such test in an article scheduled for the March 1980 issue of High Fidelity. The result of the slow comparison was exactly the same as for the fast comparison, provided the tests were run double-blind. In other words, there was no audible difference between power amplifiers when frequency response differences were eliminated by equalization.

By the way, I have more recently tried a slight improvement over the technique I discussed in High Fidelity, with, again, the same results. The improvement is to have the listening jurors file their ballots absolutely anonymously, so that neither you nor they can ever find out who got the answers right or wrong. Then there is less tendency for people to be intimidated into saying "no difference." You can't look at the ballots until more than one juror has listened.

Here's hoping that more people will try these rigorously-run tests, both slow and fast.

-- Dan Shanefield (New Jersey)

Further Discussion on Equalizers

As a comment on Collins Beagle's objections to equalizers in the October-November 1979 Speaker (pp. 11, 12), I have to admit that there could be cases where further experiments need to be done in order to settle the matter. For example, if Collins limits himself to playing direct-to-disc recordings or master tapes, there certainly are only a few components in the total audio chain, and one more (the equalizer) could be significant in a detrimental way.

In my earlier writings I was thinking of the recordings that we listen to most of the time, in which the disc is made using a non-flat mic/room system, a mic preamp, mixer amp, tape amp and EQ, Dolby A and its EQ, tape playback and EQ, de-Dolby A and its EQ, discretionary EQ, disc cutter amp and EQ, and usually even more links in the chain. It's true that one more link could have an effect which is other than zero, but I was claiming it would be masked by all the other items.

However, direct-to-disc records are becoming quite easily available for critical listening these days, and more attention is evidently given to keeping the system simple in making such recordings.

Also, if Collins Beagle has an unusually good room/speaker combination, there appears to be a somewhat diminished need to consider an equalizer. But even in that case, the microphone used for recording doesn't have an exactly flat frequency response (except in the manufacturer's own anechoic chamber -- it's not even flat in most other anechoic chambers!). And the disc cutter and playback cartridge also are never exactly flat. So an equalizer might still be of some use for obtaining the flattest overall response, providing it doesn't do any harm.

Does it do some subtle but audible harm, in an unmeasurable way? To settle this, how about doing a double-blind A-B test of an equalizer versus no equalizer, where the listener purposely adjusts the equalizer to have the flattest practical frequency response? I have done this and reported the result in an article scheduled to be published in the March issue of High Fidelity (see last item in Table 1 of that article when it gets printed). The equalizer tested was a Soundcraftsmen 20-12, but I have recently repeated the whole experiment with a Crown EQ-2. A variety of direct discs and master tapes was used, with an ADC XLM (I), Dyna PAS-3X (modified), Dyna 400, and Magnepan MG-U speakers. All the precautions listed in my High Fidelity article were followed, since I have found from previous experiments that these are quite necessary if spurious results are to be avoided.

The result was that listeners could not tell whether the carefully-adjusted equalizer was in or out, as long as the experiments were run double-blind. But I hope other people will continue to repeat these tests, with other systems, other ears, and more statistics.

-- Dan Shanefield (New Jersey)

The Hafler DH-200 Amplifier

Much press coverage has been dedicated to Ed Gately's and Dave Hafler's designs of late. For those not fortunate enough to have had much exposure to the DH-200, we offer a few comments.

1. First, whatever you've heard good about this "little black box" forget it. It's better than that. This little amp will make you forget, at times, that you're listening to electronics, if the rest of your system is in the same league as the DH-200 (most aren't).

2. The 100 watt per channel rating is most misleading. You can fill your room with more than 100 dB SPL, using inefficient speakers, before clipping occurs, and the clipping characteristics are gentle -- most likely you'll never hear it. We had to connect an oscilloscope to detect it, and could listen only for short intervals; the SPL was just too high, and none of us really wanted to incur permanent ear damage.

3. The sonic signature of this electronic wizard is near the leading edge of the technology (aren't we all tired of hearing "state-of-the-art"?). Detailing, imaging, depth, etc. are the best we've heard at less than $2,000. The Hafler's sound is not due to the MOS-FETs in the output stage, but to the way they are used. (MOS-FETs are thermally stable, so no troublesome protection circuits are needed. -- Ed.)

4. Unit-to-unit sonic variability is nil. We have auditioned six (6) units, most built in the summer and fall of 1979, and two (2) from January- February 1980. They all sound equally excellent.

5. Assembly time is three to four hours (for a kit builder who has assembled two or three kits). The boards are already "stuffed" and mounted to the heat-sinks, which also have the output devices pre-mounted, wired, and tested. The transformer is already bolted into place. All you have to do is mount some fuse holders, five-way output binding posts, power switch, line cord, rectifier block, and connect 20 or so wires and you're finished. No adjustments are necessary. It's just not necessary to buy an assembled version, unless you need one immediately and that's the only one available.

6. Reliability is very high. We've managed to blow a few fuses, but that's it. By the way, Hafler supplies both 2-Amp and 5-Amp fuses for the speaker outputs. Forget the 2-Amps. They'll blow immediately on anything other than the most efficient speakers.

7. For those of you with Dahlquist DQ-LP1 Low-Pass Filters and the like, who need to calculate the value of the crossover capacitor, the input impedance is 22K ohms. This fact was supplied by The Hafler Company in Pennsauken, New Jersey, who will gladly answer any questions, technical or otherwise.

8. The input bridging mod (to allow the DH-200 to operate at a rated 300 watts mono into 8 ohms) is not yet available. It most likely won't appear in stores until late April at the earliest. (Please note: most amps do not sound as good "bridged" as they do unbridged.) Hafler says they should hold the price at the original estimate of $24.95. For those of you running "double" speakers, like Double Magnepans or Double Advents and who like your listening levels very high, here's an idea: buy two DH-200's to use one amp for the left side and one for the right side. In building the amps, wire both inputs (left and right) together, with a twisted pair of wires, internally. The Hafler is unusual in that you get more wire than you'll ever need to build this beauty. Then, to maintain the proper frequency response, on one channel board only (not both!) unsolder one end of R1, a 470K 1/2 watt resistor. Then connect your preamp up to each amp, using either input. Each speaker is driven by a separate output; for example, Left Speaker 1 to Left Amp-Left Output, Left Speaker 2 to Left Amp-Right Output. That's all there is to it. Hafler is working on a larger amp, possibly to be released later this year. How it sounds we won't know for a while.

9. Dealer reactions to this product vary widely. Some are mad because they can't get enough; some are mad because they can't get a Hafler dealership. Some don't like the small profit for such a super product. Remember that 32% of $330 is only $105, while 32% of $1,000 is $320. Most dealers, surprisingly, have never listened to the amp, and if they have, not for much time. They just can't keep one around long enough, despite the fact that the dealer arrangement provides that the first unit be an assembled amp to be used for demo purposes.

Consequently, you may find this amp being sold against another brand. The dealer (or salesperson) needs to make the sale. He can't afford to let you walk away without buying something. And for most dealers, any amplifier you buy is okay with him -- whether it sounds worse than a DH-200 or not.

10. Dealer shipments are improving. The Hafler Company has received a lot of "bad press" from some dealers. A couple have said the company is near financial collapse (God forbid!), some say they aren't paying their suppliers, etc. The facts are that, like the entire electronic industry, they can't get parts fast enough. The good news is that many dealers are now receiving shipments every two and a half to three weeks. So before you put down a deposit with a "four to six month estimated delivery date," call around.

We thought we would share these comments and experiences that about eight of us have had with this very musical device. You probably wouldn't even believe the sonic differences we heard when comparing the DH-200 against some of the most musical (and prohibitively expensive) amplifiers available. For a real treat, take your S.0.T.A. amp and a Hafler and feed each one directly from a quality recorder, like a Revox or Crown, bypassing a preamp directly. Many manufacturers would not want you to hear the sonic results. Congratulations Dave and Ed. You've created a real beauty. This amp is going to be around a long time.

-- Mike Lulejian (Georgia)

Two Dealers

All of us have come into contact with good hi-fi retailers -- and terrible ones. This industry seems to run the gamut of qualified and unqualified dealers, just like the automobile business. Competent specialty dealers (like Spud Wilmer at Soundd Investment Company, Chamblee, Georgia, who has the best tape deals and service around) are a little easier to find.

Each of us has his own impressions of local dealers, mail-order houses, etc., and we all have different philosophies regarding supporting our local retailers. Most of us are fortunate enough to have qualified high-end dealers within a 50-mile radius of our homes. Every now and then, however, one is extremely fortunate to find an outstanding audio dealer. I have been lucky enough to find two such dealers when I planned to upgrade my system. One is Sandy Gillman of Audio Den, Ltd., Stony Brook, Long Island, and the other is Bill Gibson at House of Stereo in Jacksonville, Forida. Amazingly, I wound up doing business with both of them by telephone before I had a chance to meet them personally.

What makes these two stores so outstanding? Well, of course, they fulfill all of the typical audiophile criteria -- they have broad lines of equipment, good listening rooms, a no-rush attitude, excellent service, etc. Both Sandy and Bill, however, carry some hard-to-find products, and they have lots of experience and a desire to share this with their customers. This contrasts with the typical hi-fi store salesman who attempts to condition you and your wallet by quoting what Joe Schmoe at Hi-Fi Hype and Hi-End Magazine says is going to be his preamp choice of the week.

I also found Sandy and Bill to be very interested in their customers after the sale. They will bend over backwards to help you, if necessary. And, unlike many high-end stores, both of them respect their competitors, and will recommend another dealer if he has equipment you want. They never knock other dealers. What all this means is that both Sandy and Bill understand not only their business, but their customers as well. They realize that the quest (yours and theirs) is for better music. They appreciate music, its conception and its reproduction, and they know that the latter almost requires a "hobbyist" attitude at times. I highly recommend each of these dealers to you.

I feel strongly that the Speaker's policy of printing both good and bad impressions of dealers should continue. There are relatively few in each category, and we should know about each of them.

-- Mike Lulejian (Georgia)

In the Literature

Absolute Sound, No. 16

*Letters (p. 406): An excellent note on potential digital distortions, and another which ignorantly blames the lack of ambience in the Telarc "Firebird" on the digital process.
*Components in Review (p. 414): Scheiber Spatial Decoder (one reviewer likes it, another doesn't, neither has much experience with time-delay for comparison; costs $3,000, is fun to play with, decodes SQ well, makes nice ambience and remarkable "discrete" quad, but may not improve realism). Infinity 4.5 speaker (very transparent; crisp highs, great deep bass, boxy midbass, midrange is thin but clean and analytical). Koetsu cartridge (the world's best, costs $1,000; sweet, transparent, detailed, outstandingly clean). Precision Fidelity C-7 tube preamp (minimal flexibility, mostly very good performance). Theta Electronics tube preamp (minimal flexibility, neutral, sounds good but not great).
*Shorter Reviews: Fidelity Research Mk IIIF m. c. cartridge (first sample wonderful, second sample glares). Yamaha C-2a preamp (vastly better than the old C-2, mostly superb). Electrocompanient power amp (sounds wonderful, clips at 23 W, costs $1,100). KEF 105 speaker (dry, analytical, great imaging). Hadcock tone arm (damped unipivot design, laborious to set up, is ideal for the Sonus cartridge). A&E 2000 preamp (expensive but quite good overall). Magnepan MG-1 speaker (remarkably good at the price). Levinson ML-3 power amp (tentatively: fantastic; note that pages 466 and 467 are interchanged). Stax SRX III electrostatic headphones (uncolored, superbly analytical and revealing, but many samples have elevated top end). Signet TK 33 electret headphones (fantastically accurate, uncolored, revealing, smooth, comfortable). Infinity ES-1 electrostatic headphones (great bass, fine detail, but not ideally neutral). VPI record weight (works very well, weighs 1.3 pounds). VPI isolating turntable base (dramatically improves direct-drive tables). Electronic Specialists RFI filters (line-cord filters most effective). Audio Technica AT620 cable (makes a shocking improvement).
*Technocracy: The theory of the new Quad electrostatic, plus a thoughtful review of the Philips/ Magna Vision videodisc system (at its best, remarkably impressive; but problematic -- especially MCA's sloppy disc manufacturing).
*Interview (p. 487): Ed Wodenjak of Crystal Clear discusses his direct-to-disc recordings, why he likes D-D better than digital, and recommends the Stanton 881S cartridge.

*Record Reviews (p. 495): Digital is now the favorite demon, blamed for all faults including those more plausibly due to bad miking, plating, or pressing.

Audio, January 1980

*Video Scenes (p. 8): Whyte reviews the GE projection TV.
*Audio ETC (p. 16): Canby's memories.
*Behind the Scenes (p. 22): Dynamic range verities.
*Just Call Me Maestro (p. 48): Whyte recalls Arthur Fiedler's direct-to-disc recording session.
*Importance of Dynamic Range (p. 62): A thorough survey of the problems of recording wide dynamic range on tape or disc, leading up to the dbx encoded discs.
*Kill FM Interference (p. 68): Using a spaced pair of FM antennas to phase-cancel interfering adjacent-channel signals.
*Tone Arm Geometry Demystified (p. 76): A complete set of equations and tables for optimizing lateral tone arm geometry, with alignment errors computed for 22 representative arms.
*Equipment Profiles (p. 90): Mitsubishi microcomponent system (tuner mediocre, preamp good, power amp good but for marginal instability). Pioneer 630 turntable (good arm geometry, excellent overall performance). DB Systems DB-6 power amp (utterly clean sound, graceful overload behavior, has enough dynamic headroom and current reserves to play much louder than its 40 W rating suggests). Sherwood S32CP tuner (a good budget-priced unit).

Audio Engineering Society Journal, December 1979

*A Triphonic System (p. 965): A three-channel recording and playback system yielding good localization and ambience; it consists of a coincident pair of figure-8 mikes which are fed to side-located speakers on playback, plus an omni fed to a front-center speaker.
*Production Testing of Loudspeakers (p. 970): About how KEF uses a computer for rapid measurement of drivers and automated selection of matched pairs for stereo.
*Digital Audio Disk (p. 975): Details of Sony's proposed PCM record player, with emphasis on its error-correction schemes.
*Solid-State Integrated Filters (p. 982): About the remarkable Reticon IC filters which are slashing the cost and complexity of spectrum analyzers.

DB, January 1980

*The Pressure Recording Process (p. 31): A miking technique, to be marketed by Crown as PZM (Pressure Zone Microphony); it eliminates comb-filter colorations due to reflected sounds by placing the mike diaphragm within a few millimeters of a reflecting plate or boundary surface.
*Distortion- Measuring Microphones (p. 34): Notes on some distortion mechanisms in mikes, suggesting the virtue of using B&K measurement mikes for music recording.
*VU Meters vs. Peak Program Meters (p. 46): The advantages of peak-reading meters for tape recording are now being recognized in the U. S., but PPMs with standardized behavior have long been used in Europe.
*Re-inventing the Microphone (p. 50): A few odd designs from microphone history.

Gramophone (England), December 1979

*Critics' Choice (p. 969): Picking the best of the year.
*Sounds in Retrospect (p. 1066): A second look at recent discs.
*Equipment Reviews (p. 1084): Teac A300 three-head cassette deck (fairly good). Audio-Technica AT30E, Mayware MC-2C, Satin M117Z, Ultimo/Dynavector 30A, and Westrak 501E moving-coil cartridges (each has virtues and defects; Mayware and Dynavector preferred). Castle Kendall II speaker (lacks deep bass, otherwise a good value).

Hi-Fi News & Record Review (England), December 1979

*The Fearby Front End (p. 105): A $10 homebrew battery-powered MC head amp.
*Mikes & Miking (p. 109): All about the Calrec sound-field mike in theory and practice.
*Interference (p. 131): Notes on FM image rejection, RF filters and RFI, including buildable filter designs.

*Disc Information (p. 120): An appropriately critical view of record jackets.
*Equipment Review (p. 187): Four small speakers (Chartwell PM110 and KEF R101 excellent, Videotone GB3 and Infinity Infinitesimal mediocre). Five cassette decks: Sony TC-K75 (best of group, three heads, very flexible controls, fine performance); JVC KD-A5 (a "best-buy," fine performance at a modest price); Aiwa AD-L40 (good with ferric and ferrichrome tapes, mediocre with chrome); Teac A-510 (poor with chrome tapes, otherwise okay); Harman-Kardon 2500 (poorly set up, overpriced).

High Fidelity, January 1980

*Equipment Reports (p. 19): Carver C-4000 preamp (elaborate and flexible, requires careful and judicious adjustment for best results; autocorrelator is subtler and better than earlier versions; time-delay is "mildly pleasant"; peak unlimiter useful if adjusted right; sonic holography is "terrific," much better than stereo; measured S/N ratios are mediocre). KLH 3 mini speakers (smooth, warm sound; remarkable bass, good dynamic range). Ortofon Concorde 30 phono cartridge (low mass, superb tracking, wide separation, fine sound). SAE Two R-6 receiver (remarkably flexible controls and dubbing functions, unusually useful FM signal-strength/multipath metering; non-standard phono impedance, shallow filtering, good headroom and current capacity in power amp). Jensen System B loudspeaker (efficient, warm, smooth rather than brilliant, plays loud). Phase Linear 5100 tuner (made by Pioneer, superb in every way except mediocre selectivity). Pioneer CTF-1250 cassette deck (elaborate tape-matching system works well; overall performance very good, especially with chrome tape). Dual 731Q turntable (superb performance; ULM arm has low mass and good damping, though calibration of damping is off). Spatial Coherence TVA- 1 preamp (non-standard phono impedance; tone and tape controls unconventional but effective; preamp sounds good). JVC Zero-5 loudspeaker (high power handling, apparent crossover problems, midrange doesn't sound good).
*How Much Are Old Records Worth (p. 53): Finding treasured rarities in the closet.
*Input Output (p. 98): Details on Teac's "Portastudio" four-track self-sync cassette recorder.

Modern Recording, January 1980

*Talk Back (p. 22): On making an acoustically isolated room.
*Apocalypse Now (p. 39): How the film's extraordinary sound track was made.
*Lab Reports (p. 62): Teac 144 Portastudio (a remarkably effective multitrack cassette recorder, ideal for musicians wanting to make demo tapes). QSC A42 power amp (expensive, conservatively designed, intended for pro use). LT Thompson TAD-4 analog delay (bucket-brigade time-delay plus spring reverb, works remarkably well).

Popular Electronics, January 1980

*Stereo Scene (p. 20): Notes on amplifier feedback and ambience reproduction.
*Test Reports (p. 24): JVC KD-A8 cassette deck (microprocessor auto-bias system works well, overall performance is outstanding except for bass rolloff). Osawa MP20 phono cartridge (superb sound, outstanding tracking ability). Celestion Ditton 662 speaker (very smooth, wide range, excellent bass, good balance).
*Vented Speakers (p. 48): A simplified approach to making bass-reflex speakers based on Thiele's alignments.

Recording Engineer/Producer, December 1979

*Acoustic Effects of Space and Materials (p. 40): A good introduction to room reverb, reflectivities of common materials, calculation of reverb time, etc.
*A Fuss About Plus (p. 66): Speculating about the preservation of absolute polarity in the recording chain.
*Ferrofluid as a Loudspeaker Component (p. 80): How the black goo is used to make speakers better.
*The Calrec Sound Field Microphone System (p. 90): How it works in theory and in practice.

Stereo Review, January 1980

*Audio News (p. 28): Car stereo standards, and their limitations.
*Audio Q&A (p. 30): Live-concert sound at home.
*Audio Basics (p. 32): A historical perspective on buying components.
*Tape Talk (p. 34): Demagnetizing heads.
*Test Reports (p. 40): Hafler DH-200 power amp (splendid performance at all impedances, including an IHF dynamic output of over 450 W into 2 ohms; audibly outperforms other amplifiers of similar nominal rated power; kit version notably easy to assemble). Polk 10 speaker (wide dispersion, unusually good tone-burst response, notably open and unboxy sound). Technics 8077 tuner (very low profile, good performance). Sanyo Plus 75 stereo receiver (power amp good at 8 and 4 ohms, very flexible controls, preamp good with included MC head amp, tuner pretty good). Shure SC39ED phono cartridge (an unusual combination of ruggedness and good performance; outstanding tracking ability).
*Digital Demo Discs (p. 64): David Ranada surveys 16 digitally-mastered discs and finds them not all equal.
*Digital Decade (p. 69): A glance at proposed video disc and digital disc systems.

Studio Sound, January 1980

*Digital Recording Next Year ? (p. 40): A survey of digital tape recorders.
*AES 64th (p. 44): Report of last fall's AES convention in New York, with lots of details on proposed digital tape and disc equipment. -- Peter Mitchell

December BAS Meeting

Business Meeting

The BAS meeting of 16 December 1979 was called to order at 6:22 PM at GTE Labs in Waltham by President Peter Mitchell, who announced that the January meeting would be held one week early, on the second instead of the third Sunday. Bob Carver would be on hand to explain, "for the first time anywhere," the complete operating theory and practice of the Magnetic Field Power Amplifier.

Transition difficulties have delayed the production of the BAS Speaker, but a combined October/November issue was due to be mailed shortly. The following three issues, hopes Editor Brad Meyer, will appear at roughly three week intervals, but this depends in large part on the availability of your contributions. Articles, comments and other submissions should be typed, double-spaced, and sent, posthaste, to The BAS Speaker, Trapelo Road, Lincoln, MA 01773.

The rate of growth of membership in the Society has slowed; the Executive Committee is therefore considering advertising. While there is a risk of too- rapid growth, particularly if new members are primarily readers rather than active contributors, the potential diversity of new blood is considerable. Suggestions included placing ads in The Audio Amateur, The Boston Phoenix (a local weekly), IEEE Spectrum (to attract professionals as well as audiophiles) and leaving membership applications in area hi-fi shops.

Frank Farlow reminded the membership of the Constitutional Review and said there would probably be monthly meetings. Suggestions are welcome.

Suggestions are also welcome for the meeting program committee. Peter Mitchell eagerly awaits the as-yet-unmaterialized flood of response.

Open Forum

In addition to the usual spate of equipment offered for sale at the meeting, there were several brief personal impressions of equipment. First, A1 Foster said that his experience of the Theta tube preamp was almost uniformly negative: when the tape output was loaded, the main output level changed; the 1 kHz distortion was 0.4% at only 2 1/2 volts output. These are unacceptable qualities in an $800+ preamp. Al has also used an Adcom power amplifier, which is at least average for its price. However, it is a bridge-mode amp, even in stereo into 8 ohms, so be careful about return leads -- no common grounds ! It will produce 250 W/ch at 8 ohms, 300 W/ch at 4 ohms, and 60 W/ch at 2 ohms.

A Beckman 310 digital multimeter (for about $100) was extolled as worth the price, so long as you know most DMM's are only accurate to about 10 kHz on the AC scales. Gary Bergstrom reminded the membership of his DMR100 digital decibel meter for audio work.

Anyone out there have any experience with Speakerlab speaker kits ? There wasn't anyone forthcoming at the meeting, but if you have any, write it up!

The Audio Amateur is increasing their audiophile coverage and spinning off speaker-construction articles into a new publication called, appropriately, "Speakerbuilder." It costs $8 per year.

Alpine, makers of Motorola and Jensen car stereo gear, have introduced their own line. The Model 7307, a Dolby-equipped tape and FM unit, has extreme picket-fencing problems as the AGC time constants interact with the Dolby time constants. Caveat emptor.

Meeting Feature - dbx

There were to have been two features of this meeting: a presentation about disc mastering and pressing, by Direct Disk Labs of Nashville, Tennessee, and a discussion and demonstration of the dbx-encoded disc project, by dbx of Waltham, Massachusetts. As it turned out, Direct

Disk never arrived, having been delayed elsewhere on the East Coast, so the discussion began as Les Tyler, Chief Engineer for dbx, described the dbx encoding/decoding process.

Actually, there are two dbx product lines: dynamic range modifiers, like the 3BX, and dynamic range preservers (or noise reducers). The noise reducers further fall into two groups, "professional" (dbx Type I - the 150 series) and "consumer" (dbx Type II - the 120 series). The disc system is Type II and is fully compatible with all 120 series tape equipment.

dbx Type II is a double-ended (noise prevention) system, utilizing encoding before the storage or transmission channel and complementary decoding after. The dynamic range of the signal is compressed by a factor of 2 and high frequencies are pre-emphasized to enhance psychoacoustic masking; the process is reversed after transmission or storage to restore the original signal dynamics and frequency balance. As you can see in Figure 1, if the information transfer channel, or ITC -- a tape deck, for example -- is perfect, then the output of the encoder will be identical to the input of the decoder (the points marked "X").

By making the encoder a feedback device and the decoder a feedforward device, we can take advantage of this identity to use the same time constants and level detectors for both halves of the process. Figure 2c shows the appearance of a high-frequency tone burst at the input and then at the output of the encoder diagrammed in Figure 2a. They are very different, but this second signal will also be available at the input of the decoder (in 2a and 2b, identical signals appear at the X's, theoretically allowing perfect reconstitution.

In the real world, transmission channels are not perfect. Frequency response is not flat; tape recorders exhibit phase shift; discs have rumble. Several techniques are used to surmount these challenges. First, level detection is done by means of the RMS value, rather than the average or peak values. An RMS detector is inherently phase-insensitive -- a square wave with the high frequencies shifted will have a lower average, and a higher peak, but the same RMS value. The RMS calculations are done in the logarithmic domain by proprietary circuits developed by Dave Blackmer. "It is a tribute to today's transistors that the distortion stays so low in the log domain," says Tyler.

Second, a true RMS value is the result of an integration performed over all time. This isn't terribly useful for any human application, especially music, so a compromise must be made, in the interest of practicality, in the system time constant, i.e., rate at which the output of the RMS detector is allowed to change. If the attack time is too short, the voltage controlled amplifier (VCA) will start to "track" the input waveform; the resulting harmonic distortion is equivalent to an increase in signal bandwidth, or sideband generation. At no time do we want to exceed the bandwidth of our channel (usually a tape recorder), or we will lose some information needed to reconstruct the original signal. Fortunately, music has finite attack and decay times and human hearing has fairly long integration times, so this problem is not too severe and acceptable compromises can be made.

Third, to avoid problems with tape recorder frequency response irregularities, and to avoid audible hiss-pumping, the input signals are bandlimited and pre-emphasized as shown in Figure 3, before going to the VCA. The RMS detector input is further low-pass-filtered so that irregularities above 10 kHz will not cause inaccurate level-sensing. The decoder RMS detector has the same input filtering, and the output of the decoder VCA has complementary de-emphasis.

One additional problem arises with disc decoding, and that is turntable rumble. The disc decode switch on 120 series equipment, and on the new Model 21 decoder, adds another 2-pole 30-Hz high-pass filter at the input, before the "normal" input to the decoder circuitry.

While the instantaneous signal-to-noise ratio of the channel is unchanged by the encode-decode process, the effective dynamic range is increased because the signal level is being shifted to keep it within the available channel, above the noise floor and below the distortion ceiling. The implicit assumption here is that the inherent signal-to-noise ratio of the channel is sufficient to allow high-level signals to mask the noise floor, and that ambient noise in the playback environment will mask noise during low-level signal passages. So far, dbx's experience with the discs has been that the test pressings are quiet enough that all audible noise comes from the master tape and is thus part of the "signal."

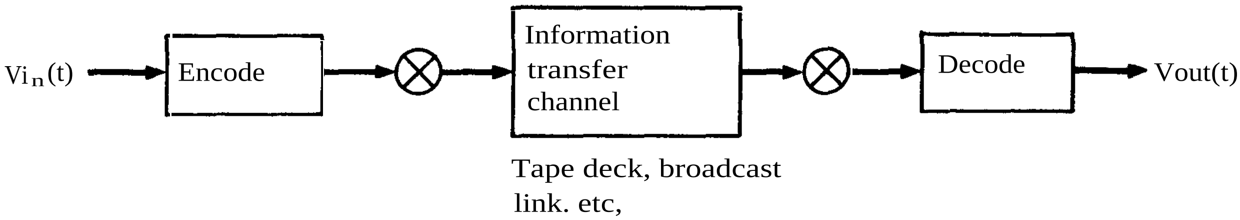
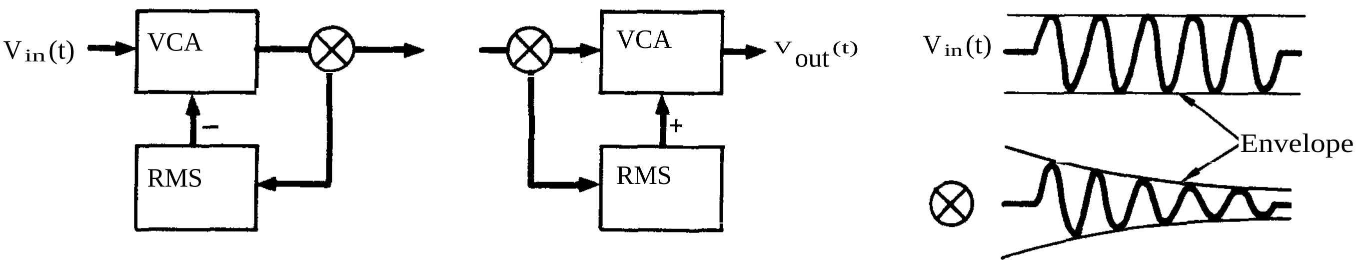


Figure 1



(a) Encoder

(b) Decoder

(c)

Figure 2

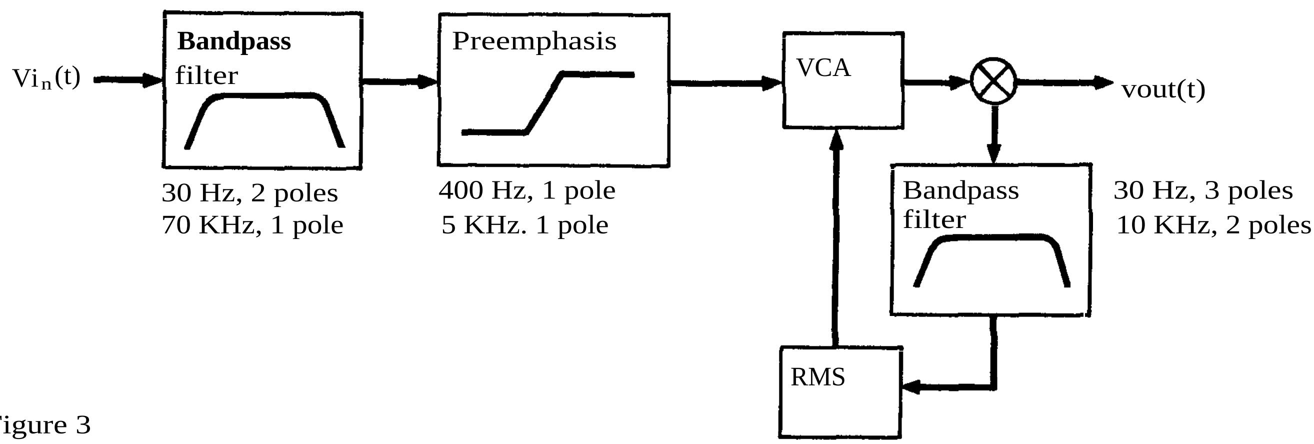


Figure 3

A dbx-encoded disc has several potential advantages over an ordinary disc. Normal mastering practice often involves the use of limiters, 50-Hz high-pass filtering, gain riding or other techniques intended to keep average levels high (and apparent noise low) and still allow reasonably long playing times. With dbx 2:1 compression before the disc cutterhead, none of these practices should be necessary, allowing the listener at home to reverse all the disc processing and restore the original balance and dynamics of the master tape with little or no additional noise.

Dynamic range, said Jerry Ruzicka, Vice-President for Marketing at dbx, carries much of the emotional impact of live music, and it is generally butchered on discs. Even if the signal is one which is often thought of as compressed, like rock or pop, the silences are a necessary part of the overall experience, and when they are filled with noise, something is lost. Digital recording, while it will reduce the noise level of the master tape and increase the potential dynamic range, will require a comparably quiet mass medium to achieve their full impact. The home digital disc is not yet here, but the dbx disc is here.

Of course, the best transfer to disc sounds like the tape -- including flaws. "Analog tape 'flakes' like crazy," said Ruzicka, and to prove it he played a dbx copy of a Copland piece which was recorded about 15 years ago. There are other flaws, too, which are no longer lost in the disc surface: overzealous editors who cut off the reverberant "tails" of final chords, tape hiss (although Dolby 'A' masters are acceptable and dbx ones are better), and comments by musicians and/or producers which were left in the "take" because they were expected to be inaudible.

People who listen to recorded music have trained themselves to ignore the absence of dynamic range and the presence of noise. This can include noise from the low-level high-gain phono inputs of your home playback equipment, hum pickup from fluorescent light fixtures, and other background annoyances. Because the dbx decoder is in the circuit after the common entry points for such interference, the system noise is reduced along with the disc noise. It is possible to get so used to this absence of noise so quickly that you lose your sense of perspective and forget just how bad most records really are.

One effect of this is that one becomes acutely aware of every tick or pop on a dbx disc. In the real world, minor imperfections are going to be unavoidable, and Ruzicka said that he finds himself having more and more sympathy for record companies. Warps are one problem, for example, that no electronic processing will prevent. Surface noise in general is pushed so far down by both the expansion and the de-emphasis (which, incidentally, also reduces the level of inner-groove distortion products) that even a pressing which would have been moderately noisy for a conventional product is often acceptable. Most ticks and pops are brief enough that they are "ignored" by the RMS detector time constants, and gouges deep enough to be expanded would probably be cause for discarding the disc anyway.

Five years ago, some small record labels started releasing selected records in dbx-encoded form. Those records are compatible with the current and future releases; the only difference is in the marketing approach. dbx is not asking any record company to take financial risk or to "double-stock" titles. The record company sells the rights for dbx encoded releases to dbx, who then arrange mastering, pressing and distribution through their own dealerships and (in some cases) regular record shops. The standard Neumann equipment used for cutting regular disc lacquers is used for dbx discs: in effect, the cutter-pressing-preamp-cartridge chain is assumed to be a perfect transmission channel between the encoder and decoder.

So far, dbx has obtained source material by calling record companies. In a continuing search for better masters, negotiations are underway with Telarc for some of their digital tapes, but no agreement has been reached. However, by the time this sees print there will be digitally-recorded material available on dbx discs: six albums by the Philharmonica Hungarica done on a modified Sony recording system, half-speed mastered using specially modified dbx encoders, and pressed in Germany. The (analog, dbx) product is expected to have a dynamic range of about 90 dB, and will be priced at $16. If you are one of those who worry about the potential for sonic anomalies in digital recording, there are some direct-to-dbx-disc recordings available, from Direct Disk Labs of Nashville.

Privately made tapes which the artist or producer may be trying to release will be considered by dbx, but obviously these people will have to contact dbx first.

One reason the dbx disc system has a better chance of making it in the marketplace now than it did five years ago is the greatly increased market penetration of dbx tape noise reduction equipment, meaning a greater initial market for the discs. Also, the introduction of the relatively low-cost ($109) Model 21 decoder means more people are likely to be interested. There are approximately 30 thousand dbx Type II units out there now; disc economics allow dbx to be reasonably happy if it sells 10,000 copies of something of which the originating company has to sell 100,000 copies.

After the break, Jerry Ruzicka hosted a discussion-cum-demonstration session. The playback equipment consisted of a Kenwood KD500 turntable with a Grace 707 arm and a Shure V15-IV cartridge, a Dunlap-Clarke Model Ten FET preamp, the dbx Model 21, a Yamaha Pro Series P2200 amplifier capable of 350 W/ch and a pair of M&K Satellite 1 loudspeakers. (The M&K speakers roll off below about 70 Hz on their own, because they are intended for use with the M&K Volkswoofer subwoofer system; they are small and capable of enormous power-handling.)

An Empire Brass quintet recording was used to demonstrate how much of the reverberant decay of music is normally masked by disc surface noise. A last note played on a triangle took over five seconds to die away on the dbx disc, compared to about two seconds on the normal pressing. The audibility of this low-level sound contributes greatly to the live experience, and is available at home because of the average 30 dB improvement in dynamic range of the dbx disc. A Bill Elgart percussion piece from Mark Levinson had audible tape hiss and print-through, and took twelve hours to cut acceptably because of the abundant high-frequency and transient content. Mark Levinson was reported to have said that it sounded alright.

A potential drawback to the absence of system noise is the risk of dropping the stylus on the record with the volume turned up. You could lose a woofer if your amp is powerful enough.

Typical commercial discs have a signal-to-noise ratio of around 45 to 50 dB. An "audiophile" pressing, if everything goes extremely well, will have between 60 and 65 dB. The dbx process immediately gains you another 25 to 30 dB of dynamic range. A special demonstration comparing the same master recording cut and pressed all three ways was made possible because Ruzicka had brought along a test pressing of the then-unreleased dbx edition of "Dreamboat Annie" by the rock group Heart, and The Audio Forum in Watertown, Massachusetts, had loaned the Society an unopened copy of the Nautilus half-speed mastered edition. Compared to the original Mushroom release, both the Nautilus and dbx versions sounded quieter and had more dynamic punch and high-frequency detail; there was still perceptible background hiss from the Nautilus, although even Ruzicka was surprised at just how quiet it managed to be.

Most commercial records are equalized and cut to be acceptable to the mass market. The dbx approach is more like the audiophile-disc producer's, and although there were differences in the sound of the dbx and Nautilus discs, they were more alike than either was like the Mushroom.

A question was raised concerning a dbx-Desmar release which appeared to be cut even closer to the center label than the original had been. Isn't one advantage of the dbx process supposed to be a reduction in inner-groove distortion by keeping the grooves narrower (lower level) and thus farther out?

Ruzicka explained that it is possible that some original releases were so highly compressed that, in returning to the dynamic range of the master tape, the music could take up more space despite the 2:1 compression employed by dbx. Also, average levels on dbx discs seem to be coming out about 2 dB <u>higher</u> in the critical 2 to 5 kHz region; the peaks should be somewhat lower, though, and that will ease the cartridge's job. The cutting levels are being chosen by experienced cutting engineers on the basis of results achieved in each case, and it seems to get easier the more experience they acquire. Ruzicka has found that the more complex the orchestration, the easier the record is to cut.

dbx discs can be taped; there is no need to decode first. But you might find, if your turntable rumbles, that the tape ought to be played back with the disc position of the decoder.

The system isn't perfect: no system is or can be. The tradeoffs seem acceptable, though; and what is left when you have removed recording noise? As Jerry Ruzicka says, "Real life; and does it sound weird !"

-- Mark Fishman

A BAS Test Report

Report of the BAS Turntable/Arm/Cartridge Clinic

Part 1

Alvin Foster, Brad Meyer, Alan Sliski, Frank Farlow,
David Ranada, and Tom Holman

On March 25, 1979, from 12 to 9 PM, members of the Boston Audio Society held a Turntable/Tonearm/Cartridge Test Clinic. The tests were based in part on the B&K application note, "Audible Effects of Mechanical Resonances in Turntables," 1977 (available from B&K, 5111 West 164th Street, Cleveland, Ohio 44142).

To obtain the playback systems for the clinic, we sent each of the 425 local members of the BAS a letter outlining the tests to be performed. To participate, they were asked to list the make of their turntable, cartridge, and tonearm. Seventy-five responded and, based primarily on their market representation, twenty-eight combinations were selected.

Sixteen different tests were performed which generated six separate graph paper plots. A Table Chairman was appointed to supervise each series of tests and to handle the massive amount of data collected. The chairpersons, in general, were already experts in the tests they performed, and had had previous experience in using the required apparatus. The use of the test equipment was donated to the clinic by BAS members. The procedural outline and the listed results are primarily the work of the Table Chairmen.

To make it easier for you to extract information from the tables and figures which follow, we will list the units along with their sample numbers on the reverse side of this page. You may want to remove this sheet from the magazine and hold it next to the appropriate page to help you translate the sample numbers, which will be used to identify the individual units hereafter.

The first stop for each unit was the table where tracking force and tonearm geometry were checked. If necessary, the alignment was optimized. Then the cable capacitance was measured, and the resulting value was subtracted from the manufacturer's recommended load, where available. The difference between the two values was dialed in to the Apt preamp that was used for the frequency response testing.

Nine of the twenty-eight cartridges were moving-coil models: sample numbers 4 to 8, 11, 14, 15, and 22. Number 11 was a high-output model which was connected directly to the preamp. Number 14 was tested with the owner's experimental head amp. The rest were used in conjunction with a Verion step-up transformer.

Table 1 - Key to Sample Numbers

Sample	Turntable	Arm	Cartridge
1	Linn-Sondek	Mayware Formula IV	AKG P8E
2	Connoisseur	Mayware Formula IV	AKG P8E
3	Sony PS 8750	- - – – –	Audio Technica AT-15Sa
4	Denon DP-2800	SME 3009-II	Denon DL-103-D
5	Kenwood KD-500	Mayware Formula IV	Denon DL-103D
6	Braun PS-600	- - - - -	Denon 103D
7	Thorens TD-126C Mk. H	– – – – –	Denon DL-103S
8	Lux PD-121	SAEC	EMT
9	JVC JL-A40	- - – – –	Grado G-1
10	Ariston RD- 11S	Grace 707	Grado G-2+
11	Thorens TD-166 Mk H	– – – – –	NAD 9000
12	Thorens TD-125	Mayware Formula IV	Ortofon M-15E Super
13	Harman-Kardon ST-7	– – – – –	Ortofon M-15E Super
14	Empire 398	– – – – –	Ortofon M-15E Super
15	Linn-Sondek	SME 3009 II (damped)	Ortofon MC-20
16	AR Xa	– – – – –	Shure V-15 Type II
17	Dual 721	– – – – –	Shure V-15 Type III
18	Technics SL-1300	– – – – –	Shure V-15 Type III
19	Thorens TD-125C Mk II	- – – – –	Shure V-15 Type III
20	AR Xa (modified)	Rabco SL-8E	Shure V-15 Type IVG
21	Prototype	Prototype	Shure V-15 Type IV-G
22	Yamaha YP-D8	- – – – –	Signet TK-7E
23	Fons CQ-30	Decca International	Sonus Blue (old)
24	AR Xa	JH Formula IV	Sonus Blue (old)
25	Fons CQ-30	Mayware Formula IV	Sonus Blue
26	Harman-Kardon ST-8	– – – – –	Sonus Silver P
27	Philips GA-212	– – – – –	Stanton 681EEE
28	Kenwood 550	– – – – –	Stanton 881S

OCRed from printed copy - errors possible.

Tonearm Geometry and Tracking Force

Table Chairman and Author - Alan Sliski

Roughly one-third of the tonearms had correct or very nearly correct geometry. About one-half were moderately misaligned, meaning that they had an error in lateral tracking angle of from two to five degrees. The remainder required a major adjustment to bring them into alignment. Fortunately, we were able to take the time to align the cartridges rather than just check them, thus preventing misalignment problems.

Only two tonearm/cartridge combinations did not have sufficient range of adjustment to bring them into alignment. The headshell of the Technics SL- 1300 does not have sufficiently long slots to allow for us to make up the design error in the mounting of the tonearm to the turntable base, resulting in less than optimum overhang. The other case involved the EMT moving coil cartridge (with integral headshell) mounted on an SAEC tonearm which had an offset angle about ten degrees too small for its length. This was an alignment job for a plumber, *as the* tonearm itself would have to be bent. It amazes me that such high-priced equipment can be sold with such a glaring design error.

Two different fixtures were used in aligning the cartridges. One was my lunch-hour-in-the-machine-shop special, and the other was a prototype of the DB Systems alignment protractor, developed by Dave Hadaway. Our biggest problem was finding a straight, flat surface on the cartridge, to align with the protractor or fixture. It would be nice to see some sort of standard reference surface that would have the same relationship to the stylus on all cartridges.

To determine the cartridge tracking force, the Shure Precision Stylus Force Gauge was utilized. It is accurate to within 1/10 gram in its primary operating range (1/2 to 1 1/2 grams). Most of the cartridges were found to be within the manufacturer's recommended tracking force range.

We did a survey of turntable spindle diameters mainly to see how many fell outside of the standard of 0.283 inch plus 0.001 inch, minus 0.002 inch for 33 1/3 rpm recordings. We measured spindles from 0.276 to 0.284 inch; the smaller ones (Fons CQ-30) may give rise to record eccentricity, and the larger spindles (notably the discontinued AR turntables) are tight on some records.

Audibility of Poor Tonearm Geometry

Tracking error results from the difference between the way a record is cut and the way it is played back. The original stylus cutting the record maintains a constant 90° angle to the groove. A conventional tonearm carries the playback stylus across the face of the record in an arc, constantly changing its angle to the groove. The difference between 90° and any other angle is tracking error.

Distortion arising from tracking error can run up to 10% on loud passages according to Dr. John D. Seagrave (Audiocraft, December, 1956). This distortion is almost entirely second harmonic even for very large tracking errors. It arises as an alternating advance and retard of the reproduced signal which amounts to a frequency modulation of the signal itself. Fortunately, properly mounted tonearms have less than 1° of error and yield distortion results of less than 1%, far below other nonlinear distortion inherent in the stylus-record interface.

Cable Capacitance Measurements

Table Chairman - Dave Ranada

The cable capacitance readings were made with an ECD Model 100 capacitance meter. Its readings are accurate to ±0.1%, ± 1 digit.

In general, I found nothing out of the ordinary. A couple of units showed abnormally high cable capacitance, which might create problems in cartridge matching. Those turntables with unequal channel capacitance will require unequal matching capacitors across the phono inputs.

Several cable resistance measurements were made, but since no significant readings were found, the test was discontinued.

Frequency Response and Channel Separation

Table Chairmen - Augustine Antoine and Frank Farlow
Author - Frank Farlow

Frequency response and channel separation of each turntable/arm/cartridge combination were checked using the latest version of the CBS test record STR-100. The signal was fed through a specially modified Apt/Holman preamp to a UREI Model 200 chart recorder with a 2010 frequency-tracking module. The RIAA network of the Apt was removed to make it compatible with the test record, and the preamp's frequency response was certified accurate to ± 0.1 dB by Tom Holman. The Apt controls allowed adjustment of the cartridge loading according to each manufacturer's recommendation. Cable capacitance of each tonearm was measured and the appropriate capacitance (50, 100, 200, 300, or 400 pF) was added using the controls. Temperature in the test area was maintained between 74° and 78° Fahrenheit.

In condensing the 28 frequency response plots into the single graph shown (Figure 1), we omitted the right channel plots since interchannel differences were judged of little significance; we removed glitches attributable to the test record at 150 Hz and 5 kHz; we smoothed occasional spikes and notches clearly attributable to the test setup, retaining them as dotted lines; and, finally, we removed the 6 dB per octave attenuation below 500 Hz in the test signal by redrawing that part of each plot horizontally, in order to make deviations from flat response more easily readable. The vertical graph scale is 1/4 dB per division.

One of the most obvious features of Figure 1 is the rarity of a genuinely flat response. More interesting, perhaps, is the striking prevalence of a broad valley in the 1 kHz to 10 kHz range, followed (with the notable exception of most of the Shures) by a peak in the extreme high end. In nearly all cases, the bottom of this trough is 1 to 2 dB down and located between 4 kHz and 7 kHz. Generally, a valley of this sort will have a dulling ("mellowing"?) effect on music signals, reducing definition and "air," and since it includes the "presence" region -- about 1 kHz to 4 kHz -- it will also create the illusion of moving the listener slightly farther away from the sound source.

Glitches attributable to mechanical system resonances (cartridge/headshell connection, removable headshell/arm tube connection, the arm tube itself, bearing chatter, arm tube/counterweight connection, etc.) occurred in the low end of nearly half the curves. Also, one cartridge (#23) provided an unsolicited demonstration of what happens when insufficiently damped stylus meets record warp. (The insufficient damping might be due to age of the cartridge, a manufacturing defect, or both.)

To save energy and space, we have reduced the channel separation plots to two figures -- separation at 1 kHz and 15 kHz (see Table 2). At each frequency, separation figures for the two channels were averaged if they differed by 3 dB or less; otherwise they are both shown. "L" signifies left channel crosstalk when the signal is fed to the right channel, and "R" the opposite.

More often than not, we found gross differences in the channel separation figures for the two channels of the cartridges. At 15 kHz this is probably indicative of differences in stylus damping between the two channels, a matter of cartridge design and quality control. At 1 kHz, however, although it could indicate misalignment of the transducer elements in the cartridge, it is more likely to be a sign of cartridge misalignment (rotation about its axis as viewed from the front).

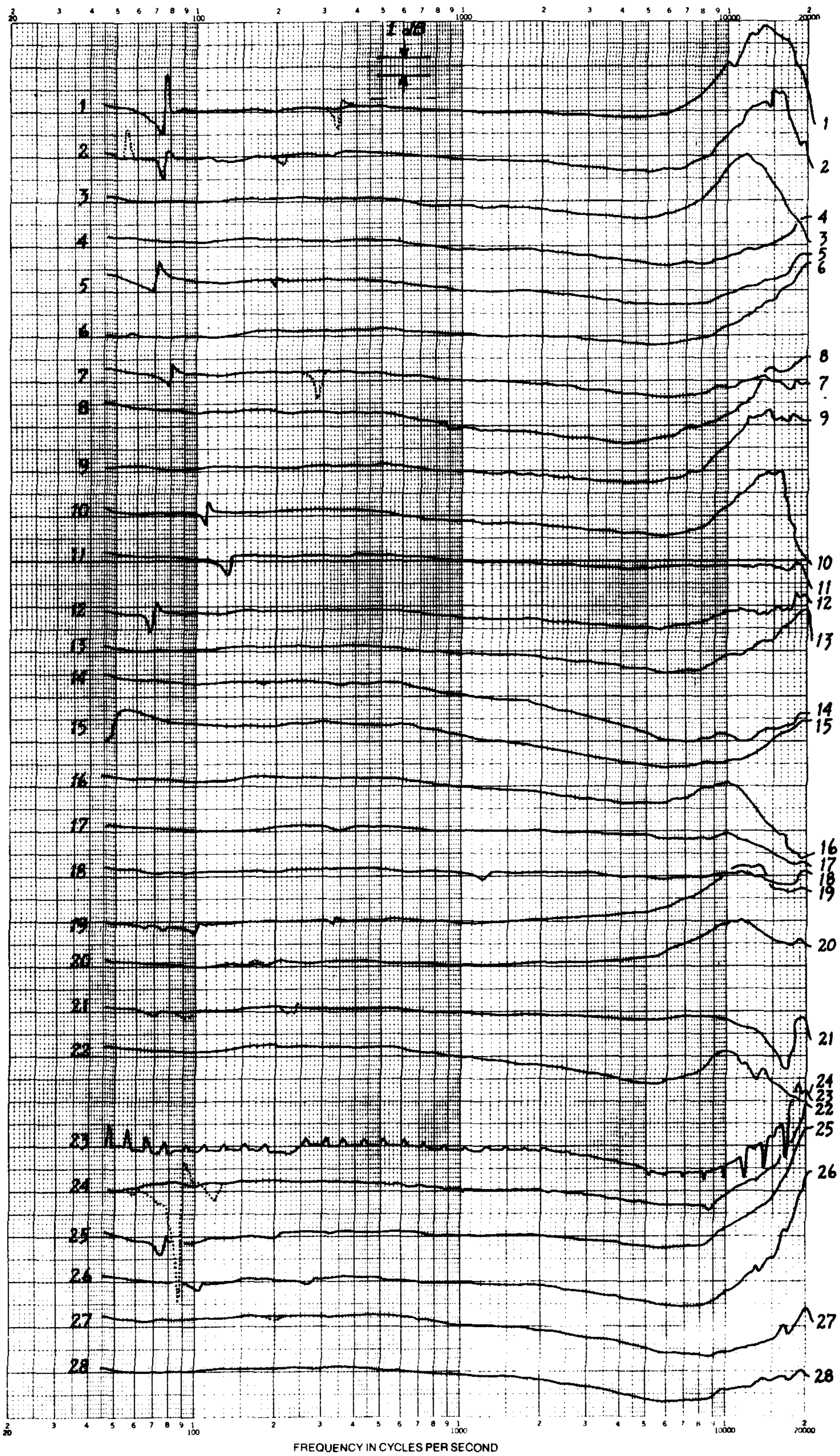


Fig. 1. Frequency response curves

Some of the frequency response plots show a rise at the bottom end. This can be caused by a tone arm/cartridge resonance which is either too high in frequency, or overdamped, so that it is lower and broader, extending its upper skirt into the audible range. Cartridges whose response is down 2 to 4 dB between 4 and 7 kHz, according to Shure representatives, have cantilevers which are too compliant or soft. The dip in this range is caused by the cantilever bending or twisting.

Frequency response peaks in the 15 kHz range are mechanical resonances set off by the cartridge-record vinyl interface. Good cartridge designers strive to reduce the 15 to 20 kHz peak resonance mechanically and/or electrically. The latter is accomplished with some magnetic cartridges by loading them with the proper amount of capacitance. Moving coil cartridges rely entirely on mechanical damping to minimize the resonance problem because their source impedance is too low for electrical damping to be effective.

A Note About Record Wear

Even though the same record was used for the frequency response tests perhaps 30 times, there was no measurable degradation of the record when at the conclusion of the tests it was again calibrated using the same cartridge. This finding coincides with those of the Empire Scientific Corporation, which found that after 6,000 plays with an EDR. 9 cartridge at 2 grams, there was "... less than 1 dB of change in output at any frequency and at all four recorded levels (3.5, 5.6, 8.9, and 14.2 cm/sec)!" Electron microscopic examination of the grooves showed negligible wear with the RCA 12-5-105 test record. According to Empire, the JVC TRS-1005 high frequency test record exhibited less than 1.5 dB change in output all the way to 50 kHz. The crosstalk was also not affected.

The same results hold for the B&K QR 2010 test record which was used for the tonearm/resonance tests. After approximately 40 plays, the record was compared to an identical record which had been played fewer than three times. The only difference in the heavily used disc was a slight (1/2 dB) decrease in output at resonance.

The results of repeated playings of a record indicate that if a record is kept clean the playback frequency response will not be altered significantly, even if the record is played every few minutes as in our clinic.

Vertical Tracking Angle, Flutter, and Speed Accuracy

Table Chairman and Author - Tom Holman

Four tests were made at this station. The first measures the vertical tracking angle of the cartridge. The method used is to find the point of minimum frequency intermodulation distortion (FIM) generated by 400 Hz and 4000 Hz tones recorded at a 4:1 velocity ratio in the vertical direction (the specially cut test record was provided by RCA, to whom we extend our thanks). The output of the cartridge was connected to an Apt/Holman preamplifier, set to the L-R mode. This maximizes vertical sensitivity for the test. The output of the preamplifier drove a modified Meguro MK-667C wow and flutter meter which serves as a combination bandpass filter, amplitude limiter (to remove amplitude intermodulation distortion as a possible source of error), and frequency discriminator. The output of the frequency discriminator was monitored on a Hewlett-Packard Model 3580A spectrum analyzer. In this way the recorded results measure the first order : 400 Hz) frequency intermodulation sidebands on the 4000 Hz tone. The test record is cut at 15 angles in 4° steps, and at 8 angles in 2 1/2° steps. The point of minimum distortion corresponds to the playback vertical tracking angle.

The second test measures the DIN weighted peak-to-peak flutter (including wow) as a percentage. A Woelke Model 102B flutter meter was used for this test. This type of measurement has become the most widely accepted of the international flutter standards, being adopted by such American organizations as the IEEE and Consumer's Union. Flutter measurements to other standards, most especially JIS weighted (WRMS) and NAB weighted standards, do not correlate.

The audibility threshold for flutter varies with the modulating frequency, but this is accounted for by frequency weighting in the measurement system. Typical listeners hearing piano music played over high fidelity systems in listening rooms have an average threshold of 0.4%; however, the standard deviations are quite large, and for the most critical five percent of listeners the threshold is about 0.14% (Stott and Axon, BBC data). H. Saki has found that the ear is able to detect as little as 0.12% peak-to-peak flutter on a complex 5 kHz tone when the modulating frequency is 3 Hz.

The flutter reading on this test has been shown to be strongly influenced by the arm/cartridge resonance. If the resonant frequency of the arm/cartridge combination coincides with flutter from the motor, for example, the flutter will be exacerbated. For this reason a third test was conducted: a measurement of the flutter spectrum to identify possible sources for the flutter.

For this test the output of the wow and flutter meter was connected to an external 1/3 octave sweeping filter which scans the frequency spectrum from 2.5 to 250 Hz. The filter output was connected to a chart recorder for a permanent record.

A fourth test measures the sensitivity of the system to speed variations induced by line voltage changes. The line voltage range for this test is 105 to 127 Vac.

Results

The average VTA of the twenty-eight cartridges tested was 25.6 degrees. These results are similar to those found by Bauer in the early '60s, when he found the VTA of most cartridges too high (average 28°) to match the then current 15° standard. With the change to an 18° standard, matters were moved in the right direction, but the average remains too high. A few notable exceptions occur among the data: cartridges measuring 20 degrees or less include all the Denons, the Audio Technica AT-15Sa, the EMT, an Ortofon MC-20 that was tracking at 1.75 grams but not one tracking at 1.45 grams (and not the Ortofon moving-magnet models), and the Stanton 681EEE (but not the 881S). In addition, a Signet TK-7E and an NAD moving coil were around 22 degrees. Shure models measured 28 to 32 degrees, and the Sonus models were the worst at 32 to 38 degrees. Note that tracking force changes the deflection of the cantilever, and therefore the VTA. Accordingly, Table 2 gives both VTA and tracking force.

The audibility of changes in VTA is still under argument. The Sonus cartridge in sample 24 had 0.8% IM distortion at 32 degrees and 4% at 16 degrees. Whether this is significant on musical material cut at normal levels is unclear. It does not predict the subjective effect claimed by believers in the critical nature of VTA, namely that too high a VTA produces a bright, harsh sound while too low a VTA makes the cartridge sound dull. It is nevertheless interesting that the moving coil cartridges, which have a reputation for smoothness and transparency, have as a class much lower VTA than most of the moving magnet designs, although VTA is not the only possible explanation.

The question is complicated by uncertainties in the technique of measurement. The conventional RCA and DIN test records can yield measured vertical tracking angles as much as 5 degrees higher than those obtained using either the geometrical method or the CBS STR-160 test record. According to Shure Brothers, "a variation of approximately 3 degrees is obtained from two different test bands on the DIN 45 542 test record."

Flutter Tests

Among all of the systems tested, the flutter rarely exceeded the 0.14% threshold of the 5% most critical listeners. However, a considerable number approach the threshold with measurements of 0.1 to 0.13%. It should be pointed out that the test record employed for making these measurements was very physically flat (being an aluminum-backed lacquer master). Record warps of the normal variety, peaked by the tonearm/cartridge resonance, could easily drive many of the systems over the limit for the most sensitive listeners.

The flutter spectra showed a number of interesting phenomena. Most tonearm/cartridge resonant systems apparently showed up in the flutter spectra, meaning that evidence for the contention that flutter is due to the turntable itself is woefully lacking. Some systems show sinusoidal

Table 2

Cartridge Loading, Channel Separation, Tracking Force, and Vertical Tracking Angle

Sample	Capacitance (pF) Cables	Preamp	Total	Channel Separation (dB) 1 kHz	15 kHz	Tracking Force (g)	VTA (degrees)
1	88	50	138	L16/R28	17	1.3	28
2	86	50	136	21	17	1.15	26
3	355	200	555	23	19	1.75	20
4	95	50	(145)	25	L28/R21	1.7	20
5	90	200	(290)	L30/R20	L26/R21	N. A.	19
6	195	200	(395)	L26/R22	L25/R18	1.4	16
7	N.A.	200	N.A.	21	L22/R14	1.8	16
8	64	200	(264)	24	14	2.5	18
9	85	300	385	L32/R24	19	1.75	25
10	N.A.	200	N.A.	L22/R33	13	1.7	N.A.
11	222	200	(422)	20	17	1.6	22.5
12	129	300	429	L27/R19	11	1.12	28
13	106	100	206	L34/R18	14	N.A.	28
14	157	200	(357)	L16/R35	18	1.75	20
15	N.A.	300	(N.A.)	L28/R21	L14/R23	1.45	24
16	166	300	466	L18/R33	L19/R10	1.3	28
17	175	200	375	L19/R34	L21/R12	2. 1*	28
18	70	300	318	L26/R21	16	1.6	28
19	309	100	409	L25/R18	L17/R12	1.2	28
20	174	100	274	23	12	23	12
21	136	100	236	21	8	1.125	32
22	239	200	(439)	L31/R24	22	1.25	22.5
23	292	100	392	L14/R20	L20/R24	2.0*	38
24	129	300	429	22	L26/R21	1.6	32
25	110	300	410	L20/R14	L15/R22	1.25	32
26	120	300	420	L15/R21	19	1.35	36
27	122	100	222	L30/R20	16	2.6**	16
28	78	200	278	L25/R20	16	2.2**	32

* With Disctraker
**With brush

flutter components. In the single case of the JVC turntable, this is apparently at servo-related frequencies, but it occurs at such low levels as to be negligible. The contention that direct drive systems have flutter problems associated with the servo system is not supported by these data. Other systems with AC motors do show sinusoidal flutter components; once again, these are probably at inaudible levels.

We had one system in which we switched the tonearm damping on and off to see what effect that would have on the flutter spectra. The result was only a minor change in the flutter, with some regions going down a few dB, and others up a few dB. The damping may, however, play a greater role with real records with their less flat surfaces.

<u>Turntable Speed</u>

Most of the turntables were within 0.3% of the correct speed at all line voltages. The most notable exceptions were numbers 2 (0.6% slow) and 14 (0.6% fast). The only models whose speed varied appreciably with line voltage were samples 23 and 25.

- To be continued in the next issue -

ALAN FIERSTEIN

The Equalization Myth

Balanced room ambience, a time decay situation, cannot be achieved by patchwork equalization.

MONITOR SYSTEM equalization is the most widely used method of compensating for control room acoustical faults. With a real-time analyzer (rta), the equalization process is fast, simple, and cheap. Unfortunately, it is also wrong, because it overlooks the basic physical factors through which rooms affect the sound of a loudspeaker.

Imagine a room with smooth, hard, totally reflective surfaces. A sound introduced into this room would never die away; it would just keep bouncing around forever. In an anechoic chamber, however, sound is absorbed almost instantly (as soon as it hits the first highly absorptive surface). These two rooms represent acoustical extremes-in real rooms sound absorption takes a finite time. This time varies for different frequencies. A carpet-lined room absorbs high frequencies quickly but the low frequencies are absorbed much more slowly. The way these reverberation times, or T_{60}s, change at different frequencies is what distinguishes one room's sound from another's. (T_{60} is defined as the interval in which sound pressure decreases by 60 dB after a steady-state sound has been abruptly shut off.)

FREQUENCY-DEPENDENT T60

How do these frequency-dependent T_{60}s affect the loudspeaker's sound? First the sound emerging from the loudspeaker reaches your ears directly. The sound then hits a surface which absorbs part of its energy, in relation to the absorption curve of the surface material. A plywood panel will absorb more energy from the low frequencies than it will from the high frequencies that impinge upon it. The carpeted room mentioned before has just the opposite effect, absorbing high frequencies. A number of reflections multiply this absorption characteristic many times and after the direct sound has passed, our ears still hear the frequency-modified reverberation.

In the carpeted room we are left with a muddy sound, since the high-frequencies were absorbed quickly. FIGURE I shows the result in a heavily carpeted room. The initial sound consisted of two tones of a low and a high frequency of equal volume. In the balanced room this sound has decayed to a faithful miniaturization of the original, but the heavily carpeted room has eaten up the highs and changed the spectrum from that of the original sound. Note that the high frequency wiggles are gone. We are left with a decayed low-frequency note only, hence the term "muddy sound." If you don't want a muddy reverberation, you must treat the room with materials that absorb low frequencies as quickly as high frequencies. If we had been listening to music. our ears would have heard the new notes, plus the muddy reverberation of past notes, giving the impression of added bass in the room.

EQUALIZATION FALLACY

Can we avoid treating the room acoustically and simply equalize down the bass in the monitor system? No. Equalization only affects the initial amplitude of the sound; it does not change the rate at which it decays. FIGURE 2 shows what happens when an attempt is made to equalize problems like this. Please note that numbers and pictures are exaggerated here for clarity.

In FIGURE 2 we see that the initial amplitude of the low and high frequencies are both 100 dB. The T_{60} of the balanced room is one second at all frequencies, so after one second both tones have dropped to 40 dB, which is essentially inaudible. Note that they both fell at the same rate from the same level and crossed the inaudibility threshold at the same time.

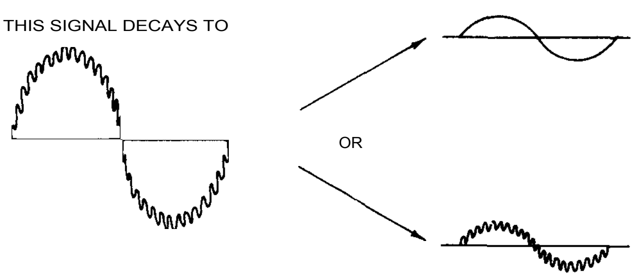
In the heavily-carpeted room with the muddy reverb, FIGURE 3, we have attempted to compensate by equalizing down the bass. The T_{60} of the bass is two seconds, and the T_{60} of the treble is one second. If we equalized down the bass 30 dB, it would start at an initial amplitude of 70 dB and fall 30 dB in the same time that the high frequencies would fall from 100 dB to 40 dB. Therefore both tones would again become inaudible simultaneously. But we have made the reverberation tonal balance correct at one point only, at 40 dB, which is useless because since the decay times are different at low and high frequencies. the tonal balance is changing throughout the decay period. Also, the direct sound is now totally non-flat.

By contrast, the balanced room of FIGURE 2 has a flat direct sound, an unchanged tonal balance for the entire decay period and both frequencies reach inaudibility together through the whole range. Clearly this is a much more desirable situation than that created in the heavily carpeted, heavily equalized room of FIGURE 3. The wonderful result of this balanced room is that a speaker that is flat in an anechoic chamber will sound flat at the mixer's ears, too, *without equalization.*

REAL-TIME ANALYZER

Contrary to popular opinion, a real-time analyzer does not display in real time, for if it did our poor slow eyes could not follow it. It integrates the input over a finite time period with a slow decay that makes observing re-

Figure 1. Signal decay under differing conditions.



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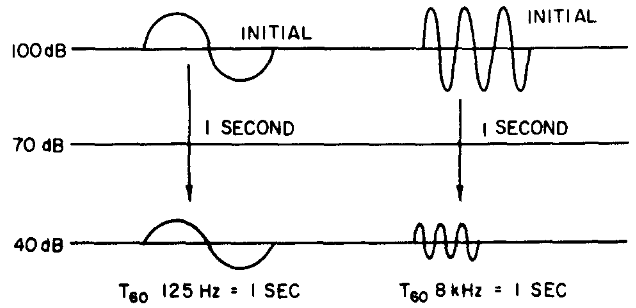

Figure 2. Balanced room decay.

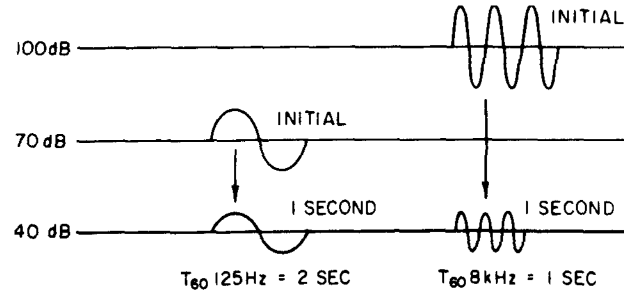

Figure 3. Decay in a room that is heavily carpeted and equalized.

verberation impossible. On the rta, the reverb of the room adds to the display of the pink noise, and a non-flat reverb characteristic will add more of some frequencies than others.

For example, on the rta, our carpeted room with the muddy reverb will add low end to the display, giving the impression that the initial sound is bass heavy and that equalization is needed. The rta's blind addition of signal and reverb is the root of the problem. Rtas are used with pink noise, which is a *static,* ***continuous sound,*** as compared with music and speech which are *impulsive* in nature. Impulse sound is defined by its initial level and time history,[1] and the rta simply adds level and time history together in a way that our ears do not. Our ears hear the effects of room reverb during the pauses of music and speech. Pink noise has no such pauses.

How real is this effect in actual control rooms? Of course, reverberation 20 dB or more below initial levels will not add significantly to the curve height on a rta, but the next 20 dB does. That the reverb is significant in affecting the rta's display is borne out by the fact that in a room with a T_{60} of 0.2 second, significant reverberant energy exists as close as three feet from the speaker. Obviously this depends upon other factors, most notably speaker Q. But when a speaker whose one foot frequency response of ±2 dB becomes ±12 dB at 8 feet (this actually occurred in a control room we measured) you can see that the room reflections have a pretty heavy influence.

This wild response was not caused by standing waves. This room was plagued by a non-uniform T_{60} vs. frequency curve. The ironic part of this story is that the speaker itself is obviously quite flat (±2 dB) and yet the room is giving this speaker a bad reputation (±12 dB). I wonder how many engineers are condemning their innocent speakersl

In addition to all this, equalizing the monitor system makes the important direct sound non-flatl Two rooms, equalized flat, can (and often do) sound different for this reason. Attempting to correct frequency-dependent time decays with initial amplitude equalization is like adding apples and oranges. This basic error occurs regardless of whether you equalize to sine waves, pink noise, or "full-spectrum" pulses.

ROOM TREATMENT

Properly treating a room is a complex job. What follows is merely a synopsis of common problems and solutions and is not meant to be a do-it-yourself guide to an acoustics diploma. Employing an experienced consultant is a wise decision if your room needs therapy.

Standing waves are a function of room dimensions and shape. Flutter echo is caused by multiple reflections between parallel surfaces. Room modes are room resonances that occur closely spaced in frequency and tend to reinforce their characteristic frequency when it is present in the program material. These problems are minimized by designing a setting with few parallel surfaces, ensuring adequate diffusion and by isolating room-resonant frequencies from each other by choosing optimum room dimension ratios. These are mentioned in Reference 3. Speaker placement can also affect standing waves.

A deep notch, characteristic of a high Q resonator, must be searched out to find out what surface is vibrating. Then stiffen it. These notches show up equally well with rtas and with sine wave reverb measurements. The need for symmetry and stereo separation must be also kept in mind. These problems, although often severe, lay the groundwork for the room T¡

With the gross problems out of the way, the absorption is added, subtracted, or modified to provide the desired T_{60} in each frequency band, usually octave bands. This can be planned in advance to an extent by using tables of absorption coefficients that have been published for various building materials. You multiply the square footage of each material by its coefficient at each frequency, and then you add up the total for each frequency and apply this to a T_{60} equation such as the Norris-Eyring. But since no one has published the absorption coefficient of your console, you'll need to take measurements of the T_{60} curve. Some may want a control room with a reverb curve approaching that of a typical living room, or perhaps a flat T_{60} vs. frequency curve is desired.

Finally, an equalizer can be used to fine tune the speaker system if its anechoic chamber response needs changing or if it was never tested in a chamber in the first place due to its custom design (often the case in studios). Usually the difference between one-foot and eight-foot frequency response curves points out the degree to which room reverb is playing a part, and here a rta is handy.

To sum up, control rooms are not equalizers or filters (though they may appear to be on a rta screen). They are time-decay absorbers. Do not correct rooms with amplitude changes (equalization); correct their T_{60} curves instead.

Equalizers are useful for fine tuning of speaker deficiencies that would show up in anechoic measurements, or for electrical modification of a recorded track, etc. When acoustical changes are not possible, as in many sound-reinforcement applications, equalization has the additional use of allowing increases of acoustic gain if applied properly.[4]

REFERENCES

1. Beranek, L. L., *Noise Reduction.* McGraw-Hill, New York, N.Y. 1960, pp. 145-151.
2. Rettinger, M., *Acoustic Design and Noise Control.* Chemical Publishing Co., New York, N.Y. 1973, pp. 27-28.
3. Everest, F. Alton, *Acoustic Techniques for Home and Studio.* Tab Books, Summit, Pa., 1973, p. 68.
4. Davis, Don and Carolyn, *Sound System Engineering.* Howard W. Sams & Co. Inc., Indianapolis, Ind., 1975, Chapter 8.