THE B.A.S. SPEAKER

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In This Issue

Volume 8 of the Speaker begins with this double issue. Small audio journals seem to be particularly susceptible to scheduling problems, and ours is no exception. We expect to reduce the time lag substantially in the next few months, so please bear with us. In the meantime this issue should give you lots to think about. The two meeting summaries alone provide an unusual variety: a thorough description of the new Carver preamp, an inside look at professional digital recording, some illuminating facts about electronics retailing, and a detailed and thoughtful look at the power amp/loudspeaker interface by Tom Holman.

The BAS has a reputation in some quarters as a defender of the "if you can't measure it, it isn't there" school, whose motto is, "It's All Just Frequency Response." In fact, the BAS has no opinions of its own. The Speaker is a forum for the opinions and experience of its members. Some of you are measurement freaks, some of you are subjectivists, most of you are skeptics, and all of you love to share your experiences and to argue about what they mean. The first item in this issue is a description of how to get something published in the Speaker. Please read it and send us your stuff. The sooner we get it, the sooner you get it.

Member James Mitchell has made an intrepid foray into the no-man's-land between the subjectivist and objectivist camps, and has sent us his dispatch. We have reports on Carver, Lux and Conrad-Johnson preamps, Collins Beagle on equalizers, Greiner on loudspeaker distortion and acoustic feedback, a heated exchange between representatives of Harman International and Phase Linear on power amplifiers, and design notes on transmission-line speakers and on a rumble filter that doesn't roll off the bass. And, of course, an extra-long version of "In the Literature" so you'll know what they're saying everywhere else.

The B.A.S. Speaker (ISSN *0195-0908)* is published monthly by The Boston Audio Society, Trapelo Road, Lincoln, MA 01773. Subscriptions are available to members of the society. Membership dues are $12 per year, October 1 through September 30 ($25 U.S. currency overseas, including air mail). $11.45 of the dues are a subscription to *The B.A.S. Speaker* including all issues of the applicable membership year. **For further information and application form, write to The Boston Audio Society, P.O. Box 7, Kenmore Square Station, Boston, MA 02215.** Second-class postage paid at Boston, MA. POSTMASTER: Send address changes to: The B.A.S. Speaker, P.O. Box 7, Boston, MA 02215.

Ins and Outs of the BAS

Articles

The Speaker is, always has been, and will remain a free and open forum for the membership. We edit for style, grammar, and spelling, but do not enforce any particular point of view. Contributions should conform to the style of the Speaker, with a title at the top and your name and state at the end. Each item should begin a new page and should be separate from other correspondence; drawings should be clear and neat, and please send originals, not copies. All material should be typed and double-spaced; this helps us enormously. Address contributions to The BAS Speaker, Trapelo Road, Lincoln, MA 01773.

Reviews

We encourage you to report your experiences with components, but we must remind you that subjective reviewing is fraught with peril for the unwary. This is especially true if the listening environment is unfamiliar; for this reason, listening sessions in dealers' showrooms are frequently misleading. Be sure to describe in detail the methods and controls used for listening tests, so that others may judge the degree of certainty of your conclusions. For other particulars, see "Articles" above.

Ads

Ads are a free service for the personal use of members only. The line between an active equipment trader and a dealer is sometimes hard to draw, but we try: commercial advertising, and non-hi-fi ads, will not be accepted. Ads should be of reasonable length, typed or neatly printed, on a sheet of paper separate from other correspondence, and mailed to The BAS Speaker, Trapelo Road, Lincoln, MA 01773. Include everything you want printed, and nothing you don't. If your name or address is not to be included, leave it out of the ad itself and put it in the upper right-hand corner of the page. We cannot honor requests to run ads in more than one issue; if you want us to run it again, you'll have to send it in again. There is a delay of four to eight weeks built into the system.

Monthly Meetings

The normal meeting time is 6 PM on the third Sunday of the month. We send meeting notices to local members only, so if you are from out of town you may check your BAS directory, find a local member, and get the information you need. Meeting notices usually arrive about one week prior to the meeting.

Directories and Constitutions

For a copy of the current BAS telephone directory or of the constitution and bylaws, send a self-addressed, stamped envelope (business size) to P.O. Box 7, Kenmore Square Station, Bos-

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The B.A.S. Speaker
STATEMENT OF OWNERSHIP, MANAGEMENT AND CIRCULATION
(required by 39 U.S.C. 3885)

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8. There are NO known bondholders, mortgagees, and other security holders owning or holding 1 percent or more of total amount of bonds, mortgages, or other securities.
9. The purpose, function, and nonprofit status of this organization and the exempt status for Federal income tax purposes have not changed during the preceding 12 months.
10. Extent and nature of circulation

10. Extent and nature of circulation	Average No. copies each issue during preceding 12 mos.	No. copies single issue published nearest to filing date
A. Total No. copies printed	1,567	**1.500**
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11. I certify that the statements made by me above are correct and complete. (S) Henry G. Belot, Treasurer
12 I hereby request permission to mail the publication named at the phased postage rates presently authorized by 39 **U.S.C. 3626. (S)** Henry **G. Belot,** Treasurer, B.A.S.

ton MA 02215, and mark it to the attention of Frank Farlow. Postage for the directory is 28 cents, for the constitution 15 cents.

Address Changes

If you move, send notice two to four weeks previously to Box 7, attention Frank Farlow. Returned Speakers cost the Society about 60 cents each and create extra work for Frank, so don't delay.

Speaker Staffing

Editorial assistance is always welcome. We are particularly in need of meeting summary writers, who are now paid for their work (see treasurer's report). Volunteers should write to the Trapelo Road address or contact Brad Meyer.

Treasurer's Report—In the Red and Healthy

At the end of the fiscal year, September 30, 1979, there were approximately $8, 400 in the BAS bank account. The following Operating Statement should be judged in that light:

BOSTON AUDIO SOCIETY
Operating Statement, Fiscal Year 1978
September 30, 1979

Sources of Funds		
Current Newsletter Subscriptions		$13, 165. 41
Back Issue Sales		1, 251. 00
Overseas Postage Income		697.29
Membership Dues		631. 40
Interest Income		592.86
Miscellaneous Income		182. 97
Total Income		$16, 520. 93
Uses of Funds		
Newsletter Production and Distribution	$10, 480. 07	
Honoraria	5, 223.20	
Overseas Postage Expense	849.79	
Meeting Notices	753. 00	
Administrative Expenses	517. 18	
Back Issue Production and Distribution	447. 78	
Turntable Clinic	94.23	
Decrease in Accounts Payable	67. 69	18, 433. 04
Net Deficit		$ (1, 912. 11)

This, of course, describes our fiscal year; that is, it pretends that we didn't have a penny in the till on October 1, 1978 when the fiscal year opened. In fact we already had collected several thousand dollars of dues intended to pay for 1978-79 expenses just as we collected dues in advance of the current fiscal year. If you adjust for the timing of these cash flows, our deficit for the membership year was about $800. When dues were established we chose a figure which would produce a slight profit if we continued to grow rapidly and an affordable loss if we stopped growing. The deficit is right in line with our estimates.

Honoraria

The constitution provides that a maximum of $1.00/member/elected officer may be allocated as honoraria to the elected officers. The usual practice has been to set aside the full amount but to distribute it to a broader group of people, including anyone who has made a sustained contribution to the success of the Speaker and the Society. This year thirteen members received honoraria. The officers (Jim Brinton, A1 Foster, Frank Farlow, Henry Belot) and the editor of the Speaker (Mike Riggs) received either $826 or $889 apiece. The rest ($904.20) went to people who maintained the computerized mailing list, arranged meeting sites, oversaw the printing and artwork of the Speaker, or wrote meeting summaries. Summary writers got $40 per meeting for the first three and $50 after that; we expect that this amount will stay about the same, although everything depends on the size of the membership.

Ads are free to members for their personal use only. No commercial, non-member, or non-hi-fi advertising will be accepted.

For Sale

*Dyna Stereo 120, $110; Dyna Stereo 80, $60; Soundcraftsmen 20-12, $175; Dyna PAT-4, $40; BSR turntable with AT-11 cartridge, $30; Heath EV4 decoder, $25; Heath 5-MHz scope, $40; two AR4x speakers, $20 each, plus shipping. D. White, (603) 888-6715.

*Custom pair of 7' high speakers in six modules, separate sand-damped cabinets for 15" CTS woofers, eight 8" CTS woofers, eight Phillips domes, eight 5" Pyles, eight piezos, and eight crossovers. Very efficient, effortless, and diffused. Come hear and compare. My cost, $950; will sell for $600, best offer, or trade. Need time delay, good turntable/arm, tuner, small (Rogers?) speakers, decent (Teac?) tape deck, Quad, Janszen, or other electrostatic speakers. Where are you Bill Moore? Don Wheaton, 209 Chelsea Street, E. Boston, MA 02128, (617) 762-4087 (leave message).

*Advent 201A, excellent condition, factory adjusted this year, $200; Dynaco AF-6 (kit built), excellent condition, factory adjusted to specs 7/78, $140; audio tubes (no rectifier) for ST-70, U.S. brands, used 50 hours, $12; DiscTraker, never used, $12. Shipping not included in prices. Jeff, (617) 325-6311 evenings.

*Dayton-Wright SPS MkIII, $250; Russound QT-1 patchbox, $150; Nakamichi 600 II, new, $500; Epicure 1 150 W/ch amp, $300; Advent 202 cassette playback deck, $125; Whisker record cleaning machine, $15; Decca Record Sweep, $10; Teac 2340 7" 4ch open reel tape deck, $500; Yamaha CT 800 AM/FM tuner, $300; J. H. Formula IV arm, $75. All prices include shipping and are negotiable. J. J. Thompson, 281 Warren Avenue, Kenmore, NY 14217.

*AR-9's, mint condition, $975. Will deliver free within 75 mile radius. Factory calibrated. Thomas A. DiPietro, 125 Gale Avenue, Haverhill, MA 01830.

*Sony TAE-8450 preamplifier. Peak-holding mirror-galvanometer level meters, many other special features, good sound. New $1400, will sell for $450. Call after 5 PM. (617) 296-2622.

*Cizek Model 1 loudspeakers. Accurate, well-respected speaker at a reasonable price, $165 each. NAD 3020 integrated amplifier with 3 dB dynamic headroom, Holman-based preamp, and highly stable operation, $155. Both new, prices negotiable. James Reach, 450 Beverly Drive, Oxnard, CA 93030, (805) 486-5707.

*dbx 119, $75; Stax UA-7 tone arm, $80; SME-III tone arm with second carrying arm, $190, or separately, $160 and $30. All in excellent condition with original cartons and manuals. Mike Pappas, (914) 686-7155 (days). (914) 234-7779 (evenings).

*Dahlquist DQ-10s, $600; Soundcraftsmen PE-2217, $300; Koss Pro-4A, $15 and Pro-4AA, $25. Karl F. Hessler, Center of Alcohol Studies, P.O. Box 969, Piscataway, NJ 08854, (201) 932-3576 weekdays until 4:30, (201) 249-9519 at other times.

*Back issues: Audio Amateur, 1970-79, $45; BAS Speaker, Jan. 1974-1979, $35. Jack Stevens, (617) 277-3164.

*Dynaco ST-400 power amp, 235 W/ch @ 8 Ohms, 350 @ 4 Ohms, $300; Wollensak 4765 Dolby cassette deck, same mechanism as Advent 201, with 50 cassettes, $125; SWTPC 20 W/ch power amp, $20. Trades possible. Fred Belsak, (617) 877-9164 6-12 p. m. weekdays and anytime weekends.

*Supex SD 901-e Super moving coil cartridge. High output, no pre-preamp needed, tight smooth, extended response. Tracks ERA-III record perfectly. Pristine condition, $40. Carlos E. Bauza, GPO Box 1220, San Juan, P. R. 00936.

Wanted

*Phase Linear 400, Ampzilla, or Crown DC-300 amp. D. White, (603) 888-6715.
*Photocopy of instructions for the legendary ADC Pritchard tone arm. Carlos E. Bauza, GPO Box 1220, San Juan, P. R. 00936.
*Mobile Advent 300/12 DC stereo receiver (negative ground). Working or non-working, but must be reasonable; extra arm mount for Micro Seiki DDX-1000 turntable (Serial No. 7033048); Zenith Trans-oceanic radio. Dwight L. Hedge, 135 Woodside Drive, Dyersburg, TN 38024, (901) 285-0615 (home), (901) 286-1512 (office).

More on the Great Preamp Myth

"Witchcraft" on Trial

At this writing, I am in Japan on an engineering assignment for one of our companies. While relaxing this evening, I decided to catch up on my reading and have just concluded Michael Riggs' article on "The Great Preamp Myth" (June 1979 Speaker). I commend his literate style, but I don't agree with a single word he says, especially the last sentence. Perhaps I believe in witchcraft, because there is no question in my mind that sonic differences between amplifiers and preamplifiers do exist and are detectable by trained listeners.

This morning's engineering session was a prime example of the hearing mechanism's uncanny ability to identify subtle differences in amplifier design. The first problem, which manifested itself as a masking of the more audible frequencies, was quickly resolved by supplying more current to the input stages. I won't bother the membership with technical gobbly-gook, but this problem exists in many audio components and can usually be remedied quite easily.

The other sonic problem was somewhat tougher, but we got a handle on it rather quickly. Judging from the circuit diagram, and having listened to several designs of this type using similar small-signal and output devices, I knew that the amplifier should have sounded spacious and clear, with no high-frequency "bite" and lots of air around the instruments. But this was not the case. Something was seriously amiss, because what we heard was just awful. Highs were too sharp (yet IM measurements were in the mud), and the balance between the middle bass, midrange, and highs was strange. In plain language, the amplifier wasn't musical.

While discussing the problem, I recalled a conversation I had had with Dr. Otala a month or two earlier, in which Dr. Otala (academician that he is) went into a long and enlightening discourse on why switches, relays, and so on create sonic anomalies. Remembering what he had said, I gave odds that the offending component was the amplifier's output relay.

After trying three other relays with different types of contacts, pressure, construction, and so forth, we concluded that all of them produced differences in harmonic articulation, balance, and most importantly, overall clarity. Two of the relays added stridency to the sound, which could only be attributed to dynamic distortion (probably intermodulation effects from the contacts).

Imaginary? Witchcraft? Not so! Pure scientific fact, even though we can only speculate as to the exact distortion-producing mechanism. Four trained listeners identified the sonic differences 100% of the time, and I'll settle for that.

Someday, if I have time, I would like to show you how much preamps (and amps) really do vary. I'll perform my witchcraft, and I can assure you that you will hear differences that are not the result of RIAA equalization or sound levels. We hear them every day, and we are perfectly sane, logical professionals who have been in the high-fidelity business for many years and have learned not to accept anything at face value. However, we also recognize that there is still a great deal to learn about high-fidelity design and how to conduct accurate listening tests.

To call sonic differences between preamplifiers a form of witchcraft is hardly professional or worthy of the editor of The BAS Speaker. -- Leon Kuby, Harman International

Mike Riggs Replies

Mr. Kuby's letter, which boils down to "I know there are mysterious sonic differences between amps and preamps, because I hear them, " begs the question. This is "pure scientific fact"? If so, where is the evidence? The point of the articles in the June Speaker was that stringent controls are necessary for valid listening tests. Without them, prejudices and extraneous variables will contaminate the results.

In the absence of detailed information on how Kuby's tests were conducted, it is impossible to assess their validity. However, I might suggest one obvious problem. Kuby guessed that the output relay was causing sonic problems. That gave him an emotional stake, however small, in the outcome of the experiment. Nothing unprofessional about that -- just human nature -- but unless accounted for in the design of the experiment, it could strongly bias the responses of all the listeners.

Recently, Dave Carlstrom told me a true story about a friend of his who was designing a tube preamp. Every time he modified it, he listened to it, and every time there was an audible improvement. After a series of such changes, he noticed he'd come full circle, back to the original design. It still sounded better. Just human nature.

Every properly controlled test conducted to date has yielded negative results. I am willing to change my mind (I already have, once, there having been a time when I would have agreed with Kuby), but only if presented with some firm, genuinely scientific evidence. Unsupported assertions won't do it.

Finally, I'm rather at a loss as to what I have done that is not "professional". Surely Mr. Kuby does not mean that expressing an opinion contrary to his, based on strong evidence from carefully controlled experiments, in a bylined article is unprofessional? Or does he?
-- Michael Riggs (Massachusetts)

And a Timely Comment from Carlos Bauza

The reprints in the June 1979 Speaker relating to "The Great Preamp Myth" were quite interesting. I can see that this is an issue of the gravest concern to golden ears, manufacturers, reviewers, and audio marketing people. It is also a natural for the BAS whose underlying principle is debunking myths for the benefit of the hobbyist.

Reading the editorials from the underground press, I find it easy to imagine that the political campaign-makers from down here have moved up there to a more profitable market. It is disturbing how a matter such as this can lead to mud-throwing and similar tactics. Whether this is right or wrong, it certainly resembles political campaigning. I'm not sure whether political tirades against an opponent are in the realm of truth, or in some realm which truth forgives.

All the reading I have done about hi-fi during more than fifteen years has led me to one principal thing: acquiring a taste for the neutral—for equipment that does not add or subtract, that does not alter what is supposed to be the true sound of music.

This is the main thrust of all hi-fi publications. In pursuit of this goal, they try to explain how certain components approximate the sound of music, but they stop short of recommending a purchase, heaven forbid, because that is a personal decision. Nevertheless, they produce a desire for components with certain specific qualities.

Many audiophiles defend accuracy as an absolute value. But this taste for the accurate is not innate. It is an acquired, learned preference. To give an example of the non-absoluteness of taste, consider that for three centuries half of all girls in the English-language countries were named Elizabeth, Mary, or Anne (boys were John, William, and Thomas). It must have felt inexplicably correct to be named thusly; during those three centuries one would have felt dislocated to be named anything else. Today, other names feel correct in the context of current taste: Jason, Brian, Jennifer, Nicole, Ryan, etc. Maybe three centuries from now we'll develop other tastes in the realm of audio. Sometimes I wake in the night pondering this contingency, and wonder if the accountants of audio publications do the same.

As for me, my taste is already formed. I seek a reasonable facsimile of the sound of live music. It does not matter to me whether preamps sound alike, nor does it matter who says it. I am my own authority. I choose equipment according to my taste, regardless of editorial opinions or test reports.

Yes, test reports are useful to me. The ones Julian Hirsch writes are meaningful to me because I can relate his results to my own use of equipment in the past, and can integrate them with present experiences. The underground press is also useful, but I read them with the salt shaker by my side.

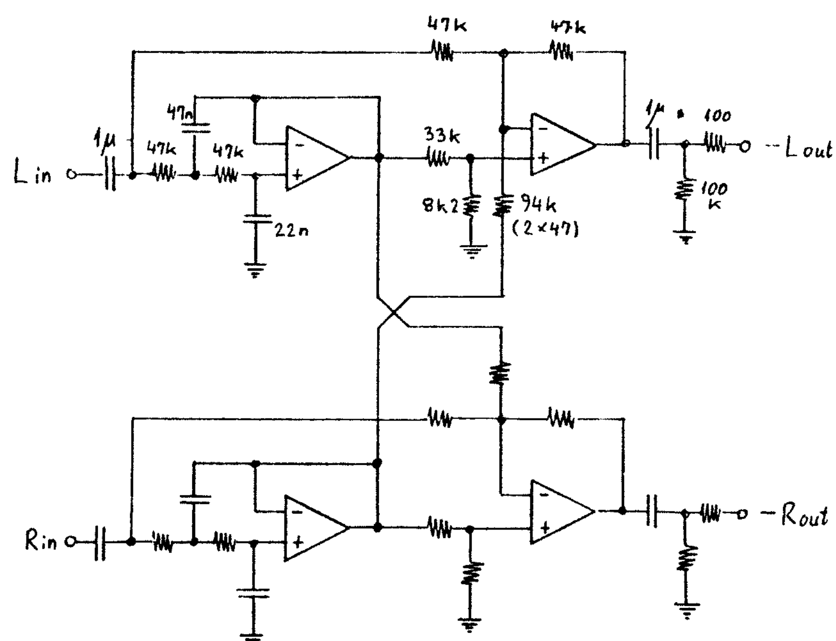
So, while the great preamp myth is being resolved, what you should do is to trust your own ears, form your own criteria, and act on your own personal feelings. But don't club people over the head with your taste. It's better to hear music with friends than alone.

-- Carlos E. Bauza (Puerto Rico)

A Rumble Filter With No Bass Cut

I would like to draw to the attention of BAS members an item from the "Circuit Ideas" department of the September, 1979 issue of Wireless World. It concerns a rumble filter that does not affect the bass response, inasmuch as this is frequently mono at very low frequencies while the rumble is mostly out of phase. Channel separation is maintained only down to 100 Hz, and below that the two channels are averaged. (There are reasons why this might be undesirable for some discs, but those that have had this process already applied should suffer no loss. -- Ed.)

Having studied the published design, I decided to build a simplified and improved version, making use of today's superior integrated op-amp technology. The diagram shows how a TL074 quad op-amp IC is used together with a simple matrixing system to form a rumble cancellation filter with near ideal characteristics. The TL074 exhibits a performance, in terms of distortion, slew rate, and bandwidth, that is hard to beat even using complex discrete designs. Expected figures will be around 10 μV noise, d < 0.002% (to 20 kHz) and f_u - several MHz.



It should be kept in mind that the rumble filter inverts the polarity of the input signal. If it is ever to be installed in a system where it may be switched in or out of service, inverting gain-of-one buffers must be used in order for the polarity convention to be preserved.

-- Jens Langvad (Denmark)

Jung and White Set Up Shop

Dave White and Walt Jung have started a modification business in response to requests to modify PAT-5s as outlined in their latest Audio Amateur article. Initially, they plan to specialize in PAT-5s, but other preamp and amplifier mods will be available shortly. For further details send a self-addressed stamped envelope to Dave White at 43 Royal Crest Drive, Nashua, NH 03060, or call (603) 888-6715.

Two Preamps

The Lux 5C50 Preamp

The search for a preamp that meets a listener's needs and wants is a monumental task, as detailed in the November Speaker's feature article. In my case, the present selection is my third in about ten years. During this period I have had a tube unit, which generally lacked bass detail, and a solid-state wonder that became annoying after a while. These previous trials, while interesting, seem to belie the seemingly logical fact that someone should be able to make a reasonable preamp that sounds good and works well for no more than about $300. Finally, I had to go into the higher price category to fulfill my personal requirements.

These requirements were always rather simple: accurate bass, smooth mid and upper frequencies, a good balanced sound across the frequency range, an accurate, well-built volume control system, an elimination of the bass and treble controls that never worked (or a truly flat position), no hum, and good switches. Yet the latest highly acclaimed units never seemed to be up to it. It was with interest that I listened to my friend's system and the changes in it over one and a half years, with the ever constant 5C50. Here is a unit that has no major backing by the exotic magazines, no hype, and superb sound and manufacture and is functionally useful.

It is always my aim to keep a system well balanced. The demands of the 5C50 are severe. This certainly is why some preamps and amps are not properly perceived by exotic magazines and one-time users. This preamp is very demanding and has required both cartridge and speaker upgrades to bring a satisfactory result. The possibility that a sonic flaw is caused by other components must not be overlooked. The Lux 5C50 (and 5L15 preamp section) are not going to satisfy everyone. Listening being a matter of taste, those who want a certain sound must find a suitable electronic ally. The sound of the Lux is one of neutrality, perhaps almost sterility, striking clarity and depth, the nearest to perfection I can imagine.

Reviewers always try to describe the sound of a product. They give numerous comparisons and cross checks. Can I do likewise to this product? Probably not. The bass is as distinct and clear as your cartridge and amp combination, in my case the Satin M18BX and JBL Se400. (The system as of this writing has also a Rabco SL8E and the newly improved Genesis II speaker systems, which replace old Genesis Hs.) The Satin is a fine cartridge in that bass transients in particular sound undamped and the mid and upper frequencies are smooth. When the recorded note stops, the sound stops instantly. This is demonstrated well by Supertramp (Mobile Fidelity MFSL1-005) and the "Dark Side" heartbeat (EMI SHVL 804), or the bass work on Brand X's "Passport" (PPSD 98019 LP). The Satin is a high-output moving coil and thus requires a setting of from 3 o'clock to max, depending on the record. Though there is a minor hiss increase above the 3 o'clock position, there is no hum even at max. One could conclude from this that maximum level is always required, but this is not the case. Not only does the volume control track to the lowest levels (the latest model has a continuous control), but the sound of the system remains well balanced at the lowest levels.

The 5C50's loudness control is a pleasant surprise: though a change is discernible, it is so slight as not to need correction. (This is because the volume control is turned up so high. -- Ed.) The Lux preamp is almost without tone controls. It has, as do several earlier Lux preamps, a "linear equalizer". The linear equalizer is a unique way of subtly altering sound sources. It has always seemed that someone could devise a simple control that would help material whose problems were brightness or dullness. This is done by tilting the frequency response around the 1 kHz point, decreasing bass and increasing treble by .6 dB at 100 Hz and 10 kHz, respectively, in the first uptilt position. The second position will do this by 1.5 dB at 100 Hz and 10 kHz. There are two reciprocal downtilt positions. The two uptilt positions are particularly useful for older records, such as Led Zeppelin II, now ten years old, or Reiner's "Spain. " The dullness can be reduced without offense to the ear. In fact, this preamp plus remastering could improve many old records without the harshness that sometimes results (Reiner's Vienna on Gold Seal).

As to switching functions, the Lux has the average amount -- Aux, Tuner, Mag 1 and 2, MC and two tape recorder loops. The MC transformer is an accessory that plugs into a socket on the back; I have not tried it. There are controls for cartridge load resistance, but not capacitance. There is a mute switch position and an infrasonic filter. The filter is inaudible in the 4 Hz and 12 o'clock position. There is also a high-cut filter. The audio of this DC-coupled preamp has the popular delayed relay turn-on to avoid pops. The only problem with this unit is that the preamp mute adjust pot (and probably the similar cart load pots) can become noisy, requiring a couple of turns to clean.

For those interested in instrument placement, the Lux is accurate. Certainly I do not possess the advantage of B&W DM7 or DM801 speakers, but can say that circular movement and so forth can be perceived, as well as correct stereo horizontal placement. The last two Charlie albums and imports Klaus Schulze X (Brain 0080.023) and Mirage (Island ILPS 9461), as well as Traffic's Mr. Fantasy (Island ILPS 9061), and Shoot Out (ILPS 9224) all provide adequate demonstrations of spatial effects and depth. A useful method of enhancing these effects is to have a pillow propped behind your head when listening.

In summary, I have presented a biased but hopefully useful view of the Lux 5C50. I believe the exclusive Lux op amps, in addition to the highest engineering values, account for the superb sound. I firmly believe it is the "sleeper" of the last two years and hope this will encourage others to objectively consider its merits. It is one of the best.

-- Norm Relich (Illinois)

Conrad-Johnson Preamp Mod

Owners of Conrad-Johnson preamplifiers prior to serial number 1251 should be aware of new modifications to the unit. Although no circuit changes have been made, the modifications substantially improve this preamp in two problematic areas: phono noise and the reproduction of depth and imaging. Changes in the cabling and an improved volume control as well as a new set of tubes are made at their service department for $70.

My unit was sent prepaid UPS from California and returned promptly a week later. The phono noise is now significantly lower than before, but frankly, I was never bothered by it except during quiet passages at very high volumes -- levels that I rarely listen at. Others I have spoken with noted substantial background hiss in their units at lower levels. This should no longer be a major complaint.

My principal enthusiasm for the change relates to the new volume control, which not only has better tracking characteristics, but apparently is responsible for the major improvement in depth and imaging. I say "apparently" because I'm not sure how much the new tubes contribute to these characteristics as well. Conrad-Johnson does imply that the volume control is responsible, and I have no reason to doubt them.

Owners of the older units are in for a very pleasant surprise, and I would strongly urge them to consider this modification.

-- Jeffrey S. Nelson (California)

Sonic Holography

The Carver C-4000 Preamp - A Preliminary Review

Wow! would be an inadequate way of describing what the C-4000 does to recorded music. The room comes alive, and the speakers seem to disappear because the sound appears to come from a wider and deeper stage. Instruments which were originally recorded up-front are close, while more distant instruments take on the natural ambience associated with their more rearward placement. With the Hologram, instrument localization is no longer confined to the space between the two loudspeakers. In my room the music sometimes appears three feet to the extreme left and right of the speakers and as much as seven feet behind the speakers and about three feet in front. I completely concur with the conclusions of Julian Hirsch and Larry Klein (Stereo Review, May, 1979). The Sonic Hologram produces a far more plausible sonic illusion of space and localization than is produced by normal stereo. (See my May, 1979, BAS article for a more detailed explanation of how the Sonic Hologram accomplishes its 3-D effect.)

Here are some additional brief notes on other features of the C-4000.

Delay. The 35 millisecond delay works well, while the 50 millisecond time interval seems too much for most recordings. The quality and positioning of the rear speakers are not critical. I am using ten-year-old ADC-404 speakers with extremely good results. They are placed to the left and right of my listening chair, about ten feet apart and facing upward on the floor. My listening chair and the rear speakers are positioned against the short wall.

The rear speakers are non-critical because they are not played at the same high level as in the traditional delay-stereo system. In fact, increasing the rear channel speakers above a barely audible level interferes with the imaging produced by the Sonic Hologram. However, when the rear speakers are played at a proper level, the "you are there" experience is enhanced.

Echo. This also feeds into the rear 25-Watt amplifiers. For my taste, it should never be used at all or only slightly with the driest recordings.

Peak Unlimiter - Downward Expander. This expands the recorded peaks and depresses the low level signals. It is intended to make up for the compression which exists on most recordings. It seems to work unobtrusively; when engaged, it is just audible but nice.

When Carver addressed the BAS briefly in September, he stated that he would not be able to produce the units fast enough initially to satisfy demand; therefore, he has selected only a few stores on the East and West Coast for distribution -- Cincinnati, eat your heart out. They are in the stores now.

However, I seriously doubt whether you will be able to appreciate the full effect of the Sonic Hologram in any dealer show room. In fact, even in my own home, it required about 20 hours of rearranging the speakers before optimal playback results were obtained. Most stores, due to the traffic demands, other equipment, etc., will not be able to devote the space or the set-up time required to properly demonstrate the unit. As with any high quality equipment, ask the dealer to let you borrow it for a weekend. Be prepared to reposition your listening chair and speakers until the loudspeakers disappear. At that point, you know the unit is working. (A Carver calibration-demonstration record will be available in April, 1980. The record will speed the speaker positioning process, which is like that for stereo but somewhat more critical.)

My room is about 25 feet long and 13 feet wide. Optimum location of my AR9s was achieved by placing my speakers near the center of the room and firing directly at my listening chair located on the short wall. The speakers are 2.5 feet out from the side walls. Even with a poor speaker location, the effect is still there, but when your room is set up correctly, the unit turns on.

An extra benefit of the Sonic Hologram is that it will encode even onto a cassette recorder. I have been making tapes for my friends who have been able to experience the 3-D effects in their own homes. -- Alvin Foster (Massachusetts)

Reservations About the Use of Equalizers

The following is a very belated response to the comments of John Puccio and Dan Shanefield in the December 1977 Speaker (pp. 12-14) in defense of the use of equalizers. Less directly, it is a response to the comments of Dan Shanefield in the December 1978 issue (pp. 1-3 at the end of the issue) on "The Great Debate" (a subject on which I hope to say more in the future), and of R. A. Greiner in the May 1979 issue (p. 16) on equalization.

John Puccio demonstrates a fundamental misunderstanding about the objections some hold against the use of equalizers. The reservation about equalizers is not an objection to alterations in frequency balance, but to the veiling of the sound caused by adding another link to the chain between the music source and the listener. Many eschew equalization for the same reason they avoid tone controls (or even the entire high level section of a preamp) -- to keep things as clean as possible. Accurate musical reproduction is a troublesome enough task without adding to the difficulties. This can be especially true with equalizers, because their designs have not yet reached the same high quality levels as those of preamps. In a revealing audio system, an equalizer can veil the sound by reducing the sense of air and three-dimensionality, as well as by affecting high-frequency detail and clarity. Mr. Puccio and Mr. Shanefield would have you believe an equalizer has only positive attributes; I and others advise you not to be blind (or deaf) to the fact that they also have negative ones.

What each person has to do is examine his or her own situation and decide if, on balance, the effects of an equalizer are harmful or helpful. Larry Hardin's comments in the August 1977 issue of the Speaker (pp. 11-12) are well taken. He had a situation where equalization did more good than harm, so he acted accordingly. But this will not always be the case.

Perhaps I have been luckier than most in my own room's acoustics and thus have found an equalizer to be harmful. I have owned both an Advent Frequency Balance Control (a rather questionable device) and, for the past few years, a Dayton-Wright SG Mk. II unit. I have used them to equalize my room, to make one stereo component sound like another, and to see what else they could do. But in the end I was always more satisfied when I switched them out.

Since my comments were published in the May 1977 issue (pp. 9-10), a friend who owns a local store specializing in equipment for band P. A. systems and small studios came over with his Shure 615 analyzer to check out my listening room. It proved to be a rather illuminating experience. My own one-and-a-third-octave band Dayton-Wright equalizer was cumbersome to use with the Shure device, so he brought along a Biamp Systems EQ/210 octave equalizer, which he (and apparently others in the recording industry -- see the review in the May 1977 Modern Recording) holds in high regard. Below 500 Hz there were the expected bass balance anomalies (the most striking in this case was a rather large hump around 30 to 40 Hz -- a problem frequency sweep tests had already pinpointed), and he gave up trying to deal with them. From 500 Hz up, the system had a better than average balance, which was similar to that suggested by Altec, as reported by Mr. Puccio, for making recordings in the home have the frequency balance of live performances (a curve that is -3 dB at 4 kHz, -6 dB at 8 kHz, and -9 dB at 16 kHz). Both channels were within a dB or so of this curve. So it would appear that I have been fortunate that my own system, unequalized, approaches the frequency balance Mr. Puccio suggests.

My friend used the Biamp equalizer to make my system's frequency balance flat according to the Shure device. The sound was hard and tinny. He then tried tuning by ear, reducing the equalizer's boost. But even in the "flat" position (I know, it may not really be flat) the sound was unacceptable, with a loss in three-dimensionality, openness, and air and with the addition of an objectionable metallic high-end coloration. The Biamp device was definitely more colored and veiled than the Dayton-Wright equalizer. Mine was about the tenth system my friend had tried to equalize, and he said this was the first where it had definitely hurt instead of helped. I would assume that the exception was because my better-than-average room required little help, plus the fact that my system was the most critical and revealing he had tried.

To put matters a bit more in perspective, the perceptibility of the negative qualities of an equalizer will depend on the associated equipment. I know from the experience of ownership that McIntosh, Crown (unless their newest designs are quite an improvement), and SAE electronics are not capable of revealing the qualities of sound that an equalizer affects, so an equalizer used with them should demonstrate only positive attributes. This is probably true with the vast majority of equipment available, so why not use an equalizer? But if one has very revealing equipment, such as that manufactured by Rappaport, Levinson, Hafler, Bryston, Audio Research, Apt/Holman, Precision Fidelity, etc., then an equalizer's negative characteristics will be evident. And each particular situation would have to be examined to see if, on balance, an equalizer would help. So try it, but do not always expect positive results.

I find Mr. Shanefield's argument that equalization does not do harm because the signal already has gone through various equalized stages to be somewhat curious (though not uncommon in audio). His argument amounts to: the signal has already been distorted (in its broadest sense), so what difference does a little more distortion make? I can understand the appeal of such an argument: I used to think that way myself. Unfortunately, in the real world it does not work that way. If you can hear the added distortion, then its effects are enough to be of concern. Following the above logic, no preamp or amp should take anything perceptible away from the signal, because its distortion contribution is so small compared to the rest of the chain. But experience in audio should demonstrate to any reasonable person that such is not the case. I wonder if Mr. Shanefield has stopped to consider that perhaps the average recording is so poor because of all the processing and equalization he describes to justify adding one more link to the chain. And perhaps the better recordings sound the way they do because they keep such processing to a minimum.

Mr. Greiner's comment that "those who deny the use of an equalizer are quite naive about the acoustical properties of rooms and the effect they have on sound, " (May 1979, p. 16) is as ill-considered as the other extreme, that "those who deny the problems of equalizers are owners of mediocre equipment who have never developed into critical listeners. " One familiar with acoustics should realize that some problems in a room can be dealt with mechanically. The proper choice of either absorbent or reflective furnishings and appointments can do much to control room reflections and the amount of high frequency energy. My room did not get all of its qualities by chance. Some of its characteristics were achieved in this fashion (part of my technical background is in acoustics). Though bass resonance problems are more difficult to deal with, they can be handled to a certain extent by the appropriate placement of massive objects (such as a full record rack or bookcase). The easiest way to handle bass problems is to be fortunate enough to have a very large room, but few of us do. If such mechanical means work, they are preferable to an equalizer. In the cases where such approaches are either impractical or insufficient to deal with a problem, then an equalizer should be considered. But for room resonance problems, one should have a one-third-octave equalizer to do the job properly. Unfortunately, these can be rather expensive. Recently, some perceptive manufacturers, such as Soundcraftsmen, have come out with equalizers with one-third-octave bands only up to the 500 to 1000 Hz region, where they are needed most, and wider bands for the rest of the frequency range. By such means one can get the proper tools at a lower price. For other types of problems, a parametric equalizer is a better bet, because of its great flexibility. I have had very good results with a borrowed Technics SH-9010 five-band-per-channel parametric unit. It is the cleanest equalizer I have had experience with, and I would recommend it with very little reservation.

One thing that Mr. Greiner did not point out is that perhaps the biggest problem with room reflections is the creation of virtual sonic images and delayed reflections, which can be very confusing to the ear in its attempt to recreate a precise sound field. Though an equalizer can average out the tonal balance resulting from the total energy of direct and reflected sound at one listening position, it will have no effect when dealing with imaging problems. These are caused either by the differences in arrival time of the different signal paths or to reflections strong enough to sound like distinct sources. These shift the perceived image, because of the Haas effect. Equalizers will not help with such problems at all, but mechanical approaches can.

So, if you have a problem an equalizer can help, certainly go ahead and try one. But be aware that in certain situations an equalizer can do more harm than good. Also, be careful not to abuse an equalizer by using exaggerated settings. This is a strong temptation, but you could end up destroying your speakers and perhaps upsetting your amp.

-- Collins Beagle (Virginia)

Just for the Record

Several months ago when I reported favorably on the Pioneer CTF-900 cassette deck I noted the wide range of its bias-trim control and speculated that it might even reach the high bias required for metal-particle tape. Now that I have finally succeeded in obtaining a couple of samples of metal tape I have tested that hypothesis. With Scotch Metafine the maximum setting of the 900's front-panel bias control is only a little too low -- close enough so that if the bias were boosted a little more via the internal servicing adjustments it probably would be marginally adequate. However, with Nakamichi ZX (which I suspect is representative of the "hotter" Japanese metal tapes, e.g., TDK and Fuji), the CTF-900's maximum bias is way too low to obtain either flat response or low distortion. Evidently there is no substitute for a recorder which has been properly engineered to cope with the demands of metal tape.

Even if the CTF-900's bias (or that of any other "standard" cassette deck) could be jacked up to the levels required for metal tape, you wouldn't want to do it, because once made, a recording on metal tape can't be erased without the aid of specially-designed dual-gap high-flux erase heads. The CTF-900 and other decks that I've tried can be jiggered up to record marginally well on Metafine (the least demanding of the metal formulations), but none can erase their own recordings, and neither can conventional bulk erasers. It seems that we're stuck with the necessity of buying new tape recorders if we want to use metal tape properly. Personally I'm going to wait until next year.

-- Peter Mitchell (Massachusetts)

An Amplifier Design Controversy

I picked up an article written by Terry Pennington of Phase Linear at the recent Chicago Consumer Electronics Show which literally made steam shoot out of my nostrils and ears. After reading Mr. Pennington's words of wisdom I sought out Dr. Matti Otala and had a long conversation with him on the subject. This letter to my fellow members of the BAS is a direct result of that conversation.

The first statement in question made by Mr. Pennington is as follows: "While this amplifier is not a pure class A design, crossover distortion, the only conceivable audible difference between class A and class AB, has been totally eliminated from the audible spectrum."

This statement is absurd! Crossover distortion is just one of many major differences between class A and AB. Other effects (all probably audible) include, for instance, secondary crossover distortion, transistor f_T variation distortion, C_{ob} variation distortion, symmetry unbalance distortion and common-mode conduction. All these distortion phenomena are severely increased by operation deeply into class B.

Another quotation from the article: "The very small amount of crossover distortion generated by the Model 300 appears only when the amplifier is driven by a very high frequency signal."

Crossover distortion as a mechanism is independent of frequency. If the power amplifier has a tendency to have common-mode conduction at higher frequencies (which is usually the case), this has a total effect of decreasing crossover distortion at high frequencies. The cited statement therefore indicates, at least to my way of thinking, that the dominant distortion mechanism in the Model 300 is not crossover distortion, but rather secondary crossover distortion which arises from the phase asymmetry of the output stage.

"Phase Linear amplifiers are direct coupled from input to output, having no capacitors or coupling transformers in the signal path. Their gain, however, is unity at DC. The reasons for this are as follows:

1. The human ear cannot respond to DC.
2. Loudspeakers are very easily damaged by DC, and do not reproduce it.
3. No audio sources produce DC.
4. No recording or broadcast media employed for audio purposes can store or transmit DC.

5. DC gain in a power amplifier requires the use of extensive offset controlling circuitry to temperature compensate the system causing excessive over-complication of design. It also leads to DC stability problems during high power output conditions at high frequencies."

This statement shows a fantastic misunderstanding of all matters concerned with low frequency response.

Granted, the human hearing mechanism does not respond to DC (at least as far as we know) and loudspeakers do not reproduce DC (this we do know). The question, however, is not DC in itself, but rather low frequency response. It is difficult, or in some cases impossible, to design an amplifier having adequate low frequency response, if feedback is applied in such a manner that the loop gain is unity at DC. This affects both frequency and phase response at low frequencies. Furthermore, employing feedback in this manner normally manifests itself in envelope distortion for signals which do not exceed the clipping level, and in gross recovery effects in the case of asymmetrical clipping. Both effects create spurious low frequency signals, which are in many cases severely audible.

The statement that DC amplifiers are difficult to design is perfectly correct. The problem, however, seems to be that there is no other way to design a high quality amplifier at the existing state of the art. There is also a rather lengthy statement on distortion which requires discussion because much of it is not true, at least in my opinion.

"Phase Linear amplifiers are all capable of distortion performance below what we believe can be heard by the human ear. Our distortion performance at low to mid frequencies is typically less than .005%. As input frequency is increased, however, distortion tends to rise to a high of .09% on our higher power amplifiers at 20 kHz but this is well below what one can hear. This only occurs at very high power levels at these frequencies. The nature of audio is such that there are always much higher power levels required at lower frequencies than at high, therefore, to achieve an audio waveform equal to 20 kHz at full power output would require severe clipping at low frequency.

"It must be remembered, when reading the specifications on power amplifiers, the distortion specification listed is the highest measurement made over the power range/frequency band of the product. Normally, the highest point will be full power at 20 kHz. Assuming the content of the measured distortion at this frequency to be mostly second harmonic, and that it measures even as high as 1%, this would mean there is being generated a 40 kHz component, at one one-hundredth the level of the 20 kHz fundamental. Considering that 20 kHz is inaudible, its distortion components, be they at 40 kHz, 60 kHz or wherever, should not be considered to be of great significance.

"Since no one seems to be able to agree just how much distortion is audible, good design practice dictates making it as low as possible at those frequencies where it would be audible, if large enough. Many amplifier designs, be they class A, DC, or whatever, claim exceedingly low distortion over a bandwidth of 20 Hz to 20 kHz. Very often these designs use very little negative feedback in an attempt to avoid slewing induced distortion, thus robbing the product of the benefits of this feedback. These amplifiers have, as a result, rather high distortion at mid and low frequencies, roughly equivalent to their 20 kHz measurements. Phase Linear uses sufficient negative feedback to insure absolute minimum distortion at the lower frequencies, while maintaining a sufficiently high slew rate."

The second paragraph, which deals with high frequency distortion, illustrates how little the author knows about the subject material. It is, of course, evident that the higher harmonics of a 20 kHz signal are inaudible, but this has little to do with high frequency distortion. High frequency distortion occurs when two, or more, high frequency signals interact in the amplifier non-linearities and produce difference-type intermodulation products, which fall directly into the audible frequency band. If the example presented in Mr. Pennington's paper is considered, it must be noted that the level of a 1 kHz signal produced by two high frequency tones of 19 kHz and 20 kHz, respectively, is the 1% cited in the example. Since the 19 kHz and 20 kHz signals were at the threshold of audibility it is easily understandable how dominant an audible effect the 1 kHz difference tone would have.

I'm certainly pleased to note that Phase Linear agrees that TIM does exist. Please read the next topic which appeared in their "Communique."

"The trend among audio manufacturers toward ever higher slew rates is an unnecessary objective. We at Phase Linear can state authoritatively that the slew rate of any stage of amplification need be only high enough to insure no slew limit contribution to the highest frequency of interest at its highest level. Therefore, when we state a distortion limit within the confines of a specific bandwidth, i.e., 20 Hz to 20 kHz, the slew rates must only be high enough to guarantee no interference at full power at 20 kHz. For an amplifier with a maximum power output of 500 Watts per channel and a maximum full power frequency of 20 kHz, the slew rate need be only 20 Volts per microsecond. A lower maximum power requires even less than that. <u>There will be no audio input ever applied to this amplifier which will cause it to slew</u>. We will agree that TIM does exist, however, no audio input can ever cause it in an amplifier meeting the above slew rate requirements."

The statement that there will be no audio input ever applied to an amplifier which will cause it to slew is grossly in error. Let us consider a signal generated by a super-fast phono pickup being fed into an audio system. Many studies (Lammasniemi, et al, Cabot) show that the nominal undistorted audio output does not contain slew rates in excess of the specified limit. However, record scratches, cartridge mistracking and even static can result in slew rates which are in excess of the given limit. These signals will be less audible if they would not be lengthened and accentuated by the amplifier entering slewing. Phase Linear's previous distortion discussion shows that the slewing behavior of their Model 300 is probably "soft" and highly asymmetrical (an assumption on my part). Therefore, it can be expected that the amplifier starts entering slewing (and TIM) at about one-tenth of the final slew rate value. It is, of course, clear that the amplifier will be useful only to the limit of one-tenth of the criterion given by Phase Linear.

And finally, the last quote from their bulletin.

"Phase Linear does not employ dual power supplies in its amplifier design for the following reasons:

1. The leading cause of power fluctuations in a power amplifier's energy supply is not transformer sag, which the use of two transformers would correct, rather the majority of fluctuations occur across the primary wiring, commencing at the breaker panel of the user's building and ending at the primary windings of the transformer(s) in the amplifier.
2. Power supply fluctuations of any sort have no effect on the sonic qualities of a properly designed power amplifier. Any change in the supply potential upon the output stage of an amplifier can have no effect on the output waveform, assuming, of course, it is not clipping. Only in the low-level input and differential stages of amplification can power supply deviations have any negative effect. These stages must have a regulated power supply to insure maximum power supply rejection. Without regulation, and with a dual power supply, co-channel interference, the purported advantage of such a system would not be as much of a problem as interference from each channel's output stage.
3. They are in direct conflict with cost-effective design, causing the consumer to pay more for a product than necessary, only to become a victim of runaway marketing inflation. The audio consumer should not be expected to bear the cost of features whose only reason for existence is to afford salesmen something irrelevant to talk about.

"We at Phase Linear feel that our dealers and customers deserve the correct information about high fidelity products. Hopefully, this communique will begin to assist us in achieving that goal."

Once again I disagree emphatically with Phase Linear's design philosophy. The leading cause of power fluctuations in a power amplifier's energy supply is not its primary voltage, as stated by Mr. Pennington. The best available power transformers, even when rated in a conservative fashion, have an average voltage drop from no load to full load of about 15%. On the other hand, a 15% voltage variation in a standard household line would be exceptional except in isolated cases.

Contrary to Phase Linear's statement, power supply fluctuations do have a strong effect on the sonic quality of even the best amplifier designs. The power supply can be considered as the heart of the amplifier. With the limitations of component technology today, it is not possible to design amplifiers which would have adequate supply rejection ratios without special precautions being taken to insure the stability of the power supplies. This is especially true for high frequencies, where the voltage variations of the supply lines are directly coupled to the main signal path of the amplifier via the collector-base capacitance of the output transistors.

I generally do not like to take issue with fellow manufacturers, especially in writing. However the Phase Linear report heated me up to a point whereby I had to express myself. If any of the members of the BAS accept or take issue with my comments I will be pleased to listen most attentively.

(Dr. Otala helped me prepare certain segments of this letter. I discussed the matter with Matti at great length because I consider him to be among the leading men in the audio field and respect his judgment in such matters.) -- Leon Kuby, Director of Product Development, Harman International

Terry Pennington Replies

I would very much like to comment on Mr. Kuby's opinions of our Communique which was made available to interested parties at the June '79 CES. Phase Linear believes in the premises put forward in that Communique and relishes the opportunity to discuss them.

To begin with, Mr. Kuby appears to assume that the results of his "long conversation" with Dr. Otala constitute the last word regarding the subjects in question. As far as I'm concerned, this is definitely not the case. It is Phase Linear's opinion that while Dr. Otala has put forward many theories and functional test procedures relating to audio hardware, the results of much of his earnest work has been to confuse the consumer and make an intelligent purchase of audio equipment very difficult.

The first subject Mr. Kuby addressed regarding Class A designs was a source of great amusement, not only to myself, but to all the engineers with whom I share an office. There is no secondary crossover distortion, transistor f_T variation distortion, Cob variation distortion, symmetry unbalance distortion, or other such contrived phenomenon in any of the AB Class amplifiers I've had the pleasure of knowing.

Mr. Kuby's statement that crossover distortion is a mechanism independent of frequency is partially true, at least when examining the situation on an open-loop basis. We do not, however, listen to or make comparative judgments on open-loop amplifiers. The differential nature of the amplifier, by way of corrective negative feedback, has more power to "fix" crossover notch at lower frequency than at high due to declining open-loop gain at higher frequencies. The assumption that the dominant distortion mechanism of the Model 300 is "secondary crossover distortion" is absurd. The primary distortion mechanism is the coupling of high level, high frequency signals from the output stage and associated power supply wiring to the front end of the amplifier. This causes only very small amounts of second and third order harmonic distortion, all of which lies beyond the audible spectrum.

Mr. Kuby's accusation that my understanding of low frequency response is lacking is an intolerable, slanderous insult. I must assume that he meant to use the term closed-loop gain, and not loop gain, when referring to applying negative feedback and affecting frequency response. Loop gain will be maximum at DC (that's why my Model 300 has only two or three millivolts of DC offset), however, closed-loop gain is unity (also contributing to the low offset figure). Phase shift in the case of the Model 300 is fourteen degrees at 20 Hz, which contributes nothing to perceived fidelity at low frequencies. Amplitude response of the system is flat to within .25 dB at 20 Hz. The envelope distortion mentioned by Mr. Kuby is a distortion which occurs exclusively in amplifiers employing DC gain. As to "gross recovery effects in the case of asymmetrical clipping," I can only say if your amplifier clips at any time while listening to music, either turn the level down or procure a higher power amplifier. (Recovery time is so short as to be unobservable, by the way.)

Mr. Kuby's reply to my distortion statements is a source of great curiosity to me. He has turned the discussion of harmonic distortion into one of intermodulation distortion. Hardly the same subject, but certainly one which I enjoy discussing since I personally dislike listening to intermodulation products much more than harmonic phenomena.

How Mr. Kuby turned the discussion to one of intermodulation amazes me, especially since he takes the liberty of taking a shot at my competence in the process. It is possible to draw a mathematical parallel between the two types of distortion, but in doing so, one finds that intermodulation distortion must be greater than or equal to harmonic distortion, which in my experience is never the case. So much for mathematics. The now famous "two tone intermod" test using two high level, high frequency signals is of value only if a realistic level is maintained with these tones. All amplifiers will intermodulate these test signals if they are of great enough magnitude to cause slewing in the amplifier. Therefore, one must not merely drive the device under test to a level just below clipping with the test tones, measure IM distortion, and judge the listenability of the product on this basis. It is simply not relevant. I personally do a swept two tone test of all products using Bruel & Kjaer equipment to insure that no gremlins lurk in the devices which would not be flushed out by conventional means.

As far as admitting the existence of TIM, I must reiterate my position. Any time an amplifier is driven beyond its slew-rate limit, intermodulation distortion will occur; however, music or speech will never exceed the slew rate of an amplifier if said amplifier is capable of delivering full output voltage at 20 kHz with no evidence of slewing. Mr. Kuby's only argument to my statement centered on mistracking, ticks, pops, static, etc. He must have forgotten that TIM was originally a concept to explain the sonic variations between amplifiers during AB comparisons. I find it hard to believe these comparisons consisted of enough mistracking and ticks to justify the TIM concept. All things considered (especially the RIAA playback curve), I doubt any of these occurrences would cause even the slowest of amplifiers to slew.

The assumption made concerning the asymmetrical slewing nature of the Model 300 is in error, as are the statements concerning the expectations of the amplifier "entering slewing at one-tenth the final slew rate value."

Once again, in disagreeing with another of my statements, this time power line fluctuations, Mr. Kuby is incorrect. In a normal household situation, the tremendous peak currents demanded by a very large power amplifier are in some cases high enough to dim any lights on the same circuit by a great amount. The larger the house, the more common and severe the phenomenon becomes due to the length of 12 gauge wire between the service panel and the receptacle into which the amplifier is plugged. It is true that the regulation of a typical power transformer utilized in an amplifier such as the ones built by Phase Linear will totally reject fluctuations well over 50%, with only a loss of headroom, and no sacrifice of sonic quality. All I can say about high-frequency power supply variations being coupled to the signal path via the output transistors is that if this occurs, there is something drastically wrong with the differential capabilities of the amplifier's front end.

Unlike Mr. Kuby, I do generally like to take issue with fellow manufacturers due to the fact that the vast majority of statements made in the press via advertising, or technical articles, stress performance characteristics which are of absolutely no value to the consumer. Ultra-fast "low TIM" amplifiers are a waste of everyone's time and attention. Audio components should not color their input signal in any manner, unless one consciously desires some modification due to personal taste. I do not allow myself to become involved in discussions pertaining to comparisons between subjective qualities of electronic components, rather I compare components one on one to live material. I feel that I am qualified to perform this task because rather than being only an amplifier designer, I am also an accomplished keyboard musician as well as a flutist; and I hear live music two to three hours per day as a result of my practice sessions and performances. I am much more qualified to judge sonic purity than one who hears only occasional live music, and I am in the audio business only for the enjoyment of the final product, not for philosophical discussions concerning irrelevant, inaudible, theoretical occurrences not related in any way to music.

As a final note, I would like to continue this discussion further with Mr. Kuby, in person, at an upcoming meeting of the HAS, if the Society and Mr. Kuby would be interested. I very much doubt that a judgment could be made as to who the "winner" of such a debate was, but I would very much like to see, in person, the "steam shoot out of his nostrils and ears."

-- Terry Pennington, Senior Design Engineer,
Phase Linear

Transmission Line Design

Robert Fabris asks a question on p. 2 of the July Speaker. The tube length considered desirable in a transmission-line speaker is the length of the quarter wave of the free-air resonance of the woofer. For example, if a woofer's free-air resonance is 30 Hz, the wavelength is 37.5 feet. Thus, the length of the labyrinth should be one-quarter of that, or 9.375 feet.

Two questions arise: (1) How do you determine the wavelength of a specific frequency? Wavelength = 1125/frequency = 37.5 feet, so one-quarter of that = 9.375 feet. (1125 is the velocity of sound in feet/second.) (2) How do you measure the length of the labyrinth? By measuring the combined lengths of a line drawn on the center of the "pipe" sections.

The second part of Fabris' question regards the matter of area of the cross-section of the tunnel. It is generally considered that the "line" should start out with an area at least 1.5 times the effective piston area of the woofer, and gradually taper to no less than the piston area of the woofer. The piston area of the woofer (with the woofer surround subtracted) is πr^2 (i.e., the area of a circle).

The Webb transmission line design ends up with a final duct area of about half what's considered desirable. The rationale for this, I believe, is that the woofer cone will be better damped below the system's resonance frequency. This design was refined during several years, so the particular dimensions are the result of trial and error.

The idea of the quarter wavelength is that at this dimension the wavefront bounces back down the tunnel and damps the woofer when it needs it the most. One drawback of calculating this in cookbook style is that inserting the woofer in any cabinet will raise its resonant frequency, and the true system resonance will be inaccurate right at the drawing board, before any wood has been cut. So, how does one determine what the system resonance will be? This way, one would have a more accurate, and desirable, damping of the system's resonance, not merely of the woofer in free air. If you know of a formula, please let me know.

There is a variation of the transmission-line design called the Daline. This was described in Hi- Fi News a couple of years back, but unfortunately, I have not been able to get a copy of the original article. There is a subsequent article in the same publication, titled "Daline + B-110," which is a construction project available from British purveyors.

The idea of the Daline is to build the initial compartment behind the woofer of a certain volume, which, if sealed, would yield a compliance of the trapped air equal to the woofer surround (this is really the design of an infinite baffle enclosure). But, at the back of this enclosure one adds a tunnel of certain dimensions, which is supposed to yield results better than standard transmission lines.

If you can get the original article describing the criteria behind the Daline, please furnish a copy. I think the subsequent article does not give enough information for one to attempt a design for a woofer of one's choice.

I am very partial to translines, because I have heard those wonderful IMFs, and because I like my Webb speakers so much. But many authors say that Thiele alignment in a bass-reflex design is a truly scientific and better way of loading the woofer. I tend to agree on paper, but those IMFs in the flesh are really something.

Another plus for transline design is that the impedance curve of the woofer can be truly flat in a properly designed system. I am led to believe that all bass-reflex designs have an impedance peak at the system's resonance. I really don't know if this is true. But you can juggle around with the length and area of the transline, and experiment with density and placement of the wool stuffing to get a flat impedance curve. The catch? One is practically re-designing the system each time a change is contemplated along these lines.

So, transline design can be thrilling at the drawing board, but at the work bench it can be murder. Before you make up your mind on transline vs. Thiele-aligned bass-reflex, go listen to IMF translines, or to the Webb design. -- Carlos Bauza (Puerto Rico)

Loudspeaker Distortion at Low Frequencies

It is with some trepidation that I bring up the issue of distortion in loudspeakers, but I really feel that this issue has not received enough attention. Loudspeaker manufacturers seem to think that if they do not mention distortion it will go away. The loudspeaker is by far the poorest link in the reproduction chain for all forms of distortion, and this includes nonlinear distortion, frequency response distortion, phase distortion, and time distortion. Compared to the electronics of a system, loudspeakers are horrible. As the recent directory issue of Audio shows, there seem to be a thousand different loudspeakers, only a few of which have the quality and loudness capability to approach realistic sound levels with the low distortion levels exhibited by even very ordinary electronic components. Clearly, this situation exists because most people want small, inexpensive boxes. So let us for the moment forget about what the market demands and discuss what loudspeakers actually give and can give in terms of distortion. Though there are many important design factors in making a good loudspeaker, I want to present data and make a few comments about one aspect of design: the distortion generated by the woofer at low frequencies and relatively high output levels.

It is fairly difficult to find data in the literature and in test reports that give consistent and comparable information on loudspeaker distortion. Some reasons are that it is difficult to make such measurements, there are no standards for frequencies and loudness levels, and manufacturers are certainly not going to give distortion figures, even when they know them, because the numbers are so high. So, with motto in hand, "non illegitimus carborundum," I have gathered as much information as I could find over the past five years and compiled just a few sample cases for this note. Most loudspeakers of a given size are about as bad as one another. The list contains a spectrum of commercial loudspeakers, but unfortunately, no samples of large professional systems, which are very likely much better than most of those listed. I have found data on about 200 loudspeakers, from which I have selected 19 for the accompanying table. Here is some of the bad news about loudspeakers. The distortion data is taken at about 40 to 50 Hz and at three loudness levels: 80 dB SPL, 90 dB SPL, and 100 dB SPL at one meter on axis. Only the second and third harmonic distortion levels are shown and are given in percent distortion with respect to the fundamental frequency. As is apparent, most loudspeakers will not reach a level of 100 dB SPL without such severe breakup and distortion as to make the measurement meaningless. Bear in mind that 90 dB SPL at one meter is not very loud at all. For a medium-sized room, this would correspond to just about that level with two loudspeakers going (stereo) at the listening position. This level is barely satisfactory for orchestral music on loud to average passages, and no allowance is made for headroom at all.

In some cases, I had to estimate the data and adjust it to insure consistency over the whole array of data. Thus, these figures should not be taken as precise, but rather used to get an approximate idea of how bad the distortion problem really is for loudspeakers at the lower end of the audio spectrum. An overview of the data seems to show that some of the better loudspeakers will deliver about 90 dB SPL at about 1 percent distortion at about 45 Hz. The distortion rises rapidly as the level is increased or the frequency lowered even a small amount. Because loudspeakers like amplifiers, should have some level headroom, so that the peaks are not totally smashed, it should be clear that the sample speakers listed do not for the most part reproduce sound at realistic levels without considerable distortion. Surveying this data is very discouraging. On the other hand, no data is presented, or available, for the really large systems that are now available. I have included distortion data for my own large system as a reference point to indicate what can be done if the low-frequency distortion problem is seriously attacked by brute force methods. In a number of A-B comparisons I have made between my reference system and some commercial loudspeakers, I have found the improvement in sound for my low-distortion system to be at least as great as one might expect from the measurements. But it should come as no surprise that the sound of a low distortion loudspeaker is better than that of most commercial loudspeakers.

I suppose most people have listened to a long string of high-distortion loudspeakers for so long that they think music is supposed to sound like hash when the level is turned up a little. In fact, it should not, and it does not. Live music sounds just fine at high levels, and there is no reason that reproduced music should not also sound fine. In fact, it does when the distortion from the loudspeaker is kept well below one percent. When the distortion in the system is lowered, it is possible to raise the listening levels to realistic values and still be pleased with the quality of the reproduction. I see no reason why distortion in loudspeakers should be any less important than that in the rest of the chain of reproduction. I am fed up with individuals who claim they can hear the difference between a power amplifier with 0.05 and 0.01 percent distortion while ignoring distortion that is 10 to 100 times greater in the loudspeakers. For the most part, I believe these people are not saying anything about the quality of the equipment, but rather are simply giving their opinions about what kinds of distortion they like.

In looking over data on hundreds of loudspeakers, there seems to be no good correlation between low distortion and design factors, such as acoustic suspension, open box, ported box, and the like (with the exception that transmission lines are much worse). There is a definite correlation between piston size and low distortion, however. Generally, the larger the piston (total piston area), the lower the distortion. Even this correlation is not perfect, because some manufacturers have demonstrated that they can make large, but very poor loudspeakers.

I have wished for years that designers would pay more attention to lowering the distortion in loudspeakers than to making them small for the sake of smallness. It appears that there is some hope in the recent appearance of new and technically improved designs in some of the larger systems. Though even the best of these has considerable distortion, they will start to educate the ears of the listeners to the higher quality that is possible with such designs.

I also hope then one day some brave loudspeaker manufacturer will meet the challenge, and have the nerve, to publish meaningful data on his loudspeakers. This would include items such as a meaningful power rating, distortion across the frequency spectrum (at several levels), efficiency, frequency response, phase response, and other information of the kind now published for power amplifiers. Perhaps then the quality of the loudspeakers would improve and someday even come within a factor of 10 of the best electronics.

(Ed. Note: Recent spectrum-level data taken during actual symphony concerts shows that the maximum levels in the one-third-octave bands at or below 50 Hz, which occur during works with loud bass drum such as the Tchaikowsky Sixth Symphony or the Rite of Spring, are typically around 94 to 96 dB SPL at the podium and 87 to 89 dB in the audience. The data will be published in the Speaker in the next few months. As for data from manufacturers, AR and Allison Acoustics, to name two, have for years been publishing frequency response curves for both individual drivers and systems, as well as distortion data at low frequencies as a function of level.)

Table of Harmonic Distortion for Selected Loudspeakers
at Low Frequencies
("-" indicates that the specified level cannot be reached)

	80 dB SPL		90 dB SPL		100 dB SPL	
	2nd	3rd	2nd	3rd	2nd	3rd
Hartly Zodiac	9.0	6.0			-	
Celestion UL 10	5.0	1.5	8.0	30.0	-	
Jensen 530	2.0	0.8	10.0	8.0	-	
Tannoy Berkeley	8.0	3.0	6.0	8.0		
B&W DM6	0.8	0.2	4.0	2.0	-	
B&W DM70	0.4	0.4	0.8	2.0		
Yahama NS1000	0.7	0.3	1.8	0.8		
Janis Sub-Woofer	0.3	0.2	0.3	0.4	0.8	0.8
EV Interface A	1.5	2.2	12.0	14.0		

	80 dB SPL		90 dB SPL		100 dB SPL	
	2nd	3rd	2nd	3rd	2nd	3rd
Duntech DL15	0.2	0.3	0.8	0.3	0.8	1.4
Audioanalyst A100X	5.0	0.6	11.0	2.0		-
Dynaco A25XI	4.0	4.0	14.0	10.0		
Fisher ST-425	1.3	0.3	1.5	1.5		
Hitachi HS300	1.5	8.0	13.0	8.0		
Philips HR532	4.0	2.0	10.0	8.0		
Avid 102	0.8	0.5	10.0	8.0		-
Technics T400	2.0	0.8	8.0	1.0	14.0	14.0
Nakamichi Reference	1.5	1.0	10.0	1.8	10.0	4.0
RAG Summaphon	less than 0.1		less than 0.3		less than 0.5	

Note: The last loudspeaker is not commercially available. It is the author's personal reference system, using four JBL LE 15 A drivers in an 800-liter box for each side. It will deliver 110 dB SPL at 40 Hz with 3 percent THD.

-- R. A. Greiner (Wisconsin)

In the Literature

Audio Amateur, Vol. 10 #4

*Editorial (p. 4): On the ease and excellence of today's homebrew designs, and news of a new magazine, Speaker Builder (subscription $8 before December 31, $10 after).
*Williamson 40/40 (p. 5): A $200 power amp kit conservatively rated at 40 W/ch.
*Peter Baxandall (p. 12): A leading sage (and author of the universal feedback tone control circuit) reminisces about his early days.
*Golden Ear vs. Meter Reader (p. 20): On the limitations of auditory perception, observer bias, and scientific method, suggesting that no one's conclusions are totally trustworthy.
*LED Power Meter (p. 23): A $140 kit. (The excellent Audio Technology 510 retails for about the same price, so this is evidently a kit for committed builders.)
*Location Recording (p. 32): Basic starting advice, sensibly recommending coincident miking (esp. ORTF). Ignores the potential ease and excellence of 2 or 3 spaced omnis and the fact that if you can't afford excellent mikes, cheap electret omnis are invariably smoother and wider in response than either cheap or medium-priced cardioids.
*Audio Aids (p. 40): Notes on making Heil tweeters, 3 dB ladders, etc.
*Integrex Dolby (p. 44): A test report on the encode/decode kit, revealing significant frequency response errors and poor low-frequency separation.
*Test Report (p. 48): A subjective review of the latest Jung/White PAT-5 mod -- mostly superb.
*The Grounded Ear (p. 52): Wondering whether the excellence and success of dbx encoded discs will make it that much easier for pressing plants to get away with sloppy work and cheap vinyl.
*Classic Circuitry (p. 54): Schematic of the Audio Research SP-6; the power supply is more elaborate than the audio circuit. A footnote indicates that the SP-6A update included a change in the RIAA equalization.
*Letters (p. 55): Lots of them, including one about the availability of a design for an inexpensive randomizing (IC logic) A/B switchbox for double-blind comparisons, from the Detroit audio society -- the SouthWestern Michigan Woofer and Tweeter Marching Society.

Audio, October 1979

*Tape Guide (p. 6): Basic questions, dumb answers.
*Audio ETC (p. 8): Canby on assorted unattractive approaches to cataloguing large record collections.
*Behind the Scenes (p. 16): Bert Whyte surveys current developments in cassette decks.
*Audioclinic (p. 32): On whistles in FM via commercial cable.
*Equipment Directory (p. 34): Specs of about 2700 products. Unfortunately the directory is chock-full of errors, especially in the sections on amps and preamps. Audio categorized many of the specs columns in accordance with the official 1978 IHF amplifier standard, and a few manufacturers provided correct numbers. But many provided data measured under the old IHF standard, and Audio simply stuck the numbers into the directory column without checking to see whether they made any sense. For example under the modern IHF standard, the best possible phono S/N is around 80 dB and achieving 75 dB is good; yet the directory listings include many ratings of 90 to over 100 dB, reflecting the meaningless old practice of using a short-circuit input. And in the majority of listings, the "IHF IM" distortion is actually the much less meaningful SMPTE IM distortion stuck in the wrong column; the IHF IM, which might be embarrassing for some manufacturers to publish, has never been measured by most of them. It's too bad no one at Audio felt that the magazine's responsibility to its readers might include even an elementary attempt to check the validity of the directory listings. (Of course in addition to the systematic wholesale errors there are also lots of individual random mistakes in the directory's specifications, but these are to some extent forgivable in so large an enterprise.)

Audio, November 1979

*Behind the Scenes (p. 6): Exotic amps and speakers seen at the June CES.
*Audio ETC (p. 16): Canby goes to a concert.
*Video Scenes (p. 38): New VCRs, and an incidental comment that a Matsushita duplicator can dub VHS and Beta 3 or 4 hour tapes in 5 minutes!
*Sound Images (p. 44): All about Jim Metzner's "sound magazines" with accompanying photos.

*Directory of Tape and Ancillary Equipment (p. 54): A one-page listing of available blank tapes which has been blown up to waste eight pages, plus scattered notes on accessory products.
*Equipment Profiles (p. 80): Pioneer TX 9800 tuner (a rave review, superb performance in every test). Acoustic Research AR-90 speaker (outstanding dynamic linearity and power-handling, smooth response, shallow depth imaging, some treble harshness). Ortofon MC-30 moving coil cartridge and T-30 transformer ("MC-30 is the top moving-coil cartridge available today," transformer is very good but the Audio Standards MX-10A head amp is better; Fig. 2 is missing, replaced by a reprint of Fig. 1). Bauman Research Pro-400 preamp (flexible cartridge loading, excellent performance; AC accessory outlets inadequate to power a large power amp; preamp is overpriced; photo of front panel is of wrong model). Crown RTA-2 1/3-octave real-time analyzer with scope display (superbly made, very accurate, flexible operation, wide range, but filter slopes are shallow and so not suitable for distortion spectrum analysis).
*The Bookshelf (p. 110): An excellent book on antennas and a five-hour aural history of radio.
*European Letter (p. 112): Notes on the Quad 44 preamp et al.

<u>Audio Engineering Society Journal, September 1979</u>

*Large-Signal Performance of Power Amplifiers (p. 638): A mathematical model of amplifier behavior is used to predict slewing distortion.
*Optimum Pivot Position on a Tone Arm (p. 648): The theory of Sansui's DOB tonearm in which the stylus becomes a "node" which remains stationary despite vibrations applied to the remainder of the arm.
*Digital Coding of Musical Sound (p. 657): Since linear PCM is costly and inefficient, this paper explores the distortions and noise modulation inherent in a floating-point scheme requiring fewer digital bits, and finds them acceptable.
*Aesthetic Microphone Techniques for Symphonic Music (p. 677): A Capitol/Angel engineer praises the virtues of coincident-pair X-Y miking and then goes on to tell how he uses "from one to three" coincident pairs plus up to a dozen accent mikes to record a symphony.

<u>Audio Times, October 1, 1979</u>

News items: Lafayette Radio, losing money, is getting out of mail order, eliminating stores away from the East Coast, and cutting the number of products carried. Sony is launching a six million dollar ad campaign to sell its tape. IBM has bought half of MCA's share of the Universal Pioneer Discovision videodisc manufacturing operation, putting IBM's financial muscle behind the marketing of laser optical discs for consumers and industrial users. The directors of the IHF have approved a merger with the EIA.

<u>Audio Times, November 1, 1979</u>

Pioneer's Series Twenty division is closing down; henceforth Pioneer's exotic audiophile products will be marketed under the Phase Linear name (Pioneer now owns Phase). Counterfeit TDK SA cassettes are being distributed in the New York area and advertised in the New York Times at $2.59 each; one clue is that the fakes come in a carton with all-brown printing (on proper TDK cartons the "C90" is green). Koss is marketing a $500 digital time-delay unit. Twenty-two makers of car stereo products have finally agreed on specification standards, mainly derived from the IHF specs for home gear.

<u>Gramophone (England), September 1979</u>

*Sounds in Retrospect (p. 539): Reviewing the sonics of selected recent discs.
*Report from America (p. 545): Notes from the June CES.
*Equipment Reviews (p. 549): Technics RS-1500 Isoloop open-reel tape deck (excellent in every way at 15 and 7. 5 ips, fair at 3.75 ips). Philips AF977 turntable (belt drive with PLL servo, excellent performance, semiautomatic arm, mediocre built-in cartridge). Tensai 1030 receiver (from Taiwan, good at its price).

<u>Gramophone, October 1979</u>

*Commentary (p. 728): Describing a gadget from Thorens for measuring rumble without a test record; while academically interesting and probably a useful tool for designers, it ignores the

important fact that the so-called "rumble" measured from a disc reflects the severity of the arm/cartridge resonance and so is a valuable index of record player performance.
*Harrogate plus Berlin (p. 733): Two hi-fi shows.
*Equipment Reviews (p. 738): Shure M95HE cartridge (good at its price). Micro-Seiki DQX-5b0 turntable (excellent performance, platter has enormous inertia, arm's effective mass is adjustable from medium to high to suit medium-to-low compliance pickups).

High Fidelity, October 1979

*The Middle-Aged Ear (p. 32): On whether response beyond 15 kHz is needed for audibly perfect reproduction -- many listeners (esp. young and female) can hear test tones beyond 20 kHz, but few audio professionals can.
*Equipment Reviews (p. 35): Dahlquist DQ-1W subwoofer and DQ-LP1 electronic crossover (excellent sonic performance; but low Q demands boundary reinforcement such as corner placement to avoid bottom-end rolloff). Allison Acoustics "Electronic Subwoofer" bass equalizer (very effective infrasonic and ultrasonic filtering, bass boost works as advertised, subtle but satisfying). RTR DAC/1 common-bass subwoofer with PS/1 satellites (satellites are okay but not really impressive at their price; subwoofer is superb, strong and very deep; but its response rises above 100 Hz, demanding a steep cutoff filter). Ohm Model N subwoofer (a good woofer for mini-speakers, but its bottom octave is rolled off; not a true "sub"-woofer). Empire EDR. 9 cartridge (measures bright but sounds smooth, tracks well, inductance is low, VTA is high). Nikko 819 receiver (undistinguished tuner, good preamp, shallow filters, pretty good power amp).
*Records to Judge Speakers By (p. 59): One man's selection.
*Subwoofers (p. 68): Victor Campos discourses on system resonance and Q, gives some buying tips, but doesn't deal with the fallacy of common-bass operation.

High Fidelity, November 1979

*Too Hot to Handle (p. 8): Qs and As.
*Equipment Reports (p. 36): Crown Straight Line One preamp with outboard phono preamp module (stripped-down design, no tone controls or tape dubbing, phono module brilliant in concept and outstanding in performance, no headphone jack, infrasonic filter is much too high at 33 Hz). Mordaunt-Short Pageant 2 speaker (smooth, neutral; note unusually high 3500 Hz woofer crossover). Series Twenty by Pioneer Model F26 tuner (state-of-the-art, better than available test gear, but lacks conveniences such as a high-blend noise filter and a signal-strength indicator). SME 3009 Series Ill-S tone arm (a slightly cheaper version of the Series 3, with damping an extra-cost option). Kenwood 801 amplifier (excellent performance, unsatisfactory tape-dubbing system). Audio Technology 510B LED power meter (fast, accurate, flexible calibration switching; outstandingly useful as a peak level meter for tape recorders; "B" version with green/red LEDs preferred to the all-red 510).
*Ambience and Space (p. 55): A six-way panel discussion on time-delay and other approaches to ambience reproduction. A clear and useful introduction to the subject (except for a typo, mixing ADC with ADS).
*Walter Legge (p. 63): Biographical notes on the great producer.
*Video Recording (p. 71): Present and forthcoming tape and disc systems.
*Prerecorded Video Tapes (p. 79): Why their technical quality is so notoriously variable -- caveat emptor.
*Discs That Vanquish Noise (p. 94): Reviews of dbx encoded discs.

Hi-Fi News and Record Review, September 1979

*Experimental Scratch Eliminator (p. 57): Design for a homebrew tick-and-pop suppressor.
*Soundings (p. 63): Notes on a visit to Thorens.
*Philips and the Mic/DIN Problem (p. 65): On a clever feedback volume control which yields enormous useful dynamic range in a mike preamp circuit.
*Audio Patents (p. 69): About designs for binaural records in 1921, lateral/vertical stereo grooves in 1927, and an induced magnet cartridge in 1918.
*Subjective Sounds (p. 71): Notes on high-priced cartridges.
*Reviewing and Consultancy (p. 73): Dealing with the prevalent conflict-of-interest problem of reviewers who are also paid consultants to manufacturers.

*Stereo Seat Postscript (p. 81): More notes on cross-angled speakers.
*Gramophile (p. 85): Interviews with Abbado and Accardo.
*Things I Hear (p. 91): An attack on the scandalous practice of ignoring the repeats specified in the score.
*Reviews (p. 131): Kenwood KT-917 tuner (the best tuner ever tested, outstandingly clean sound; best sensitivity obtained via the 75-ohm-only input). Yamaha T2 tuner (excellent overall; but the signal strength meter is easily saturated and the automatic IF-bandwidth switching sometimes makes non-optimum decisions). Hitachi FT-8000 tuner (pretty good for its price). Audio Pro TA-150 receiver (exotic one-knob all-electronic microprocessor control, hard to get used to, performance okay but not outstanding, all inputs and outputs are DIN sockets including speakers). Nakamichi 730 receiver with wireless remote control (no knobs, all controls operate by sliders or by internal motors, excellent tuner and amplifier performance, infrared remote control easy and effective). Discofilm and Diskmask record-cleaning films (expensive, takes practice to apply correctly and in optimum amount, film does not always come completely off the discs after drying; works superbly, strongly recommended for restoring old discs to playability).

<u>Hi- Fi News and Record Review, October 1979</u>

*News (p. 95): Past and present at AR.
*American Letter (p. 97): Nate Garfinkle surveys the resurgence of tube preamps, power amps, and modifications to old tube gear while raving about their euphonic sound, much preferred to the sound of solid-state components.
*Letters (p. 103): On reviewing/consultancy conflicts.
*Experimental Scratch Eliminator (p. 107): Assembling and calibrating the kit.
*Subjective Sounds (p. 111): Favorable notes on Mobile Fidelity records (which cost $30 each in England).
*Cassette Technology and Tape Developments (p. 117): A comprehensive survey of the improvements in many cassette tape formulations, an examination of the genuine improvements produced by Tandberg's new DYNEQ circuit (a high-frequency limiter which prevents the obvious squashing and IM caused by tape saturation), and a fascinating study of the Dolby HX headroom extender, found to be a dramatic success. Angus McKenzie discovered that during those moments when strong high-frequency energy is causing the HX circuit to reduce the bias, the high-frequency signal itself acts as a bias signal, thus keeping the overall distortion low!
*Quality Monitor (p. 125): Assessing the sonics of the best recent discs.
*Audio Patents (p. 129): On the expired patent for the Quad electrostatic.
*Gramophile (p. 135): Interview with Colin Davis.
*Reviews (p. 179): A composite review of five high-price turntables, preceded by a thoughtful assessment of the design features to be expected in high-quality disc players. Best of group: SME Series III arm on Monitor ET500 table, followed by Hadcock GH288-D arm on STD 305M table, Dual CS714Q, Micro Seiki DQX500, and Philips AF977. All found to suffer from feedback, the Dual and Philips from incorrect arm geometry, the Hadcock and SME from incorrect setup instructions, the Hadcock D from resonances and omitted damping (in contrast to earlier versions), the Dual from inadequate capacitance for the supplied Ortofon ULM cartridge.
*Three Mid-priced Speakers (p. 195): Castle Conway II (remarkably smooth). Celestion Ditton 332 (pretty good). JR150 (significantly better than the earlier JR149).

<u>HiFi Stereophonie (Germany), November 1979</u>

*The Production Participants of the Record Industry (p. 1532): As in the other articles in the musical section of this issue, the author is discussing cooperation between privately owned companies and state-subsidized opera houses.
*Producing Operas in Appurtenance (p. 1538).
*The Triumphant Campaign of the TV Live Opera (p. 1542).
*Record and Broadcast Cooperation (p. 1546).
*Test Reports (p. 1604): The Onkyo TX-20 receiver (excellent at its low price, especially the tuner section). The JVC KD-A8 cassette recorder (without Dolby and with other faults in conception, the B. E. S. T. system not always giving the best frequency response). The Revox BR 530 loudspeaker (top class, uncolored and powerful). The T+A elektroakustik LK-75, LK-100, and SK-80 loudspeakers, and the TSM puris 120, 92, 82, and 62 loudspeakers (very good, neutral and uncolored).
*Internationale Funkausstellung Berlin 1979 (p. 1638): A postscript, tangential tone arm revival, birth of the vertical turntable.

Modern Recording, October 1979

*While We Wait (p. 66): Favorable comments on dbx encoded records, Dolby HX, Tandberg DYNEQ, and Nakamichi Telcom.
*Reviews (p. 68): AudioControl 101 equalizer/analyzer (very good at the price, thoughtfully designed and laid out; minimum analyzer increment 2 dB, labeling hard to read). Spectro Acoustics 200SR power amp (good power/price ratio, but slew limiting and 4-ohm current limiting are noted). Teac/Tascam 80-8 1/2-inch 8-track tape deck (good design and control planning, good performance except for a bass bump, well set-up, a bargain for the small multitrack studio).

Modern Recording, November 1979

*Reviews (p. 82): Revox B77 tape deck (spectacularly good, head-and-shoulders above most other decks). AB Systems 730a single-channel tri-amplifier (3-way electronic crossover plus 3 power amps, clean and powerful into 8-ohm resistor, 4-ohm measurements omitted).

Boston Phoenix, November 13, 1979

The "Sound Ideas" supplement includes descriptions of a digital recording session, how sonic holography works, Boston Acoustics, BASF's tape factory, counterfeit cassettes, etc.

Popular Electronics, October 1979

*Stereo Scene (p. 20): A thoughtful review of ambience, time-delay, and Hafler ambience recovery.
*Reviews (p. 24): Scott Pro 100B speaker (smooth, wide range, efficient, wide dispersion). Pioneer TX7800 tuner (very good). Teac 124 Simul-Sync cassette deck (good, exceptional high-frequency headroom but limited top-end response).
*Audio Power Meter (p. 62): A $50 kit which multiplies output current and voltage to obtain "true" power.

Popular Electronics, November 1979

*Stereo Scene (p. 14): An insightful look at some of the ways in which standard specs don't fully describe sonic performance.
*Reviews (p. 22): Mitsubishi DT30 cassette deck (dual capstan drive, 3 separate heads with azimuth adjustment, limited-range bias trimmer, measures good, sounds great). Yamaha C-4 preamp (very flexible cartridge loading, good filters, MC head amp, flexible tone controls, ample headphone output, runs warm, measures great). Ortofon Concorde 30 cartridge (weighs 6 grams including headshell, response and tracking are excellent, distortion unusually low).
*Clipping Indicator (p. 75): A clever homebrew kit to flash a LED when peak audio levels approach the amp's power supply voltages.

Radio Electronics, October 1979

*50 Years of Electronics (p. 42): Some historical notes.
*Home Reception via Satellite (p. 80): Good information on antennas, low-noise amps, etc. for backyard satellite receivers.
*Headroom Test Generator (p. 96): Homebrew tone-burst generator for measuring IHF dynamic headroom.
*All About Microphones (p. 101): A basic introduction to mike jargon.
*Test Reports (p. 104): Tandberg 2080 receiver (expensive but superb; 4-ohm measurements omitted). Audio/Pulse Model 2 time-delay (good at its price; deep bass is rolled off).

Radio Electronics, November 1979

*Test Reports (p. 62): McIntosh MC-502 power amplifier (conservatively rated, superb performance). Onkyo 4090 tuner (excellent; tuning-lock correctly aligned for minimum distortion).

Radio Electronics, December 1978

*Six-foot TV Kit (p. 40): Description of the new Heathkit 6-foot 3-tube projection TV, which at $2200 may be a best-buy.
*Audio Amplifier (p. 55): A homebrew 60-100 watt/ch power amp employing bridged circuits, buildable for $100.
*Dolby HX (p. 58): How it works.
*Test Reports (p. 61): Sherwood 7210 receiver ("a little gem," a very good budget-priced unit).

Recording Engineer/Producer, August 1979

*Reverberation Overview (p. 50): On the nature of real reverberation and a comparison of various techniques for artificial reverb.
*Comb Filter Effects (p. 62): On the colorations caused by reflected/delayed sounds when they combine with direct sounds.
*Construction of an Echo Chamber (p. 73): Hints on making an acoustically smooth reverb room, some of them also useful in designing a listening room.
*Optimizing Control Room Reverb (p. 87): Measuring T_{60} time vs. frequency and applying the results to room treatment.
*Programmable Digital Reverberation (p. 92): David Griesinger describes the electronic concert hall he designed for Lexicon.
*Record Pressing (p. 96): Details on the half-speed cutting, lacquer plating, and pressing techniques used for Mobile Fidelity records.

Stereo Quarterly, Winter 1980

*Fuzzy Discs (p. 16): Keeping the stylus clean and recognizing when a new one is needed.
*Technically Speaking (p. 19): Pros and cons of meters vs. bar-graph or LED displays.
*Test Reports (p. 31): Thorens 115 turntable (superb, low-mass arm, extremely low flutter and rumble, excellent suspension isolation). Technics SL-D3 turntable (excellent except for a relatively massive and totally undamped arm). BIC 830Z automatic turntable (good performance, undamped medium-mass arm, good suspension, microprocessor provides conveniences). AKG P8ES cartridge (very neutral, clean sound, excellent tracking, medium compliance). Empire EDR. 9 cartridge (good tracking, erratic high-frequency response, sounds mediocre). Ortofon LM-30 cartridge (weighs 2.6 grams, medium compliance, good tracking, mostly superb sound especially at high levels). dbx 21 disc/tape decoder and dbx encoded discs (spectacular, incredibly clean sound at a modest cost). RG Dynamics Pro-20 dynamic expander (measures good, sounds good with rock, not good with classical). Nikko 719 receiver (fairly good, tuning-lock correctly aligned; poor filters, non-standard phono impedance). Royal Sound RC2000 preamp/equalizer and RA6000 power amp for car or van installation (expensive, excellent, clean and powerful; input impedance only 1200 ohms).
*Audiophile Discs (p. 53): Reviews of dbx-encoded, direct, and digital discs.
*Phono Equipment Directory (p. 56): Specs for turntables, cartridges, arms, and accessories.
*Perfect Match (p. 82): A basic introduction to cartridge/arm compatibility, tracking geometry, etc.
*Buzzwords (p. 88): Explaining some of the jargon of audio.

Stereo Directory and Buying Guide 1980

Stereo Review's massive directory of specs and features for a couple thousand products; as with Audio's directory issue, some of the specs have to be taken with a grain of salt because adoption of the new IHF amplifier specs has been erratic, but this volume remains a comprehensive and useful reference.

Stereo Review, October 1979

*Audio Q & A (p. 26): Tom Holman on stereo depth imaging.
*A Decade in Audio (p. 28): Ralph Hodges reviews his career with audiophiliosis.
*Tape Talk (p. 32): The idea behind Tandberg's DYNEQ system for preventing high-frequency tape saturation.
*Technical Talk (p. 41): Why imaging is a low-priority problem with Hirsch.

*Test Reports (p. 44): Mordaunt-Short Festival 2 speaker (compact, low efficiency, beamy treble, no deep bass, but they sound good considering their size and price). Pioneer SX-7800 amplifier ("non-switching" dynamic-bias output stage yields ultra-low distortion; substantial output current for low-impedance loads; shallow infrasonic filter, non-standard phono impedance; protection relay is too active; wide-range fluorescent meter is effective and educational). Audio Technica AT25 cartridge (low inductance, superb tracking, sounds fine). Rotel 2100 tuner (very good performance, useless or defective signal-strength indicator, ambiguous owner's manual). Allison Acoustics Electronic Subwoofer bass equalizer (superb audio-bandpass filter, "subwoofer" function very effective but needs a big amp and long-throw woofers for best results).
*Subwoofers (p. 71): All about the problems of putting low end into records and reproducing it at home, a buying guide to subwoofers, common-bass pro and con, and biamplification.
*What's New (p. 77): Lots of new products seen at the June CES.

<u>Stereo Review, November 1979</u>

*Audio Q & A (p. 22): A complete treatise on AC plug orientation.
*Audio Basics (p. 26): Recalling some old audio history, by Bob Greene (effective this issue, Ralph Hodges and Gary Stock have been replaced by Bob Greene and David Ranada).
*Tape Talk (p. 28): Two heads vs. three, and head bumps (with a phony-looking graph).
*Technical Talk (p. 31): How typical are the samples which magazines review -- especially the pre-production prototypes?
*Test Reports (p. 36): Akai 635D open-reel tape deck (performance superb at 7 1/2 ips; optional built-in Dolby at a bargain price). dbx 21 decoder for dbx encoded records (superb, master-tape-quality sound, no bothersome side effects even with piano recordings). GAS Thalia II preamp (only modest flexibility, excellent performance, good tone controls). Luxman L-11 integrated amplifier ("a beautiful product," very conservatively rated, lots of output current for low-impedance loads, good filters and controls, very effective linear spectrum tilt control). Acoustique 3A Andante speaker (built-in woofer amp works fine, system plays amazingly loud for *its* small size, tweeter dispersion is asymmetric, speaker sounds excellent).
*FM Broadcast Quality (p. 60): On compression and limiting.
*National Public Radio (p. 64): Enthusiastic notes on NPR's network programming and its new satellite service.

<u>Studio Sound, July 1979</u>

*Studio Diary (p. 42): Description of Columbia's enormous complex of studios, disc-mastering plants, and auxiliary facilities in New York.
*Digital Audio Discs (p. 50): Details on the hardware for several digital disc systems, with emphasis on the Philips 4 1/2-inch Compact Disc. A comprehensive and thorough survey.

<u>Studio Sound, October 1979</u>

*Ambisonics, Theory and Patents (p. 36): Matrixing, quad, and the background of the Ambisonics approach to recording a sound field.
*The Calrec Sound Field Microphone (p. 42): Details on the workings of the mike and its matrixer.
*Ambisonics Experience (p. 46): Reports on the BBC's uses of the Soundfield mike, possibly "the most important mike development since Blumlein."

-- Peter Mitchell and Jiri Burdych

September BAS Meeting

The September meeting was held at You-Do-It Electronics Center in Needham. Jim Brinton announced that membership renewals had been coming in somewhat ahead of schedule, and that because of the resulting healthy bank balance the executive committee decided to release $1,000 to the test equipment committee. This will be used to purchase an Ivie IE-10A octave band spectrum analyzer, two pink noise generators and a shipping case to hold it all. Money for additional equipment will be available later if our initial experience justifies it.

The election of officers was next on the agenda. The slate that was standing for election was identical to last year's, with one exception, Jim Brinton having announced his plan to retire from the presidency: Peter Mitchell - President, Alvin Foster - Recording Secretary, Frank Farlow - Corresponding Secretary, Henry Belot - Treasurer. There being no other nominations, the slate was elected, unanimously, by acclamation. The question of a quorum was pointedly ignored, although an informal count later revealed that there were enough people there to make it official. Jim Brinton relinquished the podium, and Peter Mitchell took over as president.

Peter then gave a brief history of the BAS, and described the years of faithful service to the Society performed by Jim. Groups of this nature usually exist because of the continuing efforts of a few people. Jim and Joyce Brinton have given many hours of their spare time, and also of time they really couldn't spare, to help make the BAS the most respected and influential group of its kind in this country. Our profound thanks go to them both. Jim has, unfortunately, been so busy that he may not be attending many BAS meetings in the near future, but we can at least make sure that he gets regular reports. Accordingly, Peter moved that Jim be made an honorary life member of the BAS. The motion was passed unanimously.

There have been several amendments to the constitution proposed recently. Since the text of these had not been published in the Speaker at the time of the meeting, the printed texts were handed out and the vote postponed to the following month.

Frank Farlow announced that the BAS wound be incorporating in the next few months. Since the corporation will be non-profit, the process is simple and costs only about $30. The purpose of the change is to protect the officers from liability as individuals for the acts of the BAS.

Scott Kent displayed several records made from his master tapes. One of these is of a baroque opera which was recorded with a single pair of crossed figure-eight microphones (Blumlein technique). The stereo imaging and the placement of the instruments and singers is good, although to keep the singers from turning toward the central microphone array Scott had to have a "soloist" mike placed in front of each soloist, wherever he or she happened to be at that moment. The singer would then face directly downstage. The third microphone was connected to a preamp but was not being used for the recording. (This record is AFKA #SK 285.) Another Blumlein recording, of piano and single woodwinds in Jordan Hall, features music by the composer/performer, Minuetta Kessler (AFKA #SK 288). There is also a recording of the Harvard-Radcliffe Orchestra in Sanders Theatre made with coincident cardioids plus a pair of spaced omnis (AFKA #S 4638). Scott also praised a tape made of the Hook organ in the Church of the Immaculate Conception, mastered by David Griesinger and cut by David Sachs at Sheffield for their Town Hall series. Although these discs are not Sheffield's most expensive line, they are often done exceptionally well.

Scott also spoke of his recent experience making organ recordings in Boston with equipment provided by Digital Recording Systems Co. Founded by former Bostonian Terry Tobias, DRS offers two-track digital recording using professional 3/4-inch U-matic tape machines and a Sony PCM-1600 sixteen-bit linear digital encoder. Scott brought his carefully-aligned 15 ips half-track Revox with professional dbx noise reduction to the session and compared the quality of the two systems. The result: the analog system sounded just as good for this application, but that is predicated on the analog recorder's being set up very carefully with the same tape used during the session. Scott spends close to an hour doing this each time he goes out, and without this degree of preparation (for which most commercial recordists rarely have the time) the analog tape would have less flat frequency response and more modulation noise. (Pipe organ is forgiving of both modulation noise and tape hiss, the former because there is a rush of air accompanying the sounding of each pipe, the latter because the organ blower tends to hiss louder than mike preamps or noise-reduced tape machines. -- EBM) The digital system, on the other hand, is always as close to flat as its analog electronics, without any preparation. According to Scott, its noise floor is slightly lower than his system, and the total dynamic range of the PCM-1600 is almost 100 dB. The noise floor on both tapes consisted of organ blower and room noise, and was well above the noise floor of the digital system even when the peak levels were more than 8 dB below the maximum allowable. In PCM encoding, the maximum level is absolute; above it, the system clips hard. Digital encoders therefore have peak-reading LED level indicators. The ones on the PCM-1600 go from -42 dBm to +20 dBm, with 2, 4, or 6 dB steps except around 0 dBm, where there are markers at plus and minus 0.4 dB. The maximum operating level of the system is higher than most mixing boards and preamps, and since there is no level control, Scott had to build a special preamp capable of +30 dBm to use the system's entire range. The PCM-1600 operated flawlessly with tapes recorded and played back on the same machine, but one of the two machines would not play tapes made on the other without dropouts. (I have been told by a reliable source that there is not a single commercially released digital recording which did not require the use of the inevitable backup tape machine at some point in its production. -- EBM)

DRS has a video editor which uses a SMPTE time code, permitting editing to the nearest 1/30 of a second. This cannot be used to splice in the middle of the music, however, because the instantaneous change in level almost always produces a click. Tobias has plans to acquire a digital editor soon. Sony makes a very nice one for only $50,000...

Scott concluded by saying that a good analog recording is better than the disc cutting and playing process, but that very few really good analog tapes get made because of the amount of effort required. Genuine digital discs, and dbx-encoded analog discs, show up the flaws in analog masters that formerly were concealed by the disc reproduction process. Peter Mitchell reported a conversation with Thomas Frost of Columbia records in which Frost stated that Columbia was not releasing any recordings whose master tapes were made outside the company unless they were done digitally. Where it will all end is far from clear, but things are definitely changing.

First Meeting Feature - Bob Carver

As open forum drew to a close, A1 Foster arrived with Bob and Diana Carver. Formerly head of Phase Linear, now of Carver Corporation, Bob displayed and described the Carver C-4000 Sonic Hologram preamplifier and the Carver Magnetic Amplifier.

The basic preamplifier section of the C-4000 has two phono preamps and three high-level inputs, two tape monitor loops with provision for copying between them, and an external processor loop.

The two phono preamplifier sections use different circuitry; one is constructed entirely with bipolar devices, the other with FETs so that the audiophile who prefers one or the other for ideological reasons has a choice. Both inputs present simple combinations of resistance and capacitance to the cartridge. The FET input offers a choice of input capacitances: 0 (!), 180, and 390 pF; the other input is 0 pF. RIAA equalization is specified as accurate within ±0.125 dB. The "old" curve (not the IEC standard with the built-in bass rolloff) is used. Signal-to-noise ratios using the new IHF method (dBA below 5 my with a dummy cartridge across the input) are 78 dB for the FET input and 80 dB for the bipolar. Carver stated that a user might want to choose one input or the other if interference problems were encountered, because the bipolar input is more immune to magnetic interference, the FET to electrical.

The high-level section of the C-4000 has bass and treble controls for each channel. The bass control has two modes, one of which is designed to be used as a loudness control, presumably similar, as is much of the C-4000's basic preamp, to the Apt. There is a stereo/mono switch, a -20 dB muting switch, a switchable infrasonic filter (18 dB/octave below about 15 Hz), and a separate headphone amp. The infrasonic filter comes before the tape outputs, and acts as a buffer for them. When the filter is out of the circuit, there is a 4.7 kOhm resistor in series with the tape outputs. The impedance of the high level inputs is 80 kOhms with the infrasonic filter, and 20 k without (this will by itself produce an audible difference with some associated equipment -- EBM). The maximum output of the C-4000 is 12 volts peak, and the slew rate, Carver says, is equivalent to 1000 V/microsecond at the output of a 200-Watt amplifier with an 8 Ohm load (if this hypothetical amp has an input sensitivity of 1 Volt, the slew rate of the preamp is 25 Volts/microsecond -- EBM).

The C-4000 contains a redesigned version of the Phase Linear autocorrelator noise reduction system, which Carver says is vastly improved over the original. It has five separate high-frequency bands, and a floating threshold which is said to prevent it from trying to track signals such as the breathiness in a vocal track. The new system gives 8 dB of noise reduction, and has a switchable automatic threshold control which will adjust the threshold to within 3 dB of the noise floor. According to Carver, it functions about as well as an automatic transmission: you can do better manually, but only with skill and practice.

There is also a new version of the peak unlimiting circuit used in the Phase Linear preamp, which Carver says is more subtle in its action than the old version. The peak detector has an attack time of 500 microseconds and a basic release time of 7 milliseconds, which is modified by a circuit which follows the envelope of the program material.

There is time delay in the C-4000, too, consisting of two delay times, 35 and 50 msec, and an echo density control which regulates the amount of recirculation of the delayed signal. There are two philosophies in current time delay designs. One, as exemplified by Sound Concepts, calls for identical delay and phase in the two rear channels to present a coherent rear image. The other, as seen in Audiopulse and ADS products, dictates that the delays to the rear channels be different to prevent localization to the rear. The Carver delay circuitry uses the latter approach.

The delayed signals are fed to two 25-Watt power amps, which are part of the preamp. There are loudspeaker connections for two rear speakers and one derived front center channel; Carver says the front channel will emit mainly short delays. The small power amplifiers are adequate for most installations, according to Carver, because so much of the spatial information they would normally be expected to convey is contained in the sonic hologram, and is therefore supplied by the front speakers.

As for the sonic hologram circuit, there will be a great deal written about its nature and performance in present and future issues of the Speaker, so this report will be confined to Carver's description of it. According to Carver:

The sonic hologram gives front-to-back depth, as well as an image which extends horizontally beyond the speakers. The image is from three to fifteen feet deep, and gives the impression that one is seated about eighth row center in the concert hall. The listener must sit within about six inches of the center line to hear the effect correctly; outside the central zone the image deteriorates into a "warm, time-delayish stereo." The system works best in very dead rooms, and requires that the speakers be placed correctly. This means pulling them away from the wall; the apparent depth of the image behind the speakers is 2.4 times the distance from the drivers to the wall behind them. Carver had little information at the time he spoke to us about which speakers would be best for creating the hologram, but guessed that models like the Bose 901 would not work at all. He claims that the first four reflections from the speakers have been programmed into the device, and that the speaker/room optimization process consists mostly of matching the system to the imaginary room in the program.

There is a "holographic injection ratio" switch on the C-4000 whose two positions are labeled "normal" and "theoretical." (As the laughter which greeted this information subsided, Carver said, "I'm doing this for fun anyway.") The "normal" position is for use with the usual multiple-mono mixdowns that comprise the vast majority of today's discs, while the "theoretical" position is meant for recordings made with "purist" techniques. The sonic hologram works by using interference effects to give the listener phase information that is already present on the record. The image it creates is not necessarily absolutely correct, but is always believable. The time delay feature complements the sonic hologram by making the room seem larger and by placing a large reverberant space around the holographic image.

The Carver C-4000 has a list price of $867, and will be sold in the Boston area by Eardrum, Tweeter, Natural Sound, and MSL. Current plans do not include the production of a separate box for those who do not want the basic preamplifier. One reason for this is that leaving out the preamp might not reduce the price all that much, since there are still the autocorrelator/noise reducer, the peak unlimiter, the time delay/reverberation circuit, two power amplifiers, and the sonic hologram.

Carver also showed the magnetic amplifier. The unit was thrown to him by an assistant in the second row, who easily tossed it eight feet or so with one hand. The chassis is a cube less than one foot on a side, weighing about what one would expect of a preamp. Its output is 275 Watts per channel at clipping into 8 Ohms. It is also FTC rated into 4 Ohms (at Watts per side) and can be bridged for monaural operation. A detailed description of the magnetic amplifier circuit will appear in the meeting summary in the February, 1980 Speaker. The amp should be in the stores by the time you read this, at a list price of $349.

Second Meeting Feature - Three Reps

After the break Jack Wilson, the proprietor of You-Do-It, introduced three sales representatives, or reps, who spoke to the Society about different aspects of electronic marketing. The first of these was Wally Schwartz, vice president of marketing for Peerless Audio, and the designer of the Peerless Audio speaker kits, who gave a brief history of the electronic parts business.

In the early days of radio, there were few service men. If there was something wrong with your set, you would remove the single "80" vacuum tube and take it down to the nearest Walgreen's Drug Store, where there was a tube tester, and check it out. If the meter stayed in the red zone instead of going into the yellow (marked "?") or green (marked "good") zone, the store would sell you a new one. As radios grew more complicated, repairmen appeared. During this era the electronic hobbyist had a difficult time getting resistors and capacitors, because the parts suppliers were strictly wholesalers, and would refer the retail customer to the repairman, who couldn't be bothered to stock a large inventory of parts and was unlikely to have what was needed. Into the gap thus created moved Lafayette, Allied, and Radio Shack. Overcoming early resistance to the idea of marketing parts directly to the consumer, the parts houses flourished, and Radio Shack, which set a goal of one thousand stores for itself in the '60s, now has over six thousand stores in the continental U.S.

The next speaker was Bob Keene, of Keene Sales, whose company represents Mitsubishi car stereo products, Rek-o-ton replacement styli, Tech-Spray cleaners and lubricants, and the Bearcat line of scanner receivers. Bob focused his talk on the latter product.

Bearcat was started by an engineer named A1 Lovell in 1963. Lovell's first product was a converter which received police radio frequencies and put out a signal at 535 kHz AM, which could be picked up on a portable radio. The device was sold to law enforcement officers so they could hear calls from the station while out of their cars; previous to the introduction of the converter, a policeman who stopped for a cup of coffee would park the squad car at the curb and leave the radio turned up loud so he could respond to his calls. Of course, the converter could receive only one frequency, and there was a need for a device which would monitor more than one, so that, for instance, officers patrolling near a town line could hear broadcasts from the neighboring town's dispatcher. Lovell's first scanner went on the market in 1964.

Early scanners had to have a separate crystal for each frequency, but the latest models can use the output of a single crystal to synthesize many frequencies. The Bearcat Model 210 can look at 2,100 different frequencies in both VHF and UHF public service bands. There are buttons on the Model 210 which will instruct the unit to scan only aircraft, or police, or radiotelephone bands. The microprocessor that controls the tuner can store many frequencies for later recall, enabling the user to choose a list of his preferred stations and scan through those. A scanner works by switching through the chosen list or range of frequencies and locking on to the first one that has a carrier on it. This scheme works best for broadcast stations in the public service bands that are in intermittent use, and would not be useful for CB, where many channels are filled with constant blather.

It is legal for any United States citizen to receive any radio transmission. This right was established by the Federal Communications Act of 1934. Bearcat believes that a citizenry that is aware of police activity is an asset, and the ratio of helpful to destructive responses among civilian listeners supports this premise. There are laws that regulate what a private citizen may do with the information he gets. So, for instance, under the FCC provision radar detectors should be legal, but it may be unlawful for a motorist, having heard his detector go off, to get on channel 19 and tell his good buddies that Smokey is taking pictures at mile 191.

The Federal Communications Act is up for review in Congress, but Keene does not think that the public's right to receive all broadcasts is in danger. (We may hear more about this question in the near future. Radio Shack is planning to market a satellite receiving station for around $1,000 that will receive up to forty channels, including the links used for internal transmission by the major networks. The few who already have such a station can, for example, pick up the Carson show live and unedited as it is beamed from Burbank to New York, where the show is recorded for editing and replay later in the same evening. -- EBM) There are some frequencies to which Bearcat scanners cannot be tuned, however: the ones used for internal communication by the FBI. This was done as a result of what Keene described as an extremely cordial visit from the Bureau to the Bearcat front office. There were expressions of disbelief in the audience at the implication that sensitive information was being transmitted in the clear, without scrambling or encoding of any kind.

In closing, Keene reported that business was good for the manufacturers he carries, with no sign of the recession we are supposed to be having.

The final speaker of the evening was Clay Anderson of Grossman Sales. Clay is well known among local audiophiles. He used to manage the hi-fi store at Minute Man in Harvard Square, and is currently the local rep for Peerless, Empire, Dual, Centrex, and, last but certainly not least, U.S. Pioneer. Clay is an audiophile who turned his obsession into his livelihood, and he had many interesting things to say about the electronics business. (He is a long-time listener to Shop Talk, although he says he now uses the program mainly to tune his receiver for Car Talk. -- EBM)

The rep is an interface between the manufacturer and the dealer. Grossman Sales is an independent rep, as opposed to a manufacturer's rep, which according to Anderson gives them more allegiance to the dealers they serve; contracts with manufacturers are written for short periods, sometimes only thirty days at a time.

Many audiophiles have dreamed of opening a dealership. It seems like a good way to earn money by playing with stereo equipment, and besides you can get your own system cheap. Unfortunately, a case of audiophilia nervosa is not enough of a resource on which to base a successful business. In fact, it doesn't even qualify you as a salesman. The best hi-fi salesmen are Fuller Brush men who like hi-fi; the worst are the audiophiles. To understand this, it helps to realize that a good salesman is someone who leads the customer into a relationship with the product, not with the salesman. This means among other things that it works far better to let the customer tell <u>you</u> about the equipment rather than you telling him. The typical audiophile is too busy trying to impress the customer with his technical knowledge to let the customer connect with the equipment. While Anderson was manager of Minute Man, many college students with some knowledge of stereo came to him for salesman's jobs. He would lead the prospect to the store's cheapest system and ask, "Could you sell this system to a customer?" If the applicant hesitated even slightly, he was lost. Anderson has been in many stereo stores, and he says that hi-fi salesmen are in far greater need of sales training than they are of technical training.

Someone in the audience told a story of having tried to open a dealership, and complained that manufacturers did not seem to want to sell to him. Anderson described what a manufacturer looks for in a dealer. The dealer must pass a credit check. Cash flow problems are widespread in the industry. He should have adequate space to display the product. The manufacturer must ask whether the dealer will bring in new customers or takeaway customers from other dealers carrying the same line. And a guess must be ventured as to whether the dealer will endure. Part of that question involves the dealer's willingness to grow, because most businesses either grow or die. For all these reasons, a new dealer has trouble getting good lines, and conversely a new manufacturer has a similar problem trying to get good dealers.

There were complaints from the audience that Pioneer is trying to sell stereo equipment everywhere, and that established hi-fi dealers don't like the line for that reason. Anderson replied by pointing out that the hi-fi business has changed from a hobbyist industry to a large-scale consumer industry, and that there are advantages for the audiophile in being at the top of a pyramid whose base is growing. The top of the hi-fi pyramid is occupied by high-end shops, and the bottom by department stores like Lechmere Sales. Nine out of ten Lechmere customers who buy a stereo system pay their next visit to the audio department ten years later, troubled by the vague suspicion that maybe they need a new needle. The tenth customer comes back in one or two months, feeling dissatisifed and wondering whether what he really needs is a better set of speakers or a bigger power amp. "He then becomes, " says Anderson ominously, "one of Us."

The aggressive marketing of the big Japanese firms has expanded the whole industry and made available money for research into new technologies such as digital audio, whose design demands a knowledge of audio, digital electronics, and video. The initial successes of the Japanese companies came in the days of fair trade, when prices were maintained at a high level. Japanese electronics firms, who could hire the best engineers because the country was not geared to a military economy or involved in a space race, developed the first solid-state hi-fi products that worked reliably. American servicemen returning from the Far East brought back Japanese hi-fi gear and helped popularize it. In those days Pioneer spent more than any other company policing fair trade. When the fair trade laws were repealed, manufacturers were forced to change their marketing strategy and broaden their base. Naturally, the older dealers didn't like it, but what they really want is a return to "fair trade." As Anderson put it, "I want the Lone Ranger back, too, but we're not gonna get him."

October BAS Meeting

Business Meeting

Peter Mitchell opened the October meeting at GTE Labs by mentioning the proposed amendments to the BAS constitution which were distributed at the last meeting. He felt that instead of voting on the amendments this evening, everyone should continue to think about them and next month a Constitutional Committee would be formed. Any suggestions about the amendments and further discussion could then be put to the committee.

A summary of the treasurer's report, appearing in full elsewhere in this issue, was presented by Henry Belot. As of September 30, the BAS had about $8,400 in the treasury. Expenses exceeded income by $1,912 this year, mainly due to inflated costs of regular expenses and a slower growth of membership than in prior years. Last year the BAS grew by 44 to a total of 1,259 dues-paying members. There are about 13 honorary members. Whether we will continue to operate on a deficit budget, Henry concluded, will depend on the Executive Committee's policy on how fast the Society should grow in the future.

Introducing himself as representing the Wisconsin delegation, Dean Slindee described his modification of the Rabco SL-8E straight-line tracking tone arm. Essentially all of the parts are replaced, with the exception of some frame pieces and two motors. This modification is intended to reduce coupling of vibrations into the tone arm, damp tone arm resonances, reduce tone arm mass, and smooth the overall operation of the arm. Dean brought along one of his arms which he displayed in the rear of the room. He will perform this modification on a customer-supplied arm for a $550 service fee. For more information, contact him through AUDIOETC, Box 55, Lansing, IA 52151.

Dick Glidewell announced that, as a dealer for Dynavector and Audio Control, he has been trying to arrange a special purchase for members of the BAS on some of their products. Dynavector has two high end moving coil cartridges, the DV/KARAT-ruby with a solid ruby cantilever and the DV/KARAT-diamond with a solid diamond cantilever. These have nearly flat, resonance-free frequency response curves up to 50 kHz and 70 kHz, respectively. They also boast reduced temperature sensitivity because rubber cantilever suspension material is not used for damping. The diamond goes for $1,000 with replacement stylus at $555. The ruby is $275; the replacement stylus is $150. Unfortunately, discounts are not available on this line. Dick's impression is that Dynavector developed the diamond cantilever as a technical exercise and as a prestigious and expensive top-of-the-line model. The one they seem most interested in selling is the ruby, which is priced competitively with the Denon 103-D. Dick has auditioned the ruby model with various step-up devices (the jewelled-cantilever models have output about equal to the Denons, unlike the other Dynavector models which have high output) and reports that it sounds excellent.

Dick also demonstrated the new Audio Control C-101 combination graphic equalizer and LED spectrum analyzer. The C-101 contains the Audio Control Model C-25 10-band equalizer, a spectrum analyzer having 8 dB/octave slopes, a pink noise generator, and a microphone, all for $550. This is just $250 over the price of the C-25 equalizer alone. The spectrum analyzer can also be had separately for $400. Dick was able to arrange a special purchase price for BAS members on these items. If you are interested, write to him at Audio Calibration, 25 Fenno Drive, Westminster, MA 01473, or call (617) 874-0706.

A1 Southwick announced he had a Pioneer CTF-900 cassette deck for sale at $290, and Peter Mitchell offered to sell a pair of ADS-910 speakers for $700.

Meeting Feature: Tom Holman Discusses Power Amplifiers

It has been a year since the Apt/Holman preamplifier was introduced. The next product from Apt Corporation, the 100 Watt per channel Apt 1 amplifier, is now appearing on dealers' shelves. As with the Apt/Holman preamp, the fundamental philosophy guiding the design of the Apt 1 amp is that it should interface benignly with the components it connects to, i.e. , that its performance should in no way be degraded by interactions with the preamp driving it or by any loudspeaker load. Apt founder Tom Holman was on hand to elaborate on a number of problems which still plague contemporary power amplifier performance and how these were designed out of the Apt 1. Also present was Andy Petite, who gave a brief description of Boston Acoustics' first product, the Boston A200 speaker. These two components were teamed for a demonstration of the outstanding features of each.

Power Amp Problems Persist

Tom introduced his discussion of power amplifier (mis)behavior by relating some of the learning experiences he has had both designing and working with power amplifiers. While involved with the design of the Model 400 table radio during his tenure at Advent, Tom concluded that if you are making a 5 Watt amplifier, it had better behave nicely when it is in overload. It also began to become obvious to him that conventional test signals and measurement procedures were not telling the whole story about amplifier performance.

On another occasion, Tom was recording a master tape of a solo piano piece for Advent, using state-of-the-art equipment (Schoeps, Studer, etc.), and monitoring the recording from the control room. It was necessary to try a number of different power amplifiers before finding one which reproduced the piano in a realistic way. The one which finally worked happened to be so powerful as to appear to be overkill in that situation. The experience led Holman to consider why some amplifiers have problems with this type of signal.

A number of afflictions plaguing modern amplifiers are the result of poor interfacing of the amplifier with the speaker load. This includes impedance mismatching, which affects the amount of power the amplifier can deliver to the speaker, and inappropriate reaction of the amplifier protective circuitry to the complex reactive load presented by the speaker.

Tom addressed the first of these problems by recalling that, in the days before solid state became almost ubiquitous, amplifiers had output transformers with various taps for connecting 4, 8, or 16 Ohm speakers. Because of their great weight, high distortion, and limited bandwidth, transformers are missed by few, but the loss of the ability to easily match impedances has meant that transformerless solid state amplifiers operate efficiently over only a limited range of load impedances. Many people tend to consider a solid state amplifier to be a good one if it delivers, say, 100 Watts into 8 Ohms, 200 Watts into 4 Ohms, and 400 Watts into 2 Ohms. Although such an amplifier has a 400 Watt power output capability, this capability goes unused unless you can arrange for your speakers to have a 2 Ohm impedance. This type of amplifier has a "stiff" or regulated power supply which puts out a constant voltage regardless of the speaker impedance or current drawn (within limits). The Crown DC-300A is an example of this type of amplifier design. In most current high-power amplifier designs, however, the power supply voltage is allowed to drop slightly with increasing output power levels. When the amp is delivering its rated output the power supply voltage has dropped to its minimum value. When operating at less than maximum power the supply voltage rises above this minimum value, so for short periods the amplifier can deliver bursts of power greater than its maximum continuous rating. This capability, called dynamic headroom, is a good thing because it allows the amplifier to play louder on transients before clipping. This type of design, while an improvement over the stiff supply, is still not optimum, because for lower impedance loads the voltage change in the supply can become excessive, causing large amounts of power to be dissipated in the supply. Also, while the output power does not exactly double each time speaker impedance is halved, the amplifier still delivers much more power into lower impedance loads. Tom explained how both of these drawbacks were overcome in the Apt 1 by allowing the power supply to be changed to adapt to different speaker impedances. This is accomplished by using a switch to change around windings on the power transformer such that, in one position, the supply has a high voltage lower current capability, while in the other position it is in a lower voltage higher current mode. Here the trade-off option between voltage and current, which occurs in the output transformer of a conventional tube amplifier is taken over by the power transformer. Rather than providing a power transformer which has the high current capability required by low impedance loads, but which is under utilized with high impedance loads, the transformer can be optimally configured for two load ranges, with resultant savings in weight and size. The circuit configuration which accomplishes this can be seen in the power supply section of the Apt 1 circuit schematic (Figure 12).

Tom mentioned that he had recently measured the headroom of 20 contemporary power amplifiers into 8 Ohm resistive loads using the IHF procedure. In this test a low level 1 kHz sine wave signal having a high level burst of 20 cycles (0.02 sec) duration every 0.5 seconds is fed to the amplifier. The amplitude of the high level burst is raised until the amplifier begins to clip and the output voltage at that level is converted to an instantaneous power level. The amount by which this short term power capability exceeds the rated continuous power output, expressed in dB, is the dynamic headroom of the amplifier. Tom feels that the small value of dynamic headroom found for these amplifiers, ranging from 0.5 dB to 1.8 dB, was unfortunate but typical. The Apt 1 has 3 dB of headroom, which is maintained for both 8 Ohm and 4 Ohm loads through the use of the power supply switch.

Safe Area Limiting

The major problem still existing in most power amplifiers, Tom asserted, was their misbehavior when playing material with demanding transients, such as piano music, into a real load, that is a loudspeaker. This type of misbehavior rarely shows up with test signals and resistive loads. It is caused by inappropriate operation of the circuitry incorporated in the amplifier to protect the output transistors from excursions of voltage and current outside of their so-called safe operating area. According to Tom, designers of early transistor power amplifiers were not fully aware of transistor limitations, and did not incorporate safe area limiting circuitry, so that output stages occasionally blew up. By the second generation they had figured out how to save the output devices, but at the expense of the sound.

The standard curve depicting the safe operating area (SOA) of a transistor is commonly plotted on a current-versus-voltage scale. It is a closed curve within whose boundaries are the safe DC operating values of current and voltage for the transistor. In such a curve, shown in Fig. 1, the upper current limit (a) is set by the melting point of the small wire connecting the chip to the terminal pin of the transistor package. The next section of the curve (b) is the power dissipation limit of the transistor, a hyperbola along which the product of current times voltage (power) is constant. This is interrupted by a third region (c) known as secondary breakdown. If one spot on the chip happens to be passing a little more current than the rest, it will tend to heat more, raising the gain in that region. More current will then flow through the spot, heating it more, increasing the gain, and causing a current runaway condition. Ultimately the device pops. The boundary (d) represents the voltage limit set by the insulating or voltage breakdown properties of the device. The output transistors may operate outside of the (b)-(c) boundaries of the SOA only for very short periods of time before destructive overheating or breakdown occurs.

Now, assume that this transistor is one-half of the output stage of a power amplifier driving a resistive load. The values of current and voltage for this transistor would lie on a sloping straight line (e) when the device was conducting and on a level line (f) when cut off. The instantaneous point at which the transistor is operating actually slides along this line with time. For a reactive load this load line opens up into an ellipse (g), or some other more complex loop, indicating that, as the operating point travels around this curve, the voltage and current are no longer in phase. It is easy to see how a reactive load line may pass outside the SOA curve, causing potential failure of the output transistors. To prevent this, amplifier designers in the early `60s built in safe area limiter circuits which, typically, attempt to turn off the drive to the output stage when unsafe conditions beyond the SOA are detected. This can cause some bizarre reactions when, at this point, the speaker tries to dump its stored energy back into the amplifier.

To illustrate the largely unappreciated, though not uncommon, demands a loudspeaker can make on an amplifier, Tom showed a number of loudspeaker load lines under various drive conditions, using the Apt 1. All were acoustic suspension speakers of varying price but similar resonance characteristics. A typical set of load lines for one speaker at various frequencies are shown in Fig. 2. These pictures are real-time plots of I_c versus V_{ce} for an output transistor showing the load line, analogous to (e)-(f) and (g) in Fig. 1. In Fig. 2a, at 160 Hz, the speaker is presenting a very reactive load to the amplifier. At one point on the curve the transistor is putting out 40 Volts at 8 Amps or 320 Watts of peak power. In his amplifier survey, Tom found some units which could not handle this speaker.

Going down to 60 Hz, in Fig. 2b, the speaker becomes a virtually perfect resistive load. At 25 Hz, near resonance (Fig. 2c), it begins to behave strangely, its impedance evidently changing character as a function of the cone excursion. Another speaker was driven at a fixed frequency (34 Hz) near resonance with increasing amounts of power with the results shown in Fig. 3. The changing slope of the load line indicates the speaker impedance is varying with drive level.

An old Advent speaker, driven at 2 kHz (Fig. 4), exhibited anomalous behavior by continuing to draw current from the output transistor when it was supposed to be cut off. The fact that the flat part of the trace near the bottom of the picture is not down at zero current means the transistor is still passing current up to its high voltage limit, where it has very little current capability. This could be an indication of why some amplifiers sustained damage while trying to drive this tweeter.

In a speaker designed with an autotransformer input for impedance matching purposes, the low-drive-level load line at 25 Hz appears fairly normal (Fig. 5a). At higher levels it develops into a non-classical curve consisting of a set of connected straight line regions (Fig. 5b). Tom speculated that the transformer was saturating at these high drive levels (22 Amps peak) and that the impedance the amplifier was seeing was just the DC resistance of the transformer.

Some amplifiers have a particularly bad reputation in some circles. Tom related that, at Advent, one older receiver was known as a tweeter eater. Observing the output of this amplifier driving an Advent speaker at 55 Hz (Fig. 6), revealed the justification for this epitaph. At some point the protection circuit decides the output devices have had enough and shuts off their drive. The woofer inductance reacts to this and succeeds in turning things back on, generating a spike. The protection circuit tries again, resulting in another spike. These transients throw a huge amount of energy at the tweeter, which may decide that it has had enough and expire.

Even more interesting behavior was elicited from a new well specified and highly reviewed power amplifier from overseas. Driving a somewhat more difficult load than the Advent, at a low frequency, the amplifier decided to shut off at the positive peak of this signal and went all the way to the negative supply and back up (Fig. 7). Observing the same misbehavior on the load line display (Fig. 8) revealed the true character of the transient. It was taking the output devices well outside of the SOA into their second breakdown region. The safe area limiter of this amplifier was actually making the amplifier less safe and more prone to destruction under these conditions.

Tom suggested two solutions to these problems. First, start out with a substantial amount of safe area. This was the virtue of the amplifier he finally found which could realistically reproduce piano music. Second, detect those conditions of voltage, current, and time which would exceed this safe area and then disconnect the output with a relay. This eliminates the interaction problems between the load and the protection circuit.

Clean Clipping

According to Tom, there is one condition with which a power amplifier must cope which normally does not occur in any other part of the audio chain: sometime during its life you can be sure it's going to be clipped. It therefore behooves the designer to insure that clipping does not incite any other kind of bad behavior in the amplifier. During loud passages minor clipping may go unnoticed, while the adverse reaction of the amplifier to the clipping condition can be glaringly obvious.

One potential cause of amplifier misbehavior during clipping is the use of capacitors within the gain loop. The usual reason given for excluding capacitors from the signal path of an amplifier (to achieve DC coupling) is that coupling capacitors can add phase shifts to the signal. Psychoacoustic studies and tests with Apt/Holman preamps have shown that phase shift is not a

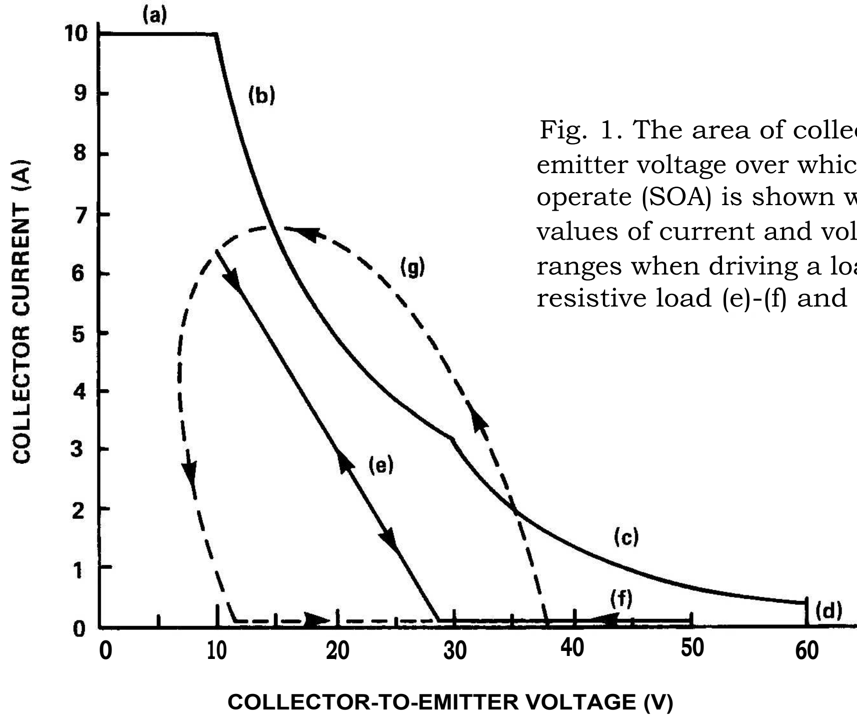


Fig. 1. The area of collector current and collector-to-emitter voltage over which an output transistor can safely operate (SOA) is shown within the boundary (a)-(d). The values of current and voltage over which this transistor ranges when driving a load (load line) are shown for a resistive load (e)-(f) and a reactive load (g).

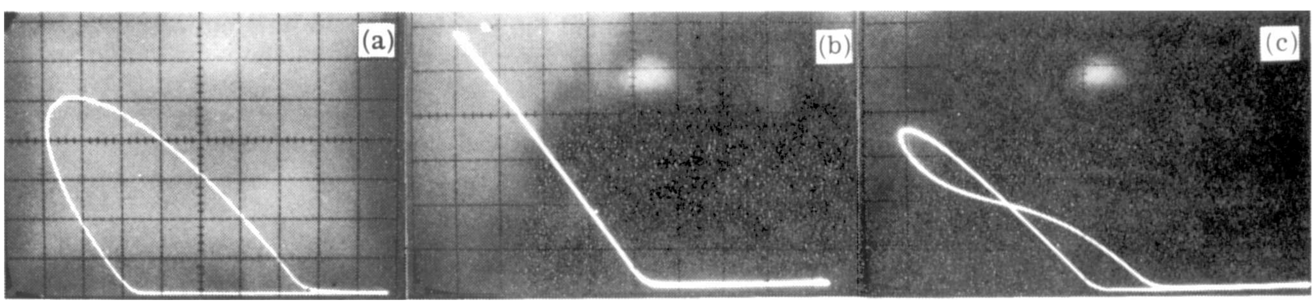


Fig. 2. The load line for an acoustic suspension speaker when driven at (a) 160 Hz, (b) 60 Hz, and (c) 25 Hz. The vertical scale is 2 Amps/division and the horizontal scale is 10 V/division.

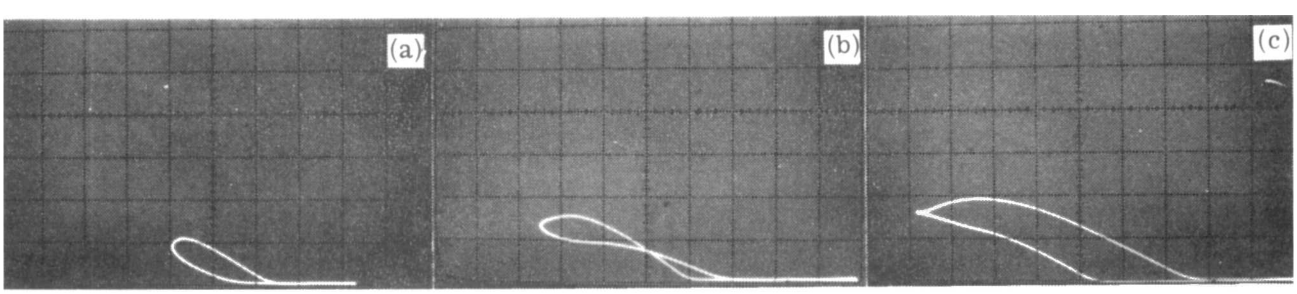
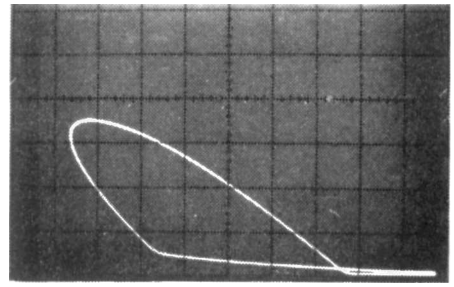


Fig. 3. The load line for a speaker driven near resonance (34 Hz) at power levels increasing from (a) to (c). Scale calibration as in Fig. 2.

Fig. 4. The load line for an old Advent speaker driven at 2 kHz. Scale calibration as in Fig. 2.



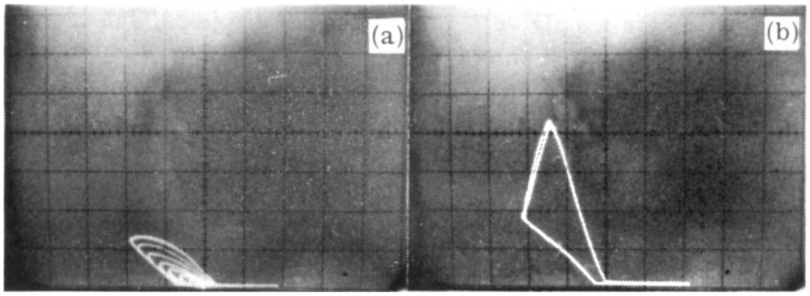
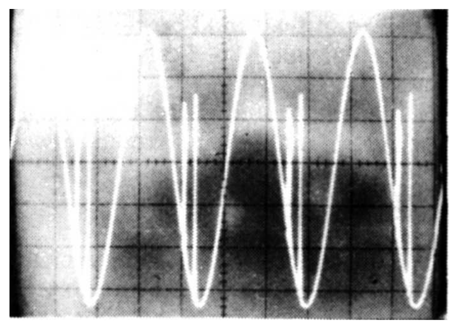


Fig. 5. The load line for a speaker with an autotransformer input driven at 25 Hz at (a) a low power level and (b) a high power level. The vertical scale is 5 Amps/division and the horizontal scale is 10 V/division.

Fig. 6. Misbehavior of an amplifier's protection circuit while driving an Advent speaker at 55 Hz is evident as transient spikes on the output voltage.



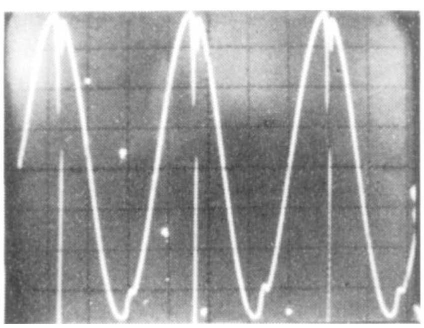
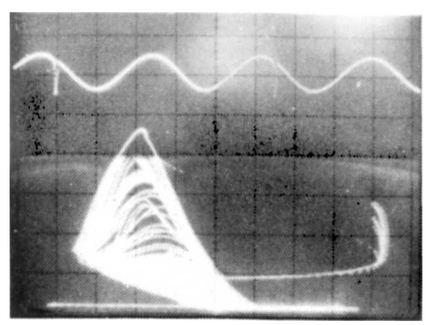


Fig. 7. The output voltage of this amplifier while driving a speaker jumps from the positive to the negative supply voltage when its protection circuit trips.

Fig. 8. The load line of the same amplifier-speaker combination as in Fig. 7 shows that the spike generated by the "protection circuit" actually takes the transistor well out of the SOA into its second breakdown region.



problem, being many orders of magnitude below what can be heard. The real problem occurs when capacitors are located where they can elicit misbehavior when they see clipped signals. One such location is the bootstrap capacitor. Bootstrapping is a circuit technique used in amplifiers to provide the output driver stage with a higher voltage drive capability without having to resort to a second high-voltage supply. It is accomplished by connecting the amplifier output back to the collector load of the driver stage through a large capacitance. When the output clips, this stage can become mis-biased, resulting in what Tom described as sounding like a motorboating-type instability.

Even without capacitors, disturbing things can happen in some amplifiers when driven into clipping. In these the output transistors can become saturated, a condition in which the base drive voltage approaches or exceeds the collector supply voltage. When this happens, the transistor stores charge in its base region and is very slow to recover after the overload condition has passed. This can upset the protection circuitry and cause the amplifier to do very strange things as the sluggish output stage tries to catch up with the drive signal.

Tom showed two examples from a number of amplifiers he had recently tested and found to clip poorly. The first, a quasi-complimentary design employing multiple feedback loops, clips asymmetrically. The negative peak clips first (Fig. 9) and remains flat until well after the drive has told it to start back up. Eventually the output comes out of saturation but it is already a little behind so it moves very fast to catch up, and generates a step-like glitch in the signal. The second amplifier clips more symmetrically but is also slow coming off the rail (Fig. 10). These clipping-induced transients are ideal for waking up tweeters.

Other amplifiers exhibited more gross behavior on clipping. A Pioneer Spec 4, clipped on positive peaks, makes a fast transition to zero output and waits there until the drive signal begins to swing negative. The protection circuit can probably take the blame for this one. A Marantz 2270, clipped at 10 kHz, manages to put the 120 Hz ripple from the power line into the load at more than the rated power of the amplifier. (This condition is exceedingly unlikely to occur during actual use. -- Ed.)

Tom concluded the discussion of clipping with a slide showing the perfect clipping behavior of the Apt 1, achieved through DC coupling techniques and elimination of output stage saturation.

TIM, SID, et al.

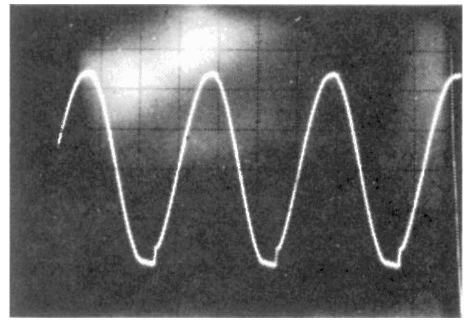
In Tom's opinion it is virtually impossible to demonstrate a need for the extremely high slew rates now being touted by many amplifier manufacturers in their promotional literature, other than as an attention grabbing spec on which to base a new marketing ploy. In May of 1976 he published the results of an extensive series of measurements on a wide range of phonograph records. It concluded that the maximum slew rate which could be expected at the output of a phono preamp was about 0.03 Volts per microsecond. Multiplying this by the high level gain and the power amplifier gain you get an amplifier slew rate requirement of 3 Volts per microsecond. Even applying a safety factor of 5 the slew requirement is only 15 Volts per microsecond. The Apt 1 slew rate, at 60 Volts per microsecond, is well beyond the demands of any audible musical signal.

Tom reminded everyone that the Apt/Holman preamp has a (defeatable) ultrasonic filter with a cutoff at 40 kHz. With this filter engaged, the preamplifier would require a maximum slewing ability of 10 Volts per microsecond from a 100 Watt amplifier following it. Unlike slew rate limiting, the filter attenuates ultrasonic garbage in the signal in a linear way. By making the amplifier substantially faster than the requirement of the filter, generation of a slewing induced distortion (SID) and transient intermodulation distortion (TIM) is completely precluded.

Around the Circuit

Tom used the schematic (Figs 11 and 12) to illustrate how the various performance features of the Apt 1 were implemented. The input stage consists of a differential Darlington pair (Q_1-Q_4) fed by a current source (Q_7, D_1, D_2). This steady current is traded between the two sides of the Darlington pair as the input signal varies; its value is such that the input stage is never allowed to saturate. Because of the Darlington configuration, very little current is required to drive the input stage. An FET input stage was considered but no American parts were available which met the current and voltage requirements of this design.

Fig. 9. This quasi-complimentary power amplifier clips asymmetrically into a resistive load, as seen on the output voltage. It is slow to recover from clipping, finally making a fast transition to catch up with the signal.



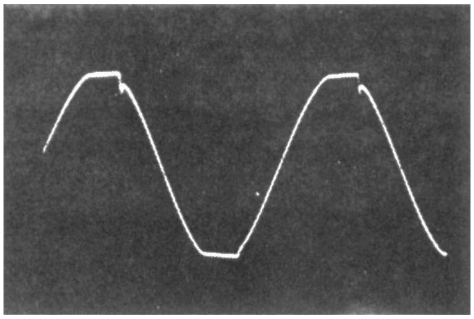
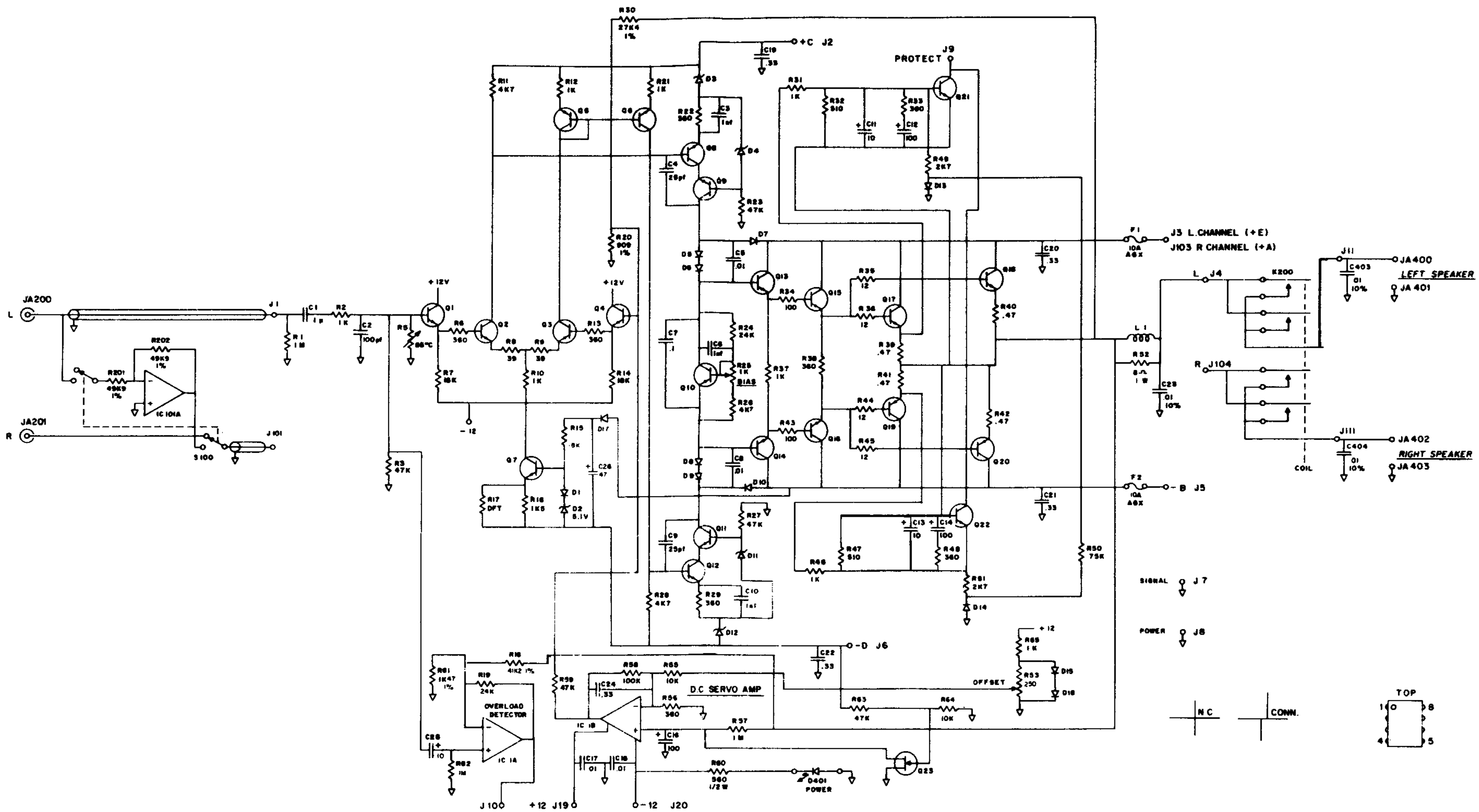


Fig. 10. When clipped into a resistive load, the output of this amplifier jumps as the output transistor came out of saturation.



Apt Corporation Cambridge, Massachusetts USA

Fig. 11. Schematic for left channel gain stages of Apt 1 power amplifier.

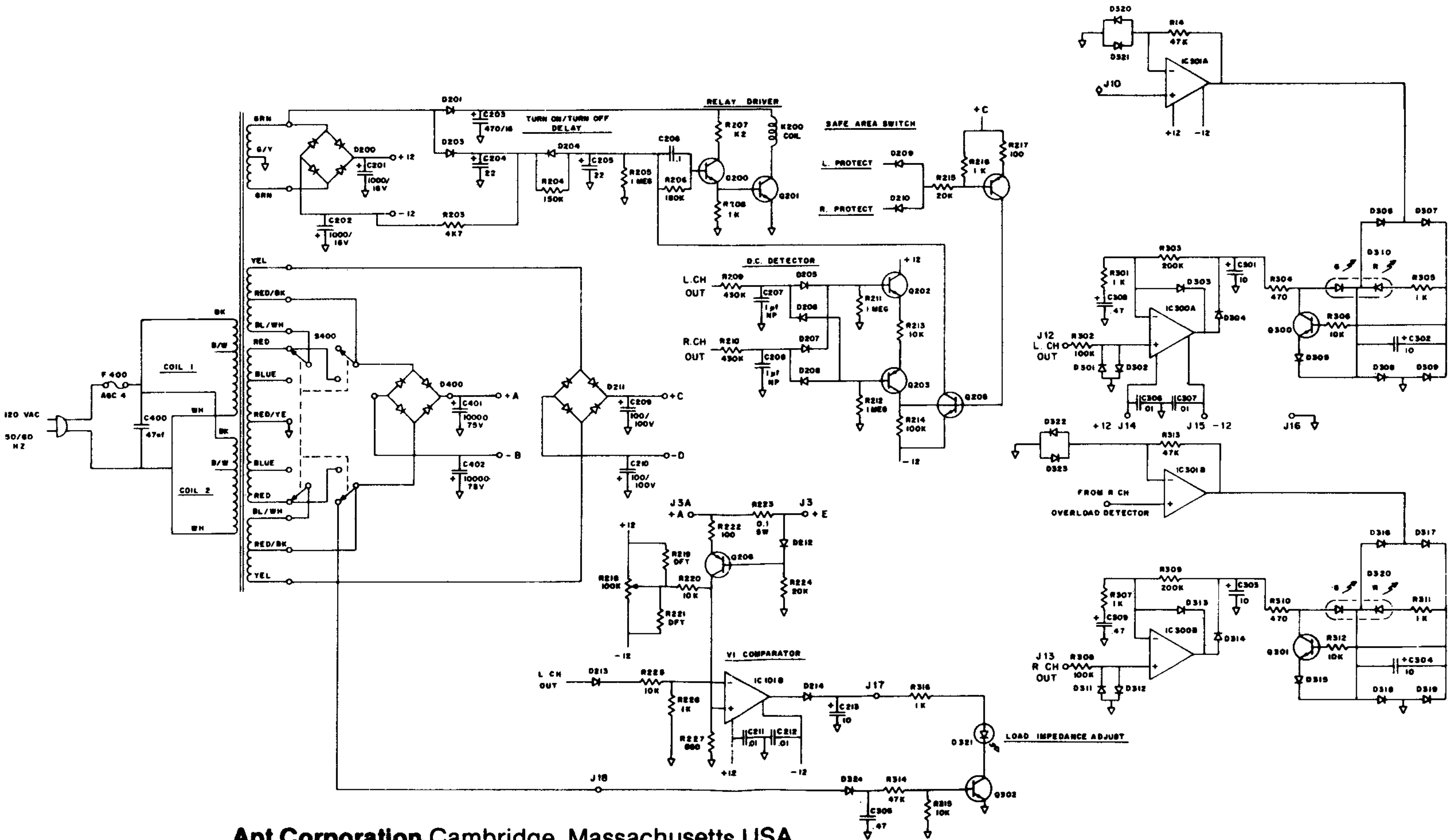


Fig. 12. Power supply schematic with some peripheral circuitry.

The output of the Darlington stage is a pair of collector currents which are out of phase with each other. Typically, only one of these currents would be used to drive the next stage. But in the Apt 1, a concerted effort was made to maintain symmetry in the circuit topology for uniform handling of positive and negative signals. Thus, one of these currents drives the positive half of the next stage while the other is turned around by a current mirror (Q_5, Q_6) and drives the negative half. This high-gain intermediate stage (Q_8-Q_{12}) was described by Tom as the heart of the amplifier. It is especially critical because the collectors swing over the entire output voltage range of the amplifier. This causes a large variation in the base-collector capacitance which can lead to distortion of the signal. To overcome this and maintain good open-loop linearity, a cascode arrangement has been used in each half of the stage.

Also in this stage are the clamp diodes (D_5-D_{10}) which are reponsible for the amp's good clipping performance. Collectively called a Baker clamp, they prevent the output stage from ever being saturated by maintaining the output stage collectors several diode drops above the base. The conventional Vbe multiplier bias transistor (Q_{10}) is mounted on the heat sink for good thermal tracking.

The output stage (Q_{13}-Q_{20}) is a triple Darlington which provides a large current gain. This reduces the current required in the previous stage and minimizes the reactions of the amplifier to strange load impedances. According to Tom, the NPN output transistor has been around for some time but the PNP is new, from Motorola, and Apt will be the first manufacturer to use it in a power amplifier. These output devices have a substantial safe operating area and require less restrictive protection circuitry.

<u>Brains for the Brawn</u>

Tom described a number of peripheral circuits surrounding the basic power amplifier which monitor its behavior and apply corrective action to alleviate any conditions which might endanger the amplifier or its load. To maintain near-zero DC at the output, a DC servo amplifier has been incorporated which checks the output and feeds back correction signals to the input stage. Heavily filtered, this circuit responds only to DC or very low frequency AC, while its gain allows the output offset to be maintained in the 10 millivolt range. Transformer-coupled speakers, whose DC resistance may be as low as 0.2 Ohms, can be a difficult load for amplifiers with larger DC offsets.

An overload detector compares the input with a divided-down version of the output to sense any conditions in which the amplifier output is not following the input (e. g. , clipping). When this occurs the overload detector produces a signal which blinks a red LED overload indicator on the front panel. The two-millisecond attack time of this circuit reduces nuisance flashing on minor record ticks. This indicator normally glows a steady green whenever there is a signal at the output jacks of about 100 microwatts or more.

For protection the Apt 1 employs a relay to disconnect the load when unsafe conditions are detected. The relay opens during turn-on and turn-off, and if DC is present at the output, or the output is short-circuited, an internal channel fuse blows, or if the output transistors spend too much time outside of their DC safe operating area. If the two protection transistors Q_{21}, Q_{22}) sense an unsafe combination of voltage, current, and time, the load is dropped in about a millisecond (the operating time of the relay). When the amplifier has recovered, the load is reconnected automatically.

Rather than the conventional "E-I" type laminated transformer core, the Apt 1 uses a "U-I" lamination which has a much lower stray magnetic field. Hum levels around the chassis are about as low as in the Apt preamp. The secondary is wound to give a ±12 V output for the low level functions, and a pair of nested high voltage supplies for the drive stages and output stage. Because windings in these two high voltage supplies are shared, the transformer can be made smaller and the switching function for accommodating different load impedances is easily implemented.

The Apt 1 contains enough heat sinking so that a fan is not required for normal signal levels. If the heat sink temperature should rise due to restricted air flow or high ambient temperature, a thermal resistor at the input lowers the input signal level so that the amplifier continues to operate, but with a reduced output.

By switching in the phase inverter between right and left channels the Apt 1 can be used in the bridging mode as a mono amplifier rated at 200 W with 2.5 dB of headroom. Tom commented that many audiophiles feel that amplifiers don't sound as good in the bridging mode. He feels that this may often be true because in this mode the load is shared by the two channels and each sees only half of the load impedance. With a low impedance load, this could cause the protection circuitry to trigger more readily.

Apt 1 plus Boston A200

After he finished discussing the Apt 1 amplifier, Tom set up a demonstration using master tapes with a Revox, Apt/Holman preamp, the Apt 1, and the new Boston Acoustics Boston A200 speakers. As might be expected, this system sounded extremely good. Equally impressive, however, was its ability to fill the large GTE auditorium with high acoustic levels without apparent strain.

Andy Petite, a founder of Boston Acoustics and designer of the A200, described some of its salient features. The driver complement consists of a 10-inch woofer, a 4-inch midrange, and a 1-inch dome tweeter, mounted in a shallow (6 3/8 inches deep) floor-standing cabinet. The woofer is a proprietary driver, designed and manufactured by Boston Acoustics. Both the midrange and tweeter are purchased from outside the company, but are modified for improved performance. The midrange cone is treated to smoothen its response and the driver is mounted in its own damped enclosure within the main cabinet. Ferrofluid is added to the tweeter's magnetic gap to obtain better control in the low end of its range and greater power handling capability through more efficient heat dissipation.

These drivers were integrated into an overall design which exhibits wide dispersion and balanced frequency response over a 10-octave range. Particular attention was paid to controlling reflections from cabinet edges and adjacent room boundaries. The woofer is mounted near the floor and, in the thin cabinet, behaves as if it were flush-mounted at the intersection between the floor and the wall. With this arrangement all reflections remain in phase with the woofer throughout its frequency range, minimizing cancellation effects. The midrange and tweeter are flush-mounted on a large front panel such that the nearest cabinet edge is more than a half-wavelength away in their frequency range. This arrangement simulates an infinite baffle, reducing the edge reflections which contribute to speaker coloration and imaging defects.

The Boston A200 is available for $700 a pair. You can add the Apt 1 for another $641.

A Publication of the BAS

Mechanical and Acoustical Feedback in Phonograph Systems

R. A. Greiner

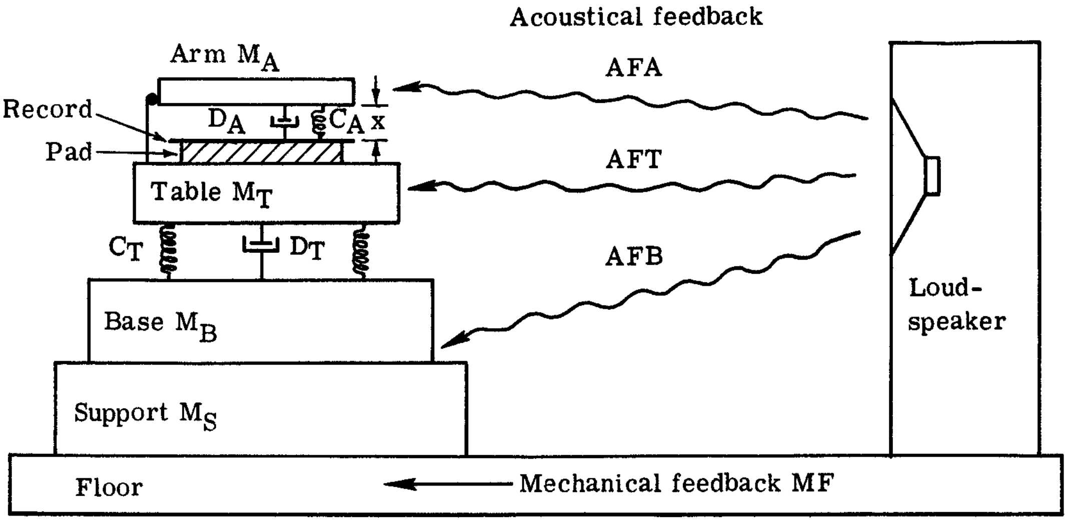
Questions asked of me recently by several students, plus several real life experiences with acoustical feedback, led me to make some measurements on the mechanical/acoustical feedback problem when reproducing phonograph records. Some of the problems came to my attention because a severe feedback problem developed when I got overenthusiastic about measuring residual disc rumble and introduced so much extra gain into the system that the whole room "took off" with a hideous clatter. (My system is capable of 5600 Watts peak and has eight 15-inch woofers, so it got my immediate attention.) Even though I have paid unusually good attention to isolation of the turntables in my system from mechanical feedback, it became apparent that if enough gain is used, it is possible to make the system go unstable.

Anyone familiar with feedback and instability theory will recognize that as a system gets near instability it reacts with great sensitivity to small changes in the system itself. It gets, for lack of a better term, sort of "quasi-stable. " A really important question, which this discussion should help answer, is: can the system increase in sensitivity, at normal gain levels, to such an extent that it will show irregularities of response or deviations from normal small signal behavior ? The answer seems to be that it certainly can and that these deviations occur well below the point of actual instability. In this note, I will describe some real system behavior, then augment the data with some techniques that can be used to evaluate and control most systems.

It is possible to determine if a system is "on the edge" of being in trouble or not. Such a system will show erratic performance and may have colorations that are not caused by the misbehavior of only one part of the system, but which may be caused by the combined behavior of several parts of the system acting together within the feedback loop that includes the loudspeakers and the room. This colored behavior may depend strongly on the setting of the overall system gain (i.e., on the setting of the volume control).

Even so common an effect as the system sounding different when people move around in the listening area may be only partly because of acoustical changes in the room, the other factor being changes in the sensitivity of the system according to changes in its loop gain. With these general considerations in hand, we'll consider some of the data, its interpretation, and the general physical behavior of the system.

Figure 1 is a simplified diagram of a turntable suspended on springs, a base for the table resting on a floor-standing cabinet, and a typical tone arm all with springs, dampers, and masses. There are four feedback paths from the loudspeakers to the table/arm system. Mechanical feedback (MF) travels through the floor and walls of the room to the base of the table (including the cabinet or shelf that it stands on). This may be a major feedback path and is the primary one for disturbances such as walking or foot stomping. Such disturbances are not fed back in a self-sustaining way, however, and can usually be eliminated entirely by brute force methods, such as massive cabinets, massive isolation bases, spring/damper feet of several sorts, and like schemes. Walking and dancing apply immense forces to the floor compared to those from a loudspeaker, and they are at very low frequencies compared to audio frequencies. Thus, it is certain that any system that has been made relatively immune to foot stomping will also be quite immune to mechanical feedback from the loudspeaker. In a well built house, it is easy to solve this problem; in a more flimsy house, it might require exotic bases, masses, and suspension feet.



M = mass C = compliance D = damping

Figure 1

The second form of feedback to the base is direct acoustical feedback (AFB). In general, if the base is massive and solid enough to shunt out the mechanical feedback, it will also shunt out the acoustical feedback. An exception might be a large area shelf that is not loaded with heavy objects. All such shelves should be loaded with books, records, amplifiers, and other nice heavy stuff to make the supporting structure for the base of the table massive. I cannot recommend wall-mounted shelves in any case: walls are very flimsy compared to floors. Turntable supporting cabinets should be floor standing and loaded with heavy stuff.

We will assume for this discussion that the mechanical and first acoustical feedback problems have been solved. This will certainly be the case for any reasonably good system.

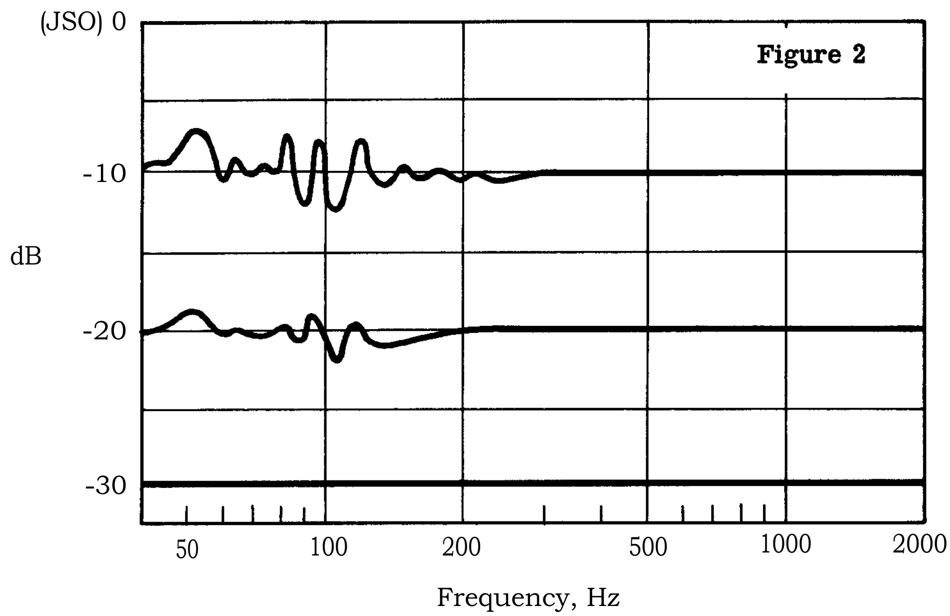
The final two feedback paths, acoustical feedback to the table structure itself (AFT) and acoustical feedback to the tone arm (AFA), are the critical performance limiting parameters in a high-quality system. They are very hard to evaluate and control and are also, unfortunately, quite significant. They set, along with the table and arm design, the upper loudness level that may be used in the system without introducing colorations into the reproduced sound. I have found that colorations may appear at levels as much as 30 dB below instability and that most systems are very poor in this respect.

Clearly, if acoustical signals in the room cause the arm to move (vibrate) with respect to the table (or vice versa), there will be an undesired electrical output from the phonograph cartridge that will feed through the system, come out of the loudspeakers, and possibly reach back to the table/arm structure. Thus, a feedback loop is closed, and the system will become sensitive and possibly even unstable. A system's normal frequency response can become irregular and its transient response underdamped at surprisingly modest loudness levels. These response anomalies are neither characteristic of the basic system nor desirable.

The details of the acoustic feedback signal paths are so complex that it is hopeless to try to write a transfer function and calculate the expected behavior of a typical system. Such an analysis would have to include the room acoustics, resonances (both mechanical and acoustical), and the like. It is clear, however, that the intensity of the sound at the table/arm structure is an important factor in the feedback analysis; considerable attention must be given to the attenuation characteristics of this path. The most straightforward, and in some ways the best, solution to getting high attenuation in the acoustic path is to place the table/arm system in another, acoustically isolated, room. If this is not convenient, an isolation booth, an acoustically treated box, or highly absorbing surfaces around the table should be tried.

To determine the actual level below instability at which the system becomes sensitive, I made some measurements with several typical turntable/arm combinations. These were done in a 3000 cubic foot, relatively dead, room with the table 20 feet from the loudspeakers. Mechanical feedback and acoustical feedback to the base were eliminated with massive support structures. The floor is poured concrete.

First, a reference level is determined at which the system "just sustains oscillation" (this is the JSO level). Sound pressure level is measured and the gain setting at the JSO level is determined as the 0 reference gain level. Then, using a sweep frequency test record, response curves for the system are run at several different levels below the JSO level. Figure 2 shows typical results for levels 10, 20, and 30 dB below JSO. At 30 dB below JSO the system is quite flat over the range measured (there are no measurable effects above 1000 Hz, so that data is not shown). At 20 dB below JSO significant feedback-resonance-reinforced irregularities are apparent, and of course, these are even worse at 10 dB below JSO. Irregularities are 4 dB peak to peak and 6 dB peak to peak respectively. Note that the irregularities are at low frequencies, but well within the important upper-bass audio band. In addition, the transient response for frequencies near these irregularities is very, very poor. Ringing of several seconds occurs at 10 dB below JSO. On the other hand, at 30 or more dB below JSO the system responds to a thump on the table with a nice, dull thud, as it should. Interestingly, the system will oscillate in the 80 to 100 Hz range at the JSO level.



I made measurements for several arms and tables, and though I obtained somewhat different results, all combinations could be made to oscillate at JSO levels within 10 dB of each other. Thus, regardless of the table/arm combination, all systems seem to show some possible troubles caused by acoustic feedback from the loudspeakers to the table/arm structure. For no irregularities to show in the frequency response curves, the system gain had to be 25 to 30 dB below JSO level. Thus, it is interesting to establish the JSO level for your system and then check to be sure you are operating enough below this level to maintain the ideal frequency and transient response characteristics anticipated in the system's original design.

This can be done as follows:

1. Place the stylus on a stationary disc.

2. Increase the gain to the point where gentle tapping on the table results in "just sustained oscillation" of the system. (Warning: increase gain very slowly when doing this.) This is the JSO level for the gain setting.

3. Record this setting as the JSO level for your system. You may not be able to reach this setting with some small, lower-power systems. However, if the tapping becomes loose and reverberant, you are probably within 10 dB of JSO.

4. Reduce the gain to the normal setting for a typical record with your normal listening preferences. Call this the normal gain of the system.

The difference between the JSO level and the normal gain level tells you the margin of safety you have. I like to call it the gain margin of the system for normal listening. This number should be as large as possible. The measurements I have made seem to indicate that it should be at least 25 dB. It is really quite hard to get a gain margin this large in a system with really good low-frequency response. The acoustic feedback problem is significant for most systems I have had a chance to evaluate. I would be inclined to question the validity of any listening tests made on a system with less gain margin than recommended here, because I think such tests would be colored by the system's marginal stability. The best way to insure that the gain margin is large is to put the turntable in another room with some reasonable acoustic isolation from the listening area.

It is also interesting to get some idea of the relative motion of the arm with respect to the table for a typical system in a given sound field. A very rough estimate can be made as follows. For one setup, the level of oscillation was 100 dB SPL at the table at 100 Hz. This was corrected to compare to a test signal, and it was determined that the arm-to-table shake was 13 microns (0.013 mm). Is it any wonder that the table/arm system is sensitive to acoustic excitation?

Because different arm/table combinations gave results that differed by about 10 dB, I made some experiments to try to find the physical design factors that most influenced the sensitivity of the arm/table structures to acoustical vibration. It was very hard to do this, and I obtained only the roughest results.

In one experiment, I measured a 20-pound table with and without an added 40 pounds of steel bars. The frequencies of the instabilities were changed a bit, but the added mass did not measurably improve the system's gain margin (this is shown in the curves of Figure 3). Mass alone is not a total answer. Adding damping to the system with some extra blocks of foam-like material seemed to help a bit, but not more than about 4 to 5 dB. The Audio- Technica isolator blocks seemed to be about 5 to 6 dB better than just plain springs with no damping. I also discovered that a small massive table is a little better than a large-surface table of similar mass. This is because the surface gathers in the acoustic vibrations. The geometry of most tables is poor in this respect, because they are more like a sheet of material than like a ball. A heavy table with good damped springs seemed to be the best.

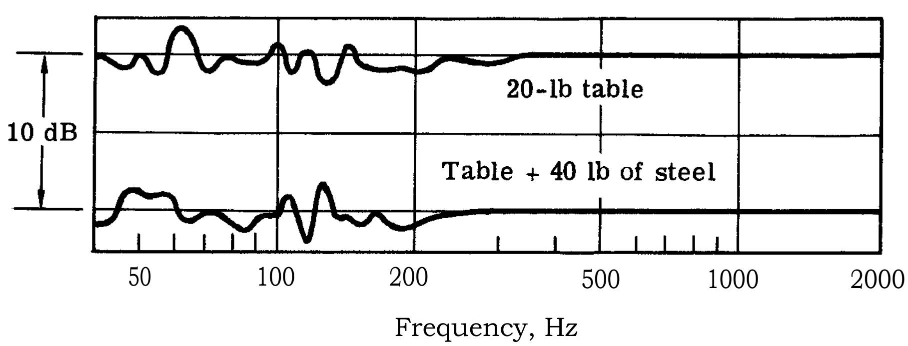


Figure 3

Even more important than the table size and weight seemed to be the design of the tone arm. A medium-weight arm with a lot of surface area, such as the old Rabco, was the poorest. An arm such as the SME was about 6 dB better. It seems that a tubular structure of the maximum weight possible, consistent with the correct effective mass at the stylus, is best. Ultra-light arms showed more acoustically induced vibration than heavier arms. Damping at the pivot and particularly at the headshell gave as much as 10 dB more gain margin. All in all these not very complete tests showed a total difference of about 10 dB between the several combinations of tables and arms tried. This is a significant range, but it also shows that one might be better off trying other tactics than changing the table/arm system to get better gain margin. The main one would be to attenuate the acoustic path in the loudspeaker to table link as much as possible.

I did check one other factor -- the effect of the turntable mat. The case of a record suspended freely on posts was by far the worst. The record is poorly supported and can vibrate freely with the acoustic excitation. Such schemes should never be used. I tried a number of mats, but obtained no conclusive results. By a very small margin, an AR mat placed on a flat table with the regular mat removed and with a 2.5 pound record puck in place was the best. It appears that a mat should touch the record in as much surface area as possible to hold the record and damp it. A record puck helps, and foam mats are the best from my data.

A lot of small factors add up to improve the isolation of the record/table/arm system from acoustic feedback. In addition, the acoustics of the room are a major factor in reducing feedback. Most acoustic coupling takes place when the loudspeaker and the table are each at anti-nodes of response in the room. If possible, the table should be placed at a dead spot in the room for low frequencies, that is, at a collection of low frequency nodes. Good diffusion of the sound in the room at low frequencies will also reduce the magnitude of the anti-nodes and make it easier to place the table. It is worth some trial and error to find the best location for the table, because this will make the system sound better and definitely improve transient response at low frequencies. From many systems I have seen, it seems likely that the opinions some persons hold about the systems are colored by important, but often ignored matters, such as described above. I am less and less inclined to trust the opinions of reviewers, many of whom not only do not control the listening situation adequately, but are also ignorant of these effects.

Making innocent changes in a system, such as placing a heavy amplifier on the same table as the turntable, can change the response of the system in a measurable and audible way. The only good solution for mechanical playback is superior acoustic isolation of the playback mechanism from the listening environment. In lieu of that, I recommend establishing at least 30 dB gain margin in the system. It seems unfortunate that mechanical playback is still with us after 100 years of the phonograph. Perhaps a really superior, low-cost method will appear soon.

A Publication of the BAS

Resolving the A-B Switchbox Controversy

James A. Mitchell

The recent articles in The BAS Speaker, The Absolute Sound and other audiophile publications concerning the David Spiegel "double blind" switchbox and discernible differences in amplifiers and preamplifiers have brought a number of audio enthusiasts into sharp debate. The questions raised are pertinent and audio development will benefit when the points at issue are clearly resolved.

As a result of my own experiments with an A-B switchbox in comparing preamps, I have developed a hypothesis which may explain the different interpretations of the tests. It has clarified a number of questions in my mind and I hope it will be useful to others who are experimenting in this field.

First, here is how the hypothesis came to be. For many years I have been critically evaluating and modifying audio equipment. I well remember my experience of replacing my Marantz 7 tube preamplifier with a Marantz 7T, a new (as of then) transistor preamp. The 7T looked great, had less hum and noise, and measured okay, but as time went on, my system sounded less and less satisfying to me. I switched pickups, adjusted speakers, and checked amplifiers until it finally dawned on me that the 7T had been a step backwards. When I borrowed back a tube Marantz 7, the listening haze and fatigue were gone. Since that time I have never changed a piece of equipment without a lengthy listening comparison.

I have changed preamps several times since then, but the old 7T is still on my spare parts shelf. I often use it as an illustration of listening fatigue, but whether that is a weakness of early transistor design or a flaw in my particular unit I do not know.

I now have two preamps which I like very much. They are the DB Systems DB-1, a recent transistor design with lots of feedback and very low distortion, and the Audio Research SP-3A, a tube design much praised by audiophiles for midrange clarity. As a result of my extensive listening comparisons of these two units, I have noted somewhat different sound characteristics and have set down in writing certain recorded passages where one preamp sounds better than the other and in what way.

About a year ago I set out to determine the source of these differences in sound and to make modifications to the units to realize the best sound of each. Fully appreciating that differences in loudness have a significant effect on sonic preference, I always precisely set the volume level using a pink noise record and an averaging voltmeter and oscilloscope at the output to the speakers. I also came to realize that the frequency response of the units being compared should be the same to eliminate that cause of difference. The DB Systems was found to be almost exactly on standard and some modifications to the SP-3A, mainly in the phono RIAA feedback circuit, brought that unit into agreement. The listening comparisons were made after these procedures were completed.

As this point, I decided to build an A-B switchbox using relays and a remote control switch. I planned to make certain modifications to the preamps and expected the switchbox to improve my power of discrimination, and to sharpen my ability to identify differences in sound.

Of course, you know the outcome. After the switchbox was checked out, and volume levels and frequency response uniformity confirmed, I was amazed to find that I could not consistently identify the two preamps. To state this more precisely, I could not distinguish significant differences when switching from A to B or from B to A. When listening for some period of time on position A to my reference recordings I could often hear the kind of sound associated with one preamp, but when I would switch from A to B to confirm the difference I could not hear it, and would become confused as to what I had heard. This is the same result that Mr. Spiegel has found and it raises some intriguing questions.

Does it mean that there are no audible differences in these preamps beyond volume and frequency response? Had my previous listening conclusions been a figment of my imagination from reading The Audio Critic, The Speaker and the like?

To examine these questions, I repeated my long-term listening comparisons of the two preamps with a new sense of skepticism, with and without other listeners. But to get right to the point, I am still convinced that I can hear differences in the sound of these two preamps. Admittedly, the differences are not great, but to a music and sound enthusiast they are important.

With this dilemma in mind, I thought of my old Marantz 7T. Of course, to make it suitable for an A-B test, I had to readjust the RIAA feedback network. After this was done, an evening's listening confirmed the "edgy" sound and the listening fatigue. Even with fairly short listening periods, I found that I could consistently characterize the sound of the 7T. And here is the experiment that turned on the light for me: when I put the old 7T in an A-B switchbox test with either the DB or the SP-3A, there was no consistent difference to be detected, and if the switching was blind, no reliable identification could be made!

I do not believe that the switchbox is messing up the sound. I also do not believe that the differences heard under longer-term listening are in my imagination.

My hypothesis to explain this paradox concerns the nature of A-B tests and the human process of the perception of reproduced music. We do not hear music only with our ears. Listening is a process involving intelligence, and both short and long-term memory. When we listen to the sound of a musical concert reproduced through a stereo system, the sound reaching our ears is only a fraction of the total information perceived by a listener in the actual concert hall, and the signal from the stereo system is degraded and distorted in varying ways and degrees. When we hear the reproduced sound, our brains analyze it, compare it to previous listening memories, and identify it, tentatively at first. Then, as more clues are recognized, we reconstruct a listening experience which we enjoy as music. It is not the instantaneous sound striking our eardrums which we perceive, it is the comparison of what we have heard over the past few moments with what we remember of the past minute -- all in the context of the musical memories of a lifetime.

Now what does that complex statement have to do with A-B tests? It tells us that what we perceive at any instant of listening to a stereo system is what we are hearing at that instant conditioned by what we have been hearing for some time in the past. Whether that moment of sound appears to us as soft or loud, tonic or dominant, sweet or strident, major or minor, distorted or pure depends on what has preceded it.

We are all familiar with how an experienced listener can seem to perceive much more than the ear actually hears. A music lover, listening on a pocket transistor radio to a selection he knows well, is thrilled by a virtuoso double bass passage as a result of clues his ears detect from the upper harmonics of the double bass, which are perceived in his mind as the full sound he knows so well. In a similar manner I am always amazed how the stereo perception of our sense of hearing allows us to pick, out of the noise and confusion of a loud and crowded party, the one conversation or the music we want to hear, but when we block up one ear the din and the chaos return.

Along these lines, it has been shown that some aural perceptions register instantly while others require a longer listening time to be brought into focus. This is true of different kinds of sound distortions or aberrations. One well known with a long time detection period is the phenomenon generally referred to as listening fatigue, a common term in hi-fi literature. I believe that this syndrome is related to the amount of work the brain must do to sort out the real musical clues from the spurious error signals which come from various distortions. Our intelligence is capable of separating music or other familiar sounds from noise, distortion and other interference because of its memory and rational logic. After some period of time the brain appears to get tired of performing this task and the distortions begin to intrude upon our perception. Soon the listener feels the urge to turn down the volume, turn down the treble control, and eventually to turn the damn thing off !

A second type of sound perception which takes some time to form in the mind is stereo imaging, or the localization in space of the individual sources of sound in the recording. While a stereo signal can be told from a mono signal fairly quickly, determining whether two components have different separation characteristics takes a longer listening time, probably because the various stereo location clues are heard over a period of time and held in memory while the stereo image is created in the mind.

And what are the differences between components that are detected almost instantly? The Spiegel switchbox test tells us the answer: (1) loudness and (2) frequency response.

Now let us consider that we have two preamplifiers that we wish to compare and that we will use an A-B test. Let us say that the preamplifiers have differences in frequency response, type and amount of distortion, and in stereo separation and discrimination characteristics. We make adjustments so that they have identical loudness, frequency response, and polarity. We run the A-B test and cannot demonstrate a difference. Why do we not hear the differences in distortion and stereo separation? Because these sound characteristics are perceived over a period of time and recognized by building up in the memory a whole sequence of listening experiences in a time-related pattern. Say it this way: if we have two preamps with different distortion or stereo separation performance, how preamp A sounds to us with respect to these parameters at instant X in time depends on whether we have been listening to preamp A or preamp B for the time immediately preceding instant X.

From my listening tests, I have concluded that if you switch back and forth between two components with different distortion or stereo separation, the distortion of both units is more evident overall and the stereo image is more confused than when listening to either preamp alone. I suspect this is because the brain is confused when the stereo clues are not consistent, and because the distortions are more evident when they keep changing in character. This is why so many critical listeners believe that the switchbox is "messing up the sound." The switchbox is not messing up the sound, it is the A-B listening mode which is confusing our perception of distortion, stereo discrimination and perhaps some other parameters.

The nature of an A-B test is that it allows us to make an instantaneous switch between two components. It seems self evident that if we try to compare the sound of two components, the less time between hearing unit A and unit B, the better we can tell if there is any difference. This may seem like comparing two color samples by holding the color chips side by side. But it is not really the same. Music is a constantly changing signal. The next two seconds of sound are not identical to the previous two. And what we have heard before influences our impression of the present. How do we tell if a difference we observe in the present moment is the result of a switch in components or in the signal? The dilemma reminds me of a suggestion of a simple minded friend that the obvious way to compare two preamps is to listen to one in the left ear and the other in the right ear at the same time taking care to randomize the selection of ears! It takes an understanding of hearing to know why this does not work.

The history of A-B tests begins with speaker comparisons. Many years ago, hi-fi fans found that they could show dramatic differences in speakers by setting up an A-B switching arrangement. This helped dealers sell customers better speakers. Speaker systems have widely different frequency responses, and A-B tests emphasize this type of difference, which explains why this technique is used so often in speaker comparisons. When we compare amplifiers or preamplifiers, where differences in performance are principally in other parameters, it is not clear that A-B tests are the most desirable.

Now I can hear the proponents of A-B tests saying that an A-B test does not have to be an instantaneous comparison. You can listen to A as long as you wish, then switch to B and listen for ten to thirty minutes before making up your mind. But that is not how an A-B test works. You listen to A or B until you begin to hear a sound quality that is associated with one of the preamps as opposed to the other; then, to confirm your impression, you switch to the other unit. When you do not hear the contrast you expect, you are confused. I have never observed anyone participating in an A-B comparison who did not rely on the moment of the switch to confirm the difference they think they have detected. After being frustrated in this attempt several times, you abandon all rationality and switch back and forth hoping to hear something which pops out at you. Often when I have made A-B comparisons I have had the impression that after hearing a sound characteristic that I recognize, I can confirm the difference better by lifting the needle from the record, waiting a minute or two, and listening to the same several minutes of music again, on the other unit.

My hypothesis, in brief, is this: the nature of our hearing perception causes an A-B test to give a different weighting to the various distortions and aberrations in components than that which occurs in normal listening. Take, for example, loudness and frequency response differences of 0.1 or even 0.05 decibels which have been reported to be evident in A-B tests. In no way are such small changes of any real significance in a normal listening situation. But if an A-B test reveals them in preference to differences in stereo discrimination or listening fatigue that require longer listening times, then the A-B test does not evaluate what we are hearing in the real world.

Before leaving this hypothesis for further investigation, I would like to compliment Mr. Spiegel for his introduction of statistical methodology into listening tests. This approach should be continued and applied to other listening tests, of which there are many that can be devised to help separate fact from fancy. The science of audio advances both by intuitive imagination and by scientific skepticism.