THE BAS SPEAKER

BOSTON AUDIO SOCIETY

Vol. 9 No.11 AUGUST 1981

IN THIS ISSUE

This issue contains the report on the BAS amplifier clinic that was held last March. You will most likely find both the data and the conclusions of great interest. Local member J.K. Pollard made up an IHF standard reactive load for his part of the test, and the difficulties of this apparently straightforward task proved surprisingly formidable. Even more surprising, to some of us at least, were the frequency response errors that many highly-regarded power amplifiers displayed when running into this speaker-like load. J.K. also devised and performed a more sophisticated version of the IHF dynamic headroom test, and he has managed to untangle the complexities of this much-misunderstood test in admirable fashion.

The test program included subjecting the amps to various other load impedances, which gave some of them fits. Units were tested into 8, 4, and 2 ohm resistors and into capacitors of various sizes. (Members were asked to sign a release form prior to the test absolving the BAS of liability for fried components.) Conventional noise and distortion tests were done, too, of course. But the final stage for many of the amps was a comparative listening test which was set up and supervised by Mark Davis. While we must be suitably cautious about the results of these tests, they are bound to surprise some of our out-of-state readers as much as they did those of us who took part in the testing.

Elsewhere in this issue there is the usual assortment of short reports from members. One of these supports the listening conclusions of J. Peter Moncrieff at IAR, and another explores the economics of small production runs of electronic gear. There is a warning about an insidious threat to your sound system that may be coming from your coffee table, more ULM notes from George Androvette, and tantalizing news of a new ambience device from Holland. Finally, note the advance announcement of the special meeting on film sound engineering scheduled for February: details are on page 9.

Coming up soon: more cartridge tests from Audio Canada, and everything you always wanted to know about digital encoding.

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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, Boston, MA 02215, and mark it to the attention of Frank Farlow. Postage is 15 cents for either.

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. Volunteers should write to the Trapelo Road address or contact Brad Meyer.

Open Forum

ROLLING YOUR OWN

A suggestion has appeared on your pages occasionally (in a February 1981 report on the Amber amplifier, and in an earlier article on the workings of the Carver Sonic Hologram circuit) that you can save money by homebrew construction, copying the manufacturer's designs. Manufacturers deny it, claiming that, for example, by buying in quantity they pay less for power-supply capacitors than you would. Please let me give my reaction as a manufacturer and construction enthusiast: Balderdash!

Almost anyone who would attempt such a project can get "jellybean" parts (resistors and capacitors) free or at trivial prices through school or work. The sheet metal and wood cabinets use so little material that scraps will do. Most of the semiconductors, switches, and power-supply parts can be purchased from surplus stores. I built all the electronics in my system from scratch and have had to pay retail prices only for my power transistors. Thus, the "avid hobbyist" route is exceptionally economical.

For those without such resources, the possibilities are less attractive. Buying 0.9-cent resistors and five-cent capacitors from Radio Shack for 15 to 50 cents each can add up quickly; and the minimum order, shipping, and handling charges at mail order outlets can be costly. The cabinet and its human engineering are exceptionally difficult for those without tools and experience.

A third low-cost route is to buy a kit. There are two levels of choice here: the low-cost enthusiast kit, produced by Southwest Technical Products in their heyday, and currently by Phoenix Systems and my company, Symmetric Sound Systems; and the no-compromise companies such as Hafler and Heath. I've already discussed the tradeoffs between these two types of companies in an earlier contribution.

Further up the price scale a used component, when available, is frequently a good buy. Then there are discount houses, and at the top of the scale, the audio salons.

Let's use as an example my company's dynamic noise filter/expander combination (see "Radio-Electronics", March-April, 1981). The well supplied avid hobbyist could get all the semiconductors for about $7 and build the whole thing for about $40. (Even I could build one for less than they cost me to sell in hundreds.) As an extreme case, consider the $130 Logical Systems kit; it has less than $3 worth of semiconductors. If you pay for all the parts (using the "true cost" of junk box parts and wholesale R and C prices), it costs about $60. The inexperienced low-resources builder would probably pay about $90-100 for all his parts. In either of these cases, figure on spending 35 to 70 hours on your project. If you choose the "enthusiasts' kit", our ASRU, it will cost you $110 postpaid. An equivalent Heathkit is $200 plus shipping. Either of these will require about 10-15 hours of labor.

A used noise filter/expander will generally cost about 1/2 to 2/3 of the new price, when you can find one. Discount houses, though they carry almost everything else, seem to carry no noise filters (though Pioneer's expander is available). Audio salons will often charge list price for Phase Linear's unit ($400) or for dbx's 3BX three-band expander ($700).

The conclusions are fairly obvious. If you really enjoy hobby electronics, or are a starving student, or believe that anything less than gold-plated phono jacks sound "gritty," then roll your own and save $50. The $1/hour savings over the kit company product is nice, but an even bigger benefit is the fun of building the product from the ground up. If you can't spare the time to start from scratch, and/or don't have easy access to tools and materials, the $100 kit plus 10-15 hours labor provide top performance, and most people enjoy the labor. Building a kit has none of the potential frustrations of the scratch approach, and is especially attractive if the thought of a wall-plug transformer, or a unit only 10" wide, or a chassis having no bottom, revolts you. If even kit construction is something you just don't want to do, consider the obvious tradeoffs between the other routes.

Rolling your own electronics is often an economical, if not always optimal, approach for anyone who can debug the unit when it doesn't turn on perfectly. Persons wishing to see a catalog from my company may write to Symmetric Sound Systems, 912 Knobcone Place, Loveland, CO 80537.

-- Joe Gorin (Colorado)

SPEAKER CABLE AND EQUIPMENT REVIEWS

For over ten years I have read most of the underground audio publications, and I have used their recommendations to assemble a system that would have some chance of qualifying as respectable hi-fi. It seemed reasonable that different brands of electronics did indeed sound different, and that given the difficulties of auditioning components in the typical hi-fi store, it would be prudent to buy equipment favorably reviewed in the undergrounds. I have continuously upgraded my components but have never experienced the dramatic improvement in sound one would expect after reading so many rave reviews. I had just about decided that either: (1) preamps, amplifiers, and the rest of the chain of electronic do indeed sound the same, as claimed by Julian Hirsch and company (that's not really what he says --Ed.) or (2) my ears cannot discriminate the subtleties of sound so evident to the golden-eared reviewers. (I have no reason to believe that my hearing is in the least deficient.) This preamble is provided as background to a recent experience that may change my attitude toward equipment reviews and the search for better hi-fi.

Peter Moncrieff, in issues 9, 10, and 11 of his "IAR Hotline" (at last printed on a light background and not that eye-defying red) , has performed extensive testing of speaker cables and states that only Polk Sound Cable faithfully transmits the high frequencies. He further asserts that a combination of cables (the Polk for the highs plus a cross-four 4-wire Romex in parallel for the lows) must be used in order to accurately transmit the full audio spectrum to the speakers. In summary, low cable resistance, purity of conducting wire, and fancy configurations do not guarantee accuracy as claimed by the specialty cable makers. Moncrieff provides lengthy descriptions of the various cable types, and based on his listening tests, offers detailed and apparently logical explanations of how each cable affects the sound.

Well, I approached this review with suspicion, based on my recent experiences of having made large investments for minimal improvements, but I decided to spend $50 for two 28-foot lengths of the Polk Cable (my local retailer was discontinuing this cable in favor of other brands, most of which were more expensive). My old cable consisted of 30-foot lengths of 14 gauge zip cord feeding JR 149 Speakers and an M & K subwoofer. The result was, for lack of a better word, FANTASTIC. The high end suddenly became open and extended without the transient smear I had thought impossible to eradicate. The wire brushes on the Sheffield Drum record and the guitar overtones on the Mobile Fidelity Gordon Lightfoot album quite obviously sounded more like the real thing.

Associated components in my system include a Denon 103D cartridge in an SME III tonearm, an Oracle turntable, a Marcof PPA-1 pre-preamp, and the Apt preamp and power amp. Ah, the Marcof. Moncrieff reviewed it as being one of the worst head amps he's ever heard. So I borrowed a Denon HA-500 head amp, plugged it into my system, and WOW -- another improvement in clarity, imaging, and other non-imagined subtleties, not as dramatic as the cable, but nevertheless obvious. Tapes I had recorded years before on my Nakamichi 500 with a Dynaco PAT-4 Preamp and a Sonus Blue Cartridge had always sounded pretty much the same in A/B comparisons with ones made with my current system. However, after I changed the speaker cables, the same comparison resulted in a totally different sound. It appears obvious that zip cord, even in heavy gauges and possibly even in short runs, will certainly mask subtle differences between components, and maybe some gross ones.

This leads to several questions: (1) do other inactive devices, such as interconnect cables and A/B switches mask component differences? (again, Moncrieff says yes); (2) Do these inactive devices cause reviewers to prefer inaccurate components that offset their errors?; and (3) can any subjective equipment review be used as a guide without knowing all the specifics of the playback system, i.e. speaker cables and their lengths, absolute phase, interconnect cables, polarities, etc? This would appear to make the pronouncements of Julian Hirsch even more tenuous since his specific playback components, and their possible built-in deficiencies, are never known. At least with the undergrounds some information of this type is available and general biases can be extracted.

Few audiophiles have the time, knowledge, or access to an array of test equipment necessary to fine-tune their systems. I believe the average audiophile, no matter how dedicated, stands little chance of obtaining accurate results from his components unless he is sold a complete, scientifically tested and assembled package with exact lengths of speaker wire, matched cartridge and tone arm, accurate interconnecting cables and a detailed instruction book for assembly and checkout. The best most of us can do is to trust to luck and rely on bits of audio truth to filter through the underground publications and the BAS Speaker.

--Jay Clawson (Colorado)

CLEAN FURNITURE, DIRTY SOUND

From the August 1981 issue of "Hi Fi Answers" comes the assertion that one of the most pervasive and severe sources of poor contacts in stereo systems is silicone. It seems that the stuff migrates fiercely, both over a surface and through the air, so that if furniture polish containing it is used regularly anywhere near your system, the chances are it has gotten into connectors, relays, etc.

Silicone is a very good insulator; fortunately it is fairly easy to clean off, requiring only a good solvent that leaves no residue. Freon is one of the best, and is available from Radio Shack (as tape head cleaner, stock no. 44-1011) in handy small spray containers. However, if silicone gets into speaker relays, and the relays drop out while a signal is passing through them, the resulting arc will produce silicon dioxide (Si02) which is an even better insulator than silicone, and is much harder to remove.

"Hi Fi Answers" is published by Haymarket Publishing, Ltd., 38-42 Hampton Road, Teddington, Middlesex TW11 OJE, England. Overseas subscription rates are not given in the front of the magazine, but the rate in the UK is 16 pounds per year.

--E. Brad Meyer (Massachusetts)

A NEW AMBIENCE DEVICE FROM EUROPE?

I want to react to Cary Lu's article in the December, 1980 Speaker. He wrote that few persons would appreciate REAL concert hall sound in their living rooms, because (among other problems) environmental and mechanical noises from traffic, air conditioners, etc. would be recorded. Of course he is right in stating that no one wants these noises, but I hope nobody will conclude that our records are perfect now in simulating concert hall sound. Far from it! The airiness and sweetness (and occasional roughness) of strings so often heard in the concert hall have not, until now, been reproduced in a 1: 1 way. The sound from my own systems (Beveridges; four Quads) is so "sweet" that I can live with it, but it is not real; something is still missing.

I think the problem lies in the fact that we must listen to stereo, that is, with only two speakers. Despite any "tricks" one may play, stereo remains confined more or less to a single plane. I don't pretend to know everything about concert hall acoustics or concert hall realism in our living rooms. But I am familiar with the experiences recordings create in my mind, having made recordings in several halls, including the famous Concertgebouw in Amsterdam, which is near where I live. I regret I cannot copy the sound of this concert hall, even with my live recordings. Although these master tapes are in a class by themselves, and outperform any record in dynamics, impact, depth, and so on, they still do not sound like the hall.

I have done much experimentation with time delay systems, and find that the experience is spectacular during the first few minutes, but I give a sigh of relief when the unit is switched off. The effect is exaggerated; everyone is everywhere. I never experienced parts of the orchestra occasionally playing behind me in a concert hall!

I think that hifi is dominated by electronic engineers, and that acoustic engineers have not yet entered the hifi domain. Every time delay system made now (I cannot say anything about the future) is, acoustically speaking, wrong. I have also tried the Sound Concepts imaging device and in my opinion it works very well in an anechoic chamber if the one and only listener does not move his head.

But salvation is coming! Recently I heard a demonstration of an elaborate but affordable concert hall simulator, and there it was: spacious, life-like sound, without aggressiveness. Only when the device was switched off did I notice that the sound had become plain; there was no exaggeration. This unit, which is made in Holland (no, not by Philips or Heineken) is the one for me; the others are all toys.

I am very sorry that I am not allowed to tell you more about this simulator as patents are not yet granted, but it will be in production by the end of 1981 and will certainly be exported to the United States. The price will not be far from that of the Benchmark Acoustics device ($800), but it will include the necessary amplifiers. I am not the producer of this ambience simulator, neither am I related in any way to its manufacture; it was demonstrated for me, and my opinion was asked. As soon as it is possible to say more, I will inform the BAS about it.

Giap Tan (Holland)

(Note: If a time delay system produces the effect of everyone being everywhere, with part of the orchestra behind you, it isn't set up right. A correctly adjusted time-delay preserves a solid, stable stereo image in front with no wander. Of course the sad truth is that most delay systems, like most two-speaker stereo systems, are not optimally set up. -- PWM)

MORE NEWS FROM CALIFORNIA

I have more information on the Ortofon ULM55E, which I am falling deeper in love with daily. This cartridge weighs under 3 grams (by the balance-against-a-penny method) and is easy to find at any Dual dealer for an attractive price. Many dealers (I think foolishly) encourage their customers to have the ULM taken off new Dual changers and use a standard pickup instead. This results in a box of Ortofon ULMs accumulating in the dealer's stock room. Ask, and ye shall be rewarded -- perhaps at no charge, or for a token price. The ULM images better than any cartridge I'm familiar with (it replaced a

Sonus Blue and/or Decca Plum in my setup), and its astonishingly low mass improves the performance of any tone arm, hefty or light. It seems to track best at around 1.75 grams which, combined with the reduced total effective mass, contributes to any record player's tracking of warped discs and resistance to footfall skipping.

The ULM is difficult to install in a standard headshell, but not overly so. Its mounting platform is just over a half-inch wide, and with an X-acto blade you can easily make two small notches in the platform to allow the insertion of the screws to hold it in place. You must remove two plastic nipples from the top of the platform, and allow for about 1/16" space under it to accommodate the spring-loaded stylus retainer nipple that protrudes above the platform. If you do not allow this space, the button will be compressed against the surface of the headshell and the stylus will lock in place, necessitating the removal of the cartridge from the headshell to change the stylus. The stylus is protected by a nice-looking but unnecessary clear plastic flip-down guard, which I have removed to lower the mass still further.

As mounted in the Dual turntables, the ULM has a slip-on connector which fastens to the tone arm wires. If you leave this connector in place, it makes the cartridge body too long to fit into some arms and still have correct overhang. Besides, it adds unnecessary mass. The connector does have standard connecting pins, however, and if you remove it you'll have to crimp your arm wire lugs for a good fit over the tiny pins of the ULM. The wiring arrangement for these unmarked pins is (viewed from the back, with the the stylus down):

Upper left -- Left channel hot
Lower left -- Right channel hot
Upper right -- Common ground
Lower right

So far, I have added ULMs to (aside from the Decca, in which it works splendidly) two other arms. They are -- get ready to cringe -- the arm in an Elac Miracord 50H/II which I keep around to play records automatically, and the arm in my bedroom stereo's Lenco L-75. In the Elac the ULM gave a remarkable improvement in trackability over a Shure V 15/IIIHE, and in the Lenco there was some reduction in sensitivity to footfalls and much better tracking than I got with the old A-T 14S. Or course, these arms are both pretty massive, but the ULM's light weight makes an impressive difference.

It strikes me that this cartridge represents one of the few remaining genuine hi-fi discoveries, a true bargain in this era of price-fixing and look-alike design. Recent ads in the magazines indicate that Ortofon is now selling this cartridge with a standard mounting system, but it must surely be expensive that way. If you can scare up a couple of discards from a Dual dealer as I did, great! One warning, by the way: the ULM's stylus breaks easily. Careless handling will surely snap the cantilever or knock the diamond chip out of its bezel. Be gentle.

I don't know how many of you have seen Sharp's new vertical record player, which is built into a rather expensive compact system, but it's worth a look. This model combines idiot-proof operation (you simply drop the disc into a padded well and close the door to play it) with stylish vertical operation and straight-line tracking. It has two identical tone arms, each with its own magnetic pickup, which move across the disc on a common carriage; the motor reverses direction automatically at the end of side one, or at the touch of a button. Since the two arms move together, the side-one arm, which is visible through a clear plastic cover, serves as an indexing guide for either side. It's altogether rather like a Seeburg jukebox mechanism, but is much gentler to the records. The compact system contains a cassette deck and a tuner as well; both seem to work adequately. Unfortunately, it's not adaptable to 12-volt operation -- perhaps Sharp will correct this oversight in the future and come up with the perfect van or boat system for RV nuts. Also, it is available only in a package deal with two rather ordinary speakers.

A final note: I picked up a pair of Advent 3002 speakers yesterday to upgrade my bedroom setup, which consists of the aforementioned Lenco and a delightful old Fisher 250. While I won't say these speakers are state-of-the-art, their simple good looks and quite acceptable sonics make them a bargain at under $200 the pair. They have a single woofer about 7 inches in diameter in an acoustic suspension cabinet and a dome tweeter. They have neighbor-stirring bass, sweet highs, and more midrange than a two-way has any right to have. Good job, Advent. Quality control, as is usual with Advent products, is very good, although I did have to exterminate a couple of genuine New Hampshire silverfish when I opened the box!

- George Androvette (California)

July BAS Meeting

The July BAS meeting was a victim of cancellations and inadequate backup planning. Peter Aczel, the acerbic publisher of The Audio Critic and grandfather of the Fourier 1 loudspeaker, was originally scheduled to speak at the May 17 BAS meeting. It was understood that in addition to speaking about his magazine and its reviewing policies, he would also bring the Fourier 1 speaker and its designer for discussions and demonstration. But when it became known that Alan Hill would be on the East Coast at that time and wished to speak to the BAS about the Plasmatronics speaker system, Aczel withdrew. Busy with the many chores of getting the speaker into production (and preparing to demonstrate it at the June CES), not to mention the problem of the slipping publication schedule of The Audio Critic, Aczel requested that we re-schedule his visit for July; since we were eager to hear Hill's talk, we happily obliged. Arrangements were completed for Aczel's visit to the BAS on July 19.

But without warning, a few days before the July meeting we received a message from a member of Aczel's staff cancelling the visit. There was no explanation, nor any followup call to suggest a further re-scheduling -- a disappointing development since Aczel is a stimulating critic and the promise of the Fourier speaker was tantalizing.

As a substitute we hastily organized a tape-recorder clinic for members, intending to measure the performance of their tape decks and -- where practical -- to perform re-biasing and other minor adjustments. That plan was not worked out with sufficient care; we got bogged down in servicing a few of the tape decks, and time ran out before the basic measurements could be made on the remaining members' machines. We apologize to those members who brought recorders and waited in vain for them to be tested; our next recorder clinic will be better administered.

The open-discussion portion of the meeting was devoted mainly to an analysis by Brad Meyer of why it has proven unexpectedly difficult to bring the BAS Speaker's lagging production schedule up to date. When Brad took over as Editor of the Speaker two years ago the publication schedule was running about four months behind, and despite a great deal of effort -- and the help of an occasional substitute editor or double issue -- it still is. The problem, it appears, is at least partly due to "the system" -- the arrangement by which we have spread out the work among a larger number of people in order to reduce the burden on any one person.

Under the old regime, raw copy (members' letters and articles, meeting reports, In the Lit, etc.) was forwarded from the P.O. Box to the editor, Mike Riggs, for sorting and editing. All of the edited material for an issue, when completely assembled, was given to Joyce Brinton in Boston, who drove it to Wayland; then Jim Brinton delivered it to our typist, Julie, located in the town of Harvard about 30 miles west of Boston. She typed the entire issue on large galley sheets, leaving appropriate gaps for illustrations and the bold-face headings. Jim then delivered the galley sheets to Bob Borden in Lexington, who personally oversaw all of the rest of the production: proofreading, correction of typographical errors, creation and pasteup of bold-face headings, pasteup of illustrations, page numbering, any cutting and pasting needed to correct the order of material or adjust its pagination, delivery of the issue to the print-shop, pickup of the cartons of printed issues, etc.

The key element is this: once Mike Riggs turned the edited copy over to Joyce, he had no further involvement with an issue and could devote his time and energy exclusively to prodding writers and editing the next issue. All of the transportation of the copy here-and-there, and the many production chores, were handled by others. But after five years of heroic and largely unrecognized effort, Bob Borden resigned from his untitled job as production manager of the Speaker. So in the system that developed, Brad Meyer not only took over as Editor but also took on many of the tasks that had been efficiently unified before. Under this system, after prodding writers, assembling contributions, and editing all of the raw copy by hand -- and typing some contributions that come in handwritten -- he drives out to Harvard to deliver the assembled copy to Julie for typing, meanwhile making arrangements for preparation of any illustrations, then retrieving them and delivering them to Cassandra, our layout artist in Acton. The typed galleys are also delivered to Cassandra, who does the pasteup and produces the boldface headings with guidance from Brad; meanwhile he also proofreads the galleys, which then go back to Julie for insertion of any required corrections, after which Brad delivers the issue to the printer in Cambridge. Thus Brad spends about as much time driving materials around, and overseeing production matters, as he does in editing. A more efficient system is needed.

[Postscript: As a result of this analysis, during the following month Peter Mitchell appointed himself production manager and instituted a new system intended to eliminate a lot of the transportation delays and built-in inefficiency of the above arrangement. The new system is based on a Heath/Zenith Z89 word-processing microcomputer. Rather than hand-editing copy and then having the galley sheets final-typed, which creates a need for a later round of editorial proofreading and typing corrections, the raw contributions from members are typed directly into the computer (by a hired typist) without editing. Then the editing is done on-screen, and typos are corrected at the same time that the copy is edited for content, clarity, and grammar. All corrections are made, and sections are re-shuffled, electronically, with no retyping of pages required. When the editor is satisfied, his job is done and he can start working on the next issue; meanwhile the galley sheets are final-typed by the computer's printer, requiring only the pasteup of headings and illustrations before delivery to the print-shop. (Volunteers, anyone? We very much need a pasteup artist who lives or works close to Cambridge.) Of course there have been start-up delays in this new system which have compromised its advertised efficiency, but a speed-up in Speaker production is now a prospect as well as a promise.]

-- Peter Mitchell

FEBRUARY MEETING PLANS

Under the new system of dues approved at the June meeting, meeting announcements will be sent only to those members who pay for their production and mailing, which means that fewer of you will get the notices. We would like to print advance schedules here, but our guest speakers are sometimes planned only a month or two in advance. In the case of the February 1982 meeting, however, the planning is firm and its content is sufficiently special that we want everyone to know about it.

The meeting will be held at the Wellesley Community Playhouse theater, which features a state-of-the-art cinema sound system designed by John Allen. Tom Holman, chief sound engineer at Lucasfilm (the studio responsible for the "Star Wars" and "Raiders of the Lost Ark" sagas) will discuss and demonstrate the recording and mixing procedures used in making modern film soundtracks; a multi-channel mixer will be brought in and synchronized with the projector for the demonstrations. We can't tell you here which feature-film excerpts will be shown, but it promises to be interesting.

The meeting is scheduled for SATURDAY, February 27, starting at 9:00 AM sharp and running until noon. If the weather is severe the meeting will be postponed to Sunday morning. Directions: from Rte. 128 go West on Route 9 to Route 16 (Washington St.), then south about mile. In order to defray the cost of hiring the theater and projectionist, there will be an ADMISSION FEE of $2.00 per person. This is a joint meeting with the Audio Engineering Society's Boston section, whose members will pay the same fee. If you have any questions, contact the BAS; please do not call the theater.

BAS Amplifier Clinic

INTRODUCTION by Alvin Foster

As a consumer organization, the BAS sponsors occasional test clinics, in which members bring in their equipment for bench and listening tests. Unlike "Consumer Reports", we do not specify a "best buy" or even rate units according to the best value per dollar. Our goal is simply to find out how closely our equipment conforms to its specifications in actual use, and if possible to highlight units which produce outstanding sound.

The amplifier clinic was held on March 29, 1981 from 1:00 PM, when the first units arrived for testing, until about 10:00 PM, when the last of the listening tests was concluded. To obtain our samples we mailed out, in early March, over 500 letters to local members soliciting units for testing. The letter described the tests to be performed; members were encouraged to bring along several pairs of the largest fuses recommended by the manufacturer, A disclaimer whose purpose was to absolve the BAS of responsibility for damage to the units under test was also included.

From the flood of replies I selected thirty-seven amplifiers for testing as representative of units owned by the membership. To minimize duplication, I generally accepted only one model of a particular brand. I also accepted the offers of a couple of local retailers to supply examples of some well-known high-end brands (Audio Research, Mark Levinson, Bryston) for inclusion in the tests.

The tests required considerable equipment and skill. Five BAS members, including myself, who owned or had access to the necessary test equipment agreed to be table chairmen. The responsibilities of each table chairman included designing meaningful tests that would still be efficient enough to allow the thirty-seven amplifiers to be tested in less than nine hours and, as far as possible, to orient the tests to reflect real-world operating conditions. The table chairmen wrote separate reports on their results. What follows is a blow-by-blow account of what happened.

NOISE MEASUREMENTS by. E. Brad Meyer and Peter Mitchell. (Report by EBM.)

At the first test station measurements were made of turn-on transients, mechanical noise, electrical noise and DC offset. First, each amplifier was "logged in". The brand, model, and serial number, as well as the purchase date and the price paid, were recorded on a data sheet which traveled with the unit throughout the tests. A short length of speaker wire with a GR connector (double banana plugs) was attached to the left channel speaker output, allowing quick connection to the various loads used at the other test stations.

The left channel output was connected to three devices in parallel: an Ivie IE-30 real-time analyzer, an oscilloscope, and an AR-4X loudspeaker. While the amp was turned on for the first time the tester watched the output on the scope while simultaneously holding his ear to the woofer of the speaker. It proved necessary to make the observation carefully the first time the unit was switched on, because after that the power supply capacitors tended to be partially charged and the turn-on transient was (mostly) smaller. The audibility of the turn-on noise was initially rated on a three-point scale as inaudible, audible or very audible. In practice it was found necessary to add a fourth category -- barely audible -- as many units could be heard at close range but would be inaudible in most listening situations.

It is possible for an amplifier to send an appreciable amount of current into the loudspeaker every time it is turned on without making any noise. A turn-on transient with only low-frequency components can push the woofer cone against both stops in succession, and do it slowly enough that you'll never hear anything. You should know if your amp is doing this, especially if its behavior at turn-on changes, as that may presage more severe power supply troubles. (It's easy to check your own amp: just take the grill cloth off your speaker and watch the woofer as the unit is turned on.) The scope was set to read about five volts at maximum deflection. The peak excursion of the waveform, if significant, was noted.

At low levels the ear is much more sensitive to upper midrange frequencies than to bass; that's why the A-weighting curve is good for low-level noise measurements. Audible turn-on noise accompanied by a small (or absent) peak voltage reading indicates that the transient is brief, that is, contains mostly high frequencies. A big spike that is inaudible is infrasonic.

Next, the degree of mechanical noise made by the amplifier was evaluated with a specially developed noise probe survey technique, using a small pressure transducer in conjunction with a sophisticated acoustical-analysis computer. That is, the tester held his ear as closely as possible to the unit, and moved his head all around to see if he could hear anything.

The significance of this test to the user is hard to know accurately, because whether the noise will be audible in the user's home installation depends very strongly on where the amp sits in the room, and equally strongly on the background noise level. In a big-city apartment, during the day, in a room with a single-glazed window opening onto a busy street, even amplifiers with fairly noisy cooling fans will be inaudible. In a rural environment, in a fairly live room, at night, the slightest transformer hum becomes annoying. The "measurement" was complicated by the fact that the environment where the bench testing was carried out was more like the former than the latter. Still, amplifiers with slight but occasionally significant mechanical noise, like the Audionics CC-2, could be heard under the test conditions. The rating scale is the same as for the turn-on thumps.

Next, the electrical noise was measured with the Ivie. The A-weighting curve was used, and the figure given is the number of dBA below 1 watt into eight ohms. If you

TABLE 1

Amplifier	Mech. Noise Audible?	Turn-on noise Audible?	Turn-on noise Volts	DC Offset	Elec.Noise A-wtd. re 1 watt	IHF Dynamic Headroom 20ms Burst	IHF Dynamic Headroom 100ms Burst	Baseline restoration 20 ms Burst	Baseline restoration 100ms Burst	Notes
Amber 70	No	Barely	1V	9 mV	-86 dB	1.3dB	1.3dB	0 ms	0 mS	1
Apt 1	No	No	0	0	-92	3.3	3.0	0	0	
Arcam A-60	No	Yes	0.5	14	-85	3.1	2.5	30	30	
Audio Research D-60	No	No	3	1	-93	2.1	2.1	0	0	
Audio Research D-90	No	No	0	(note)	-73	1.0	0.9	0	0	2
Bryston 2B	No	No	0	20	-89	2.8	2.2	0	0	
Bryston 3B	Barely	No	0	11	-91	2.2	2.1	0	0	
Carver M400	Barely	No	0.1	1	-84	2.2	2.2	10	10	3
Crown DC 300A	No	Very	2+	2	-97	2.5	2.2	0	0	
DB Systems DB-6	No	Barely	0.4	4	-99	3.8	3.6	20	20	
Dunlap-Clarke 500 mod.	Very (buzz)	Yes	3	0	-86	1.3	1.1	0	0	
Dunlap-Clarke 1000	(note)	Barely	4	0	-81	1.3	1.2	20	20	4
Dyna 150 kit	Yes	Yes	1	42	-100	1.4	1.2	30	50	5
Dyna 150 (mod)	Barely	Barely	1	60	-87	2.5	2.2	30	50	
Hafler DH-200 kit	No	Yes	2	1	-98	3.2	2.9	0	0	
Hafler DH-200	No	No	0	16	-95	3.0	2.6	0	0	
Integral Systems 200	No	Barely	0.2	21	-81	--	--	--	--	6
Leach II kit	No	Barely	4	89	-99	3.5	3.2	0	0	
Levinson ML-2	No	No	6	5	-87	4.5	4.5	0	0	7
Levinson ML-3	No	No	0	0	-82	3.5	3.1	0	0	8
Marantz 9	No	No	0	0	-90	2.2	2.2	0	0	9
Marantz 15	No	No	0	0	-93	3.0	2.6	40	40	
McIntosh MC2105	No	No	(note)	13	-80	--	--	60	60	10
NAD 3020	No	No	2	NT	-100	4.0	3.6	20	20	
Nikko Alpha 3	No	No	0	11	-94	2.3	2.0	0	0	
Phase Linear 700	No	Very	NT	NT	NT	2.9	2.4	60	60	
SAE Mk 31B	No	No	0	32	-100	3.0	2.6	0	0	
SAE X25A	Barely	No	0	8	-98	2.4	2.1	0	0	
Sony 3200F	Barely	Yes	0.5	27	-103	2.6	2.2	50	50	
SWTP Tiger	No	No	0	3	-85	1.9	1.9	20	100	
Threshold 400A	No	No	0	33	-89	NT	NT	NT	NT	
Yamaha CR-620	No	Barely	2	0	-89	2.4	2.2	40	40	

NT = Not Tested

1. The Amber Series 70 showed what looked like low-level program material in the Ivie analyzer during the noise measurement. It looked like WBUR, whose transmitting tower is mounted on the roof of the building in which the tests took place. A little way into the second test, the Amber appeared to die of power supply failure. Then when its owner got it home, it fired up properly and sounded fine. It seems to have been sensitive in mysterious ways to the high RF field (which was certainly unusual) in the building. At least, that's the only explanation we can come up with.
2. The Audio Research D-90 exhibited a 1 Hz oscillation of about 20 mV P-P shortly after it was turned on; then it stopped oscillating and began to put out about the same signal level in the form of low frequency noise. Its noise spectrum included a 240 Hz hum peak at -69 dB re 1W.
3. The Carver M400's commutator noise was visible in the spectrum analyzer. Its noise level was low enough, though.
4. The Dunlap-Clarke makes lots of noise when its cooling fan is on high; when the fan is on low speed, or off, it is inaudible.
5. This amp had a very loud turn-OFF transient whose peak was over 2 volts.
6. Protection circuit tripped, preventing headroom test.
7. Mono amp, $3000 ($6000/pr. for stereo).
8. Stereo amp, $5000, weighs 63 kg (140 lb).
9. Tube amp. Tested in triode and pentode configurations, both measurements about the same; headroom 0.3 dB lower in pentode mode.
10. At turn-on the output went 2V positive, then 1.5V negative, and the woofer moved in and out inch. The noise level varied with gain setting; -74 dB with gain all the way up, predominantly hiss. This amplifier could not drive the IHF reactive load at its rated power, so the headroom measurement could not be made.

wish to translate the measurement to dBA below 1 volt, add 9 dB to the figure in the table (that is, make the figure worse by that amount, so that a figure of -77 dBA re 1 watt becomes -68 dBA re 1 volt.)

The results are summarized in Table I. It's gratifying to see how quiet modern power amps are. Only the Integral Systems, the big Dunlap-Clarke and the McIntosh had any significant amount of noise, and even these wouldn't bother most people, in most installations.

It's hard to say how fussy one should be about DC offset, but anything over about 20 mV is considered gauche, if not necessarily ruinous to the sound. The two Dyna 150s and the Leach were definitely beyond the pale.

FREQUENCY RESPONSE AND DYNAMIC HEADROOM WITH A REACTIVE LOAD by J.K. Pollard

A frequency response deviation of a fraction of a dB is known to be audible if it extends over a broad frequency range, so it is of interest to know what response differences amplifiers may exhibit under real-world operating conditions. The response test at our second test station was similar to a standard frequency response measurement, except that the 8 ohm non-inductive resistor ordinarily used was replaced by a reactive load intended to simulate the impedance of a typical dynamic woofer. Testing was carried out at modest output power, about 14 watts. The reactive load was constructed using component values specified by the IHF ("Standard Methods of Measurement for Audio Amplifiers," IHF-A-202, 1978, p.18) as shown in the accompanying illustration (Figure 1).

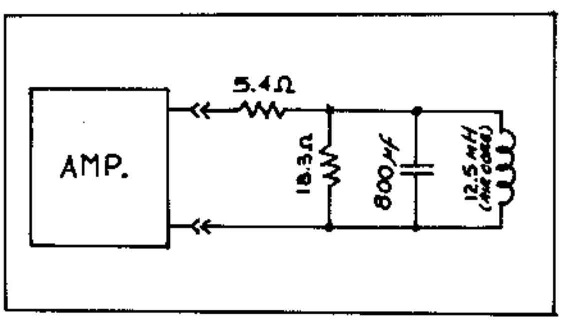

FIGURE 1

IHF Reactive Load

Although the components used in our load were within ±2% of the nominal values, the resulting impedance curve differed significantly from IHF specifications. This is hardly surprising, since the IHF spec is based on ideal components, that is, inductors and capacitors with zero resistance. Even though our 12.5 mH inductor was wound with 12 gauge wire (available from Transcendental Audio, 6796 Arbutus St., Arvada CO 80004), it still had 0.87 ohms resistance, giving it a Q of about 4.5 at resonance. Because of these shortcomings of real-world components, the impedance peak was shortened and broadened quite a bit compared with the IHF-specified 23.7 ohms.

The abovementioned quirks aside, the test equipment behaved very well. Its overall flatness was ±0.05 dB, allowing us to use an expanded scale factor of 0.25 dB per division on the plots.

The results are presented in Figure 2. We were frankly surprised at the large number of amplifiers showing significant frequency response deviations. Only about a third of our sample had perfect (±0.1 dB) frequency response over the full 20 Hz to 20 kHz range. About a quarter showed errors greater than 0.1 dB but less than 0.25 dB. Another quarter fell in the 0.25 to 0.5 dB range, while 4 (13%) had errors of more than 0.5 dB. The errors were of three types: (1) low end rolloff, (2) a peak at the 60 Hz resonant frequency of the IHF load, and (3) high-end rolloff.

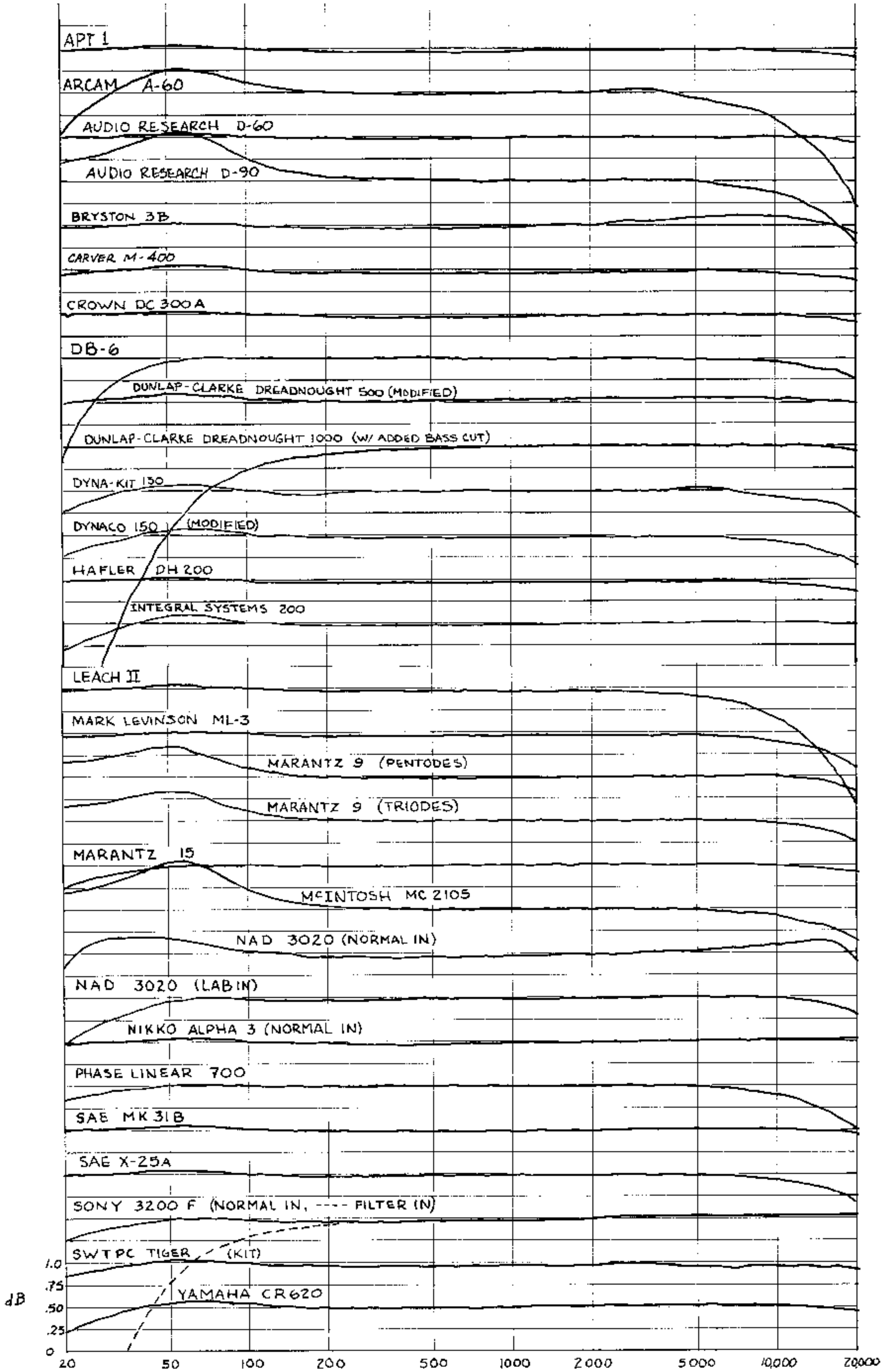

FIGURE 2. Amplifier Frequency Response into IHF Reactive Load.
Note expanded vertical scale: 0.25 dB per division.

The low-end rolloffs were clearly associated with coupling capacitors or with infrasonic filters; for example the DB Systems amplifier is quite flat down to 50 Hz and its infrasonic filter is down 1 dB at 20 Hz. The low-end rolloffs were not particularly significant except for the one amplifier (a Dunlap Clarke 1000) which had been especially modified with extra bass filtering for disco use. The peak at the load resonance in several amplifiers simply means that the amplifier's damping factor is not very high, a characteristic of the tube units and a few of the smaller solid-state amps. The presence or absence of such defects could probably be brought out in a listening test designed for the purpose, but could easily pass unnoticed on ordinary program material. Only the amplifiers with high-end rolloffs were immediately identifiable to listeners, and then only with program materials featuring cymbals and similar percussion.

DYNAMIC HEADROOM

Dynamic headroom is a measure of the amount by which an amplifier can exceed its continuous power rating during a brief interval. Although the concept of headroom has been recognized for decades, it was only as recently as 1978 that a standard test procedure was specified by the IHF. This test calls for a 1 kHz sine wave whose amplitude remains constant for 480 ms, and then increases by 20dB for 20 ms.

In carrying out the test, one simply connects the dynamic headroom signal generator to the amplifier and observes the amplifier's output on a scope connected across the load. The amplitude of the test signal is raised to the maximum possible without clipping. The ratio between this RMS amplitude and the amplifier's rated continuous output, expressed in dB, is the dynamic headroom.

Figure 3 shows the equipment setup used at the BAS clinic. The dynamic headroom test generator (DHTG) was built by the author through an evolutionary process beginning with the circuit devised by Doug Farrar and published in Radio-Electronics, October, 1979. Farrar's circuit provides basic capability for under $20 in parts. However, as Al Foster and I discussed the forthcoming clinic with various BAS members, suggestions for enhancements of the basic test proliferated. Bob Carver noted that several of the state-of-the-art discs he had examined contained peaks as much as 100 ms wide (e.g. cannon shots).

Fortunately, Farrar's circuit lends itself to operation in a 900 ms/100 ms mode with the addition of only a DPDT switch and a few pieces of wire. Tom Holman pointed out that in the course of his experiments, he had found a few amplifiers which misbehaved when driven to clipping with an asymmetric signal, i.e. one with DC offset on the peak portion. Incorporation of Holman's suggestion required complete replacement of the analog portion of Farrar's circuit with a DC-coupled design. Finally, in order to speed up the test procedure, three 10-turn pots were added: (1) to normalize gain at rated continuous output power, (2) to add a calibrated amount of additional gain and (3) to add a calibrated amount of DC offset. Achievement of the desired accuracy in these controls (±1%) required the addition of tightly regulated power supplies (a Lambda LXD-3-152).

In the actual test procedure the output of the amplifier under test was first normalized, i.e. raised to its continuous rated power as specified by the manufacturer. The DHTG was then switched to its test mode and the gain increased with the calibrated pot until clipping was observed on any part of the waveform. Gain was then reduced until the clipping had just disappeared. (A dual time-base scope was used in the mixed sweep mode so that any part of the burst wave form could be examined in detail.) The increase in unclipped signal level above the rated continuous output, expressed in decibels, is the measured dynamic headroom. This procedure was repeated in the 100 ms burst mode and a second, usually lower, headroom measure was recorded.

After the headroom tests, the DHTG gain was returned to the normalized value and the offset test begun. Figure 4 illustrates what the waveform of the dynamic headroom test looks like, with no offset and with 100% offsets of positive and negative polarity.

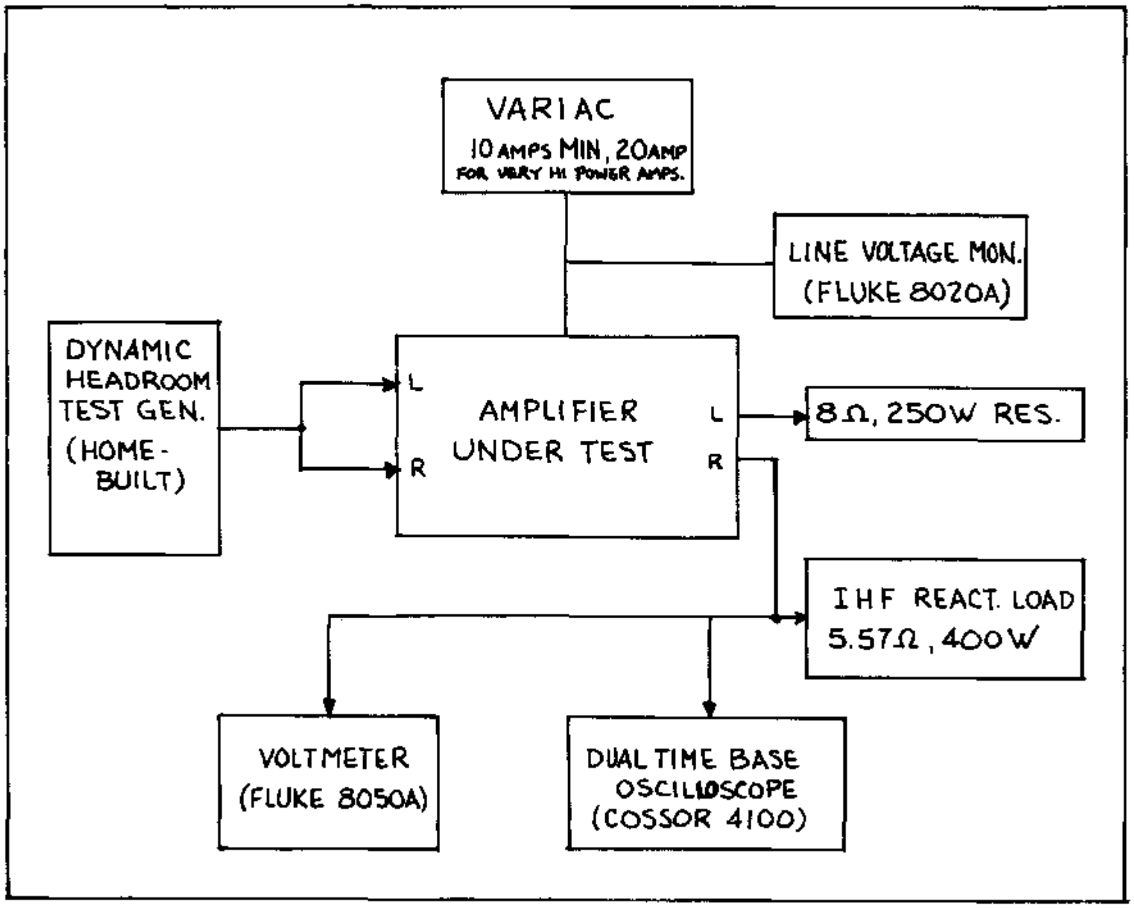

FIGURE 3

Headroom Test

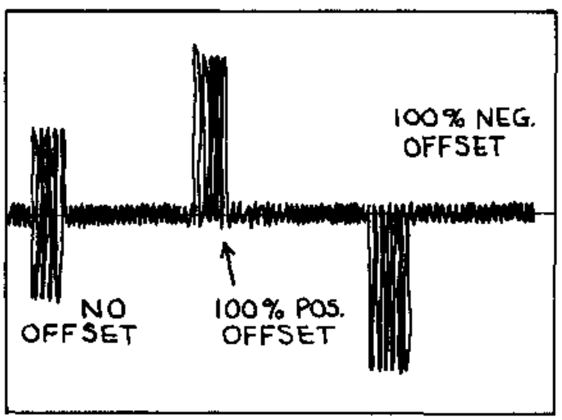

FIGURE 4

Dynamic Headroom Waveforms

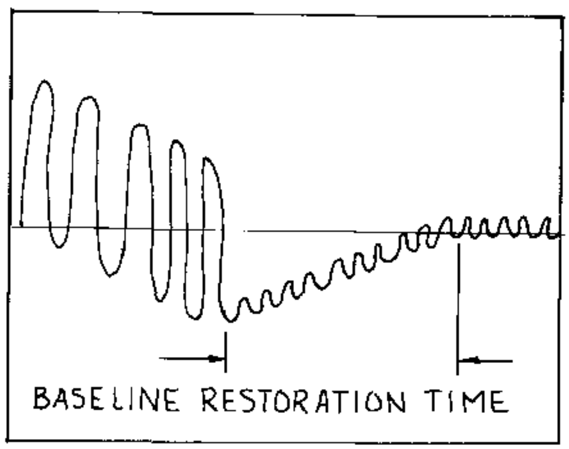

FIGURE 5

Measuring Baseline Restoration

The dynamic headroom measurements are presented as part of Table 1. Our sample of 32 amplifiers showed dynamic headroom figures ranging from under 1 dB to more than 4 dB. It must be stressed that these figures are not directly comparable with manufacturers' data or other published reports since they are based upon the IHF reactive load rather than the 8 ohm load ordinarily used. At 1 kHz, our load had an impedance of 5.57 ohms, almost purely resistive. The principal justification for the use of this impedance is that it is more nearly representative of the mid-band impedance of the larger acoustic suspension systems favored by many BAS members. Thus the headroom figures in Table 1 provide a better estimate of the peak powers available into a typical speaker than would data from an 8-ohm load.

Now, an amplifier whose maximum output voltage is exactly the same into 5.57 and 8 ohms would be delivering 1.57 dB (almost 50 per cent) more power into the 5.57 ohm load. Thus one would expect 1.6 dB higher dynamic headroom ratings from our test than the standard test. (Remember the rating is referred to the manufacturer's specified continuous power, which is an 8 ohm rating.) However, because peak voltage ordinarily falls along with load impedance (because of sagging in power supply voltages as more current is drawn), there really is no simple quantitative relationship between dynamic headroom ratings at 8 ohms and those at 5.57 ohms.

In the offset test, no really bizarre behavior appeared, that is, nothing comparable to the instabilities Holman found (BAS Speaker, October/November, 1979). Most of the amplifiers showed the effects of capacitive coupling at one or more points in their circuits and produced outputs which looked like Figure 5. The time required for the baseline waveform to re-center itself around 0 VDC is designated "Baseline Restoration" time in Table 1. Values ranged from 0 to 100 ms in our sample; i.e. in some amplifiers DC voltage offsets are created by asymmetric audio signals and take as much as a tenth of a second to return to normal. About a third of the tested amps had zero restoration time; i.e. the output looked like the input, unaffected by the waveform asymmetry.

Interpretation of these results are incomplete at this writing. We simply have not established a correlation between the decidedly unfaithful response patterns of most amplifiers to asymmetric signals (as observed on the scope) and their audible consequences (if any). Some members have suggested that inability to reproduce an asymmetric waveform accurately is associated with the sensation of "muddy bass", but this remains to be proven scientifically. I hope to explore this further and would be pleased to discuss the matter with other BAS members who are inclined to experiment.

DISTORTION AND INPUT IMPEDANCE by Alvin Foster

The purpose of the distortion test series was to try to find any objective test which would be helpful in identifying amplifiers whose sonic difference may be due to the introduction of unwanted signals. My search for "revealing" tests included writing to leading amplifier designers (Bob Carver, David Hadaway, Tom Holman, etc.) and researching the current literature. The IHF manual, "Standard Methods of Measurement for Audio Amplifiers", was very helpful in outlining the importance of maintaining identical testing conditions for each unit.

My choices finally boiled down to three: 1) total harmonic distortion (THD), 2) input impedance, and 3) IHF intermodulation distortion (IHF-IM), which uses a composite signal composed of two relatively high frequency sinusoidal signals.

Harmonic Distortion. The THD test bench consisted of a Hewlett-Packard Model 333A Distortion Analyzer and a modified Heathkit IG-18 Sine-Square Wave Audio Generator. The amplifier under test was fed a 1 kHz sine wave (whose distortion is less than 0.01%) while both of its outputs were connected to 8-ohm resistors and to the input of the HP-333A. Each amplifier was driven to an output level of ten volts (12.5 watts) , a level which will drive most speakers to an SPL of between 95 and 100 dB at 1 meter. To minimize the possibility of contamination of the readings by hum, the built-in 100 Hz high-pass filter in the 333A was engaged. Hewlett-Packard conservatively rates the distortion limits of the 333A as less than 0.03%; however, figures as low as 0.01% can be verified.

Input Impedance. If an amplifier's input impedance (the resistance and capacitance) is too low for the preamplifier being used with it, the sound may be too thin or bass shy. The classic Marantz 7 preamp, for example, sometimes sounds that way. The high output impedance of the 7 requires a corresponding input resistance in excess of 200K ohms. Most solid state amplifiers are, therefore, incompatible with the Marantz 7.

To measure input impedance I used a bridge invented by Mark Davis. The instrument measures resistance to an accuracy of 10% and capacitance to within 20%. The advantage of the Davis device is its ability to measure the amplifier while it is on and in the circuit. Most preamplifiers are designed to drive an impedance as low as 10K ohms and a capacitance of 600 pF; therefore, any amplifier exceeding these specification passes this portion of the test.

IHF IM Distortion. A composite signal composed of two relatively high frequency sinusoidal signals (18 and 19 kHz) was used to measure the IM components of the amplifiers. The signals were of equal amplitude and drove the amplifier to an output level of 12.5 watts. Again, each output of the amplifier was loaded with an eight-ohm resistor. The Heath IG-18 and a Wavetek model 30 were used to supply the composite signal for the test. The two signals were fed into a passive mixer, and from there to the amplifier. Connected to the output of the amplifier was a Quan-Tech Laboratories model 303 wave analyzer set to 1000 Hertz, the frequency of the second-order distortion component, f1 minus f 2.

RESULTS. While selecting distortion tests for the amplifier clinic I decided that those for my test station were to be somewhat conventional and unstressful, making it a sort of a resting place where the amplifiers could cool down between jolts. The resulting specifications, I concluded, would also reflect the type of figures manufacturers love to quote! The measurements I obtained are included in Table 2.

Harmonic Distortion. Of the 32 amplifiers submitted for testing only twenty-nine made it to the third stop in the test chain. For some, table one proved to be too much: The Amber blew fuses (but performed satisfactorily when its owner took it home) , one of the two Apts blew its fuse twice (but it, too, performed satisfactorily when returned the next day to the factory), the Audio Research D-90 required a floating ground which my test instruments lacked, and for some unknown reason the Threshold 400A was not tested.

Thirteen of the amplifiers, or forty percent of our sample, had distortion below the residual of our analyzer. The remaining units did well and/or were not tested. The Levinson ML-2, Hafler (kit), Marantz 15, NAD, Audio Research, and the SAE Mk 31B exhibited unusually high levels of harmonic distortion which I later traced to a flaw in my testing procedure. The spurious results were somehow related to an interaction between my Wavetec, Heathkit, passive mixer, and the particular amplifier under test. After I became suspicious, I retested about three of the units which had previously measured poorly; their results were now excellent. Due to time limitations, I was unable to retest the other six units.

Impedance. The purpose of this test was to isolate those amplifiers that had an unusually low or high input impedance. The latter would render the amplifier compatible with all preamps, while a low input impedance could cause some preamps to misbehave.

The average input resistance and capacitance for the 29 units I measured was 62K ohms and 250 pF. The Crown amplifier was not included because I lacked adaptors for its phone plug inputs. The Carver had the lowest input resistance. (15K ohms) and the Arcam and Marantz 9 the highest (200K ohms). The lowest and highest input capacitances were measured in the Integral Systems (25pF) and the Levinson ML-3 (525 pF) respectively. None of the units should, as a result of input impedance, cause any reasonably designed solid state preamplifier to misbehave.

IHF-IM. None of the tested units exhibited any measurable intermodulation distortion when driven by two high-frequency sine waves. On most of the amplifiers I choose to examine for all possible distortion combinations: f1+f2, 2f1-f2, etc. However, no IM distortion was found. These results are consistent with my similar findings two years ago at the BAS Preamplifier Clinic. It seems that whenever well designed units are driven with signals within the audio band and at output levels consistent with their design limitations, no measurable high frequency IHF-IM distortion can be found.

CONTINUOUS POWER AND REACTIVE LOAD STABILITY by David Hadaway

The purpose of the tests at this station was to investigate the behavior of each amplifier under various output load conditions to determine if it would manifest any form of undesirable behavior that would cause it to sound different from an "ideal" amplifier. In addition to the conventional resistive load, various capacitive loads were used to determine their effect on stability, frequency response and power output. A measurement of peak crossover distortion was made in order to investigate if high peak-to-average spikes could cause audible effects that would not show up on the more typical "average" type of distortion measurements. As will be seen in the following discussion, a certain amount of discretion was used to decide which tests would be done on each amplifier (I didn't want to get a reputation for destroying people's amplifiers). As it turned out, aside from a small amount of smoke emitted from one unit, all amplifiers passed safely through this stage of testing.

The signal source for all measurements was the ultra-low distortion Morrey/Heathkit IG-18 oscillator. The power line voltage was taken from a large variable autotransformer and monitored by an RMS-responding voltmeter, the VIZ 120B (as a moving vane type meter it is an old-fashioned way of reading true RMS.) Since all tests were done with pure sine waves, an average-responding voltmeter was satisfactory. The Ballantine 303 AC voltmeter was chosen because of its flat response and expanded dB scale. Monitoring was done with a B & K 1470 Dual Trace 30 MHz oscilloscope. All test equipment had either floating grounds or adaptors to isolate from the power line ground. This was to avoid possible hum or stability problems with grounded amplifiers.

The first test was of maximum power at 1 kHz into 2, 4, and 8 ohms. The load used was a precision non-inductive load bank constructed by J. K. Pollard. Because time was limited only one channel was driven and tested. The 2 ohm load was applied only if the owner of the amplifier consented.

When large amounts of power were being delivered into the load the power line voltage would drop as much as 8 volts from the nominal 120. At first I was increasing the voltage on the autotransformer to compensate for this. However, I realized that in the hectic pace of testing I might remove the signal input before reducing the voltage and it could jump to almost 130 volts. In the interest of safety the voltage was set to 120 for no signal and allowed to sag under load. The voltage was noted at the same time as the power into the load was measured so an approximate correction factor to 120 volts could be applied.

TABLE 2

	THD @ 12W	INPUT IMPEDANCE ohms	INPUT IMPEDANCE pF	Continuous power, 1 kHz, one ch. driven 8 ohm	4 ohm	2 ohm	20 kHz clip-ping	Sta-**bili-**ty w/ cap. load	Frequency response w/2uF cap. 7.5	15kHz	Cross-over dist. at 12W	Notes
Amber 70	NT	NT	NT	NT	NT	NT	NT	NT	NT	NT	NT	
Apt 1	um	35K	30	153	223	273	ok	ok	0dB	+1dB	0.02%	1
					146	197						2
Arcam A-60	.07%	200K	375	50	73	NT	S	ok	0	+0.1	0	
Audio Research D-60	m.e.	48K	200	73	136	NT	S	0.1uF	NT	NT	0	3
Audio Research D-90	NT	80K	125	NT	NT	NT	NT	NT	NT	NT	NT	
Bryston 2B	um	49K	200	72	112	NT	ok	ok	0	+0.3	0	
Bryston 3B	um	48K	200	123	204	NT	ok	ok	0	+0.4	0	4
Carver M400	.02%	15K	200	278	489	552	ok	1uF	NT	NT	0.3%	5
Crown DC300A	um	NT	NT	190	332	356	S	2uF	NT	NT	0.1%	
DB Systems DB-6	um	50K	300	58	97	145	ok	ok	NT	NT	0	
Dunlap-Clarke 500 mod.	um	19K	300	298	499	821	ok	0.06uF	NT	NT	0.7%	7
Dunlap-Clarke 1000	.02%	20K	200	288	482	879	ok	0.01uF	NT	NT	0.3%	8
Dyna 150 kit	um	35K	155	111	160	NT	ok	ok	0	+0.3	0.04%	6
Dyna 150 (modified)	.02%	49K	225	94	126	NT	ok	ok	0	+0.2	0.1%	
Hafler DH-200 kit	m.e.	23K	340	147	219	--	ok	ok	+0.1	+0.9	0.02%	9
Hafler DH-200	um	25K	350	138	210	NT	ok	ok	0	+0.4	0.02%	
Integral Systems 200	um	32K	25	118	162	NT	ok	0.002uF	NT	NT	0.3%	
Leach II kit	.18%	22K	500	113	194	NT	ok	ok	0	0	0.03%	10
Levinson ML-2	m.e.	100K	100	52	100	191	ok	ok	0	+0.1	0	
Levinson ML-3	um	20K	525	282	484	779	ok	ok	+0.4	+1	0	
Marantz 9	.15%	200K	100	43	46	NT	ok	ok	+0.2	+1.2	0	11
Marantz 15	m.e.	100K	400	61	80	NT	S	ok	0	+0.4	1.0%	
McIntosh MC 2105	.02%	50K	155	85	36	NT	NT	1-2uf	+0.4	+1.0	0.15%	1,12
					82	38						2
NAD 3020	m.e.	38K	500	38	63	--	ok	ok	+0.1	+0.4	0	13
Nikko Alpha 3	.11%	45K	300	114	138	50	ok	0.07uF	NT	NT	0.03%	14
Phase Linear 700	um	100K	50	376	556	--	ok	0.42uF	NT	NT	0.03%	15
SAE Mk 31B	m.e.	48K	250	75	123	--	ok	ok	0	0	0.15%	16
SAE X25A	um	37K	500	318	561	896	ok	ok	+0.2	+0.3	0.03%	
Sony 3200F	um	62K	250	158	236	191	S	0.007uF	NT	NT	0.15%	17
SWTP Tiger	.03%	20K	---	89	78	41	ok	ok	+0.2	+0.6	0.04%	
Threshold 400A	NT	NT	NT	NT	NT	NT	NT	NT	NT	NT	NT	
Yamaha CR-620	.07%	42K	100	55	73	6	S	0.2uF	NT	NT	0	

um = unmeasurable (below 0.01%)
m.e. = measurement error

NT = not tested
S = sticking

NOTES

1. 8 ohm tap.
2. 4 ohm tap
3. Lots of 3rd harmonic distortion at 10v into 8 ohms.
4. 15mV DC offset, funny clip ckt.
5. Fuzz on 20 kHz waveform.
6. Blew fuse (4A) at 4 ohms; not tested at 2 ohms.
7. Tripped Relay into 0.06uf.
8. Tripped Relay into 0.01uf; 15 mV DC offset.
9. Blew Fuse at 2 ohms.
10. 90 mV DC offset.
11. 4 ohms power measured at 4 ohm tap.
12. Asymmetrical clipping. Intermittent operation into 1-2 uF.
13. Thermal breaker trips after 30 seconds full power into 4 ohms.
14. Relay trips into 2 ohms.
15. 250 mV DC offset (!)
16. 37 mV DC offset.
17. Waveform kinks and parasitic oscillation at clipping.

A 1 kHz input signal was applied, and increased until distortion was just visible in the waveform on the oscilloscope screen. Aside from one unit that showed asymmetrical clipping (i.e. reaching its power limit at different voltages for the positive and negative parts of the waveform) all performed acceptably on this test. At 4 and 2 ohms some units blew output fuses or had tripped relays, but this has little to do with their performance with musical material.

A 20 kHz signal was applied, and increased until the output was driven well past the initial point of clipping. The waveform was then examined for signs of misbehavior. People may not be aware how easily amplifiers can be driven into overload with dynamic program material. Ideally an amplifier should recover immediately once the input signal has dropped below the clipping point -- some exhibit sticking or latch-up, which corresponds to the internal circuitry taking time to recover to its normal condition, long after the original overload has passed. This means that the audible effect of the clipping is greater than it would have been otherwise. The worst case is at high frequencies, so 20 kHz was chosen for this test. None of the units performed badly; six exhibited small amounts of sticking. Two were somewhat unusual: the McIntosh had a limiting circuit which kept the amplifier from going into clipping no matter how much signal was applied. Apparently there is a circuit that reduces the input signal when clipping is sensed. When this circuit was switched out some very bizarre waveforms were observed, so it was obviously a good idea. The Bryston 3B showed a visible "cut-in" of a limiting circuit -- when clipping began the clipping level would jump to a lower level. It was a slight effect, but worth noting. One unit exhibited parasitic oscillations at clipping.

Through all these tests the oscilloscope was an invaluable tool. In a recent article entitled "'Just Detectable' Distortion", James Moir stated that the limit of audibility of distortion corresponded approximately to the limit of visual detection on a sine waveform. This is in the neighborhood of 0. 5% distortion.

The next test was for stability under reactive loads. The worst case is a capacitor since its effect is to load the output impedance of the amplifier at ultrasonic frequencies causing phase shift in the direction that brings the amplifier closer to being an oscillator on its own. An inductor (as are most loudspeakers) becomes an open circuit at high frequencies so it has no effect on stability. For this test a Heathkit capacitor box (IN-27) was used to switch capacitance values in 100 pF increments from 100 pF to 1000 pF, 1000 pF increments to 0.01 uFd, and .01 uFd increments to 0.1 uFd. Separate capacitors were then applied with values of .22, .42, 1 and 2.2 uFd. There was no resistive load and all capacitors were the low inductance type.

Both shunt resistance and series inductance would tend to improve the stability factor -- this was a "worst case" test. The output was driven to 10 volts at 1 kHz . If oscillations were observed the test was immediately terminated. Almost one third of the amplifiers were unstable under some value of capacitance. On a few amplifiers that oscillated, a higher value of capacitance was tried, and the oscillations got worse. (Although such loads are not often encountered, electrostatic loudspeakers are capacitive and some speaker cables have as much as 0.01 uFd of capacitance.) This says to me that some designers didn't do their homework.

On a select few units a Safe Operating Area test was performed. This involved driving a 200 Hz signal into a 230 uFd load and tests the ability of the amplifier to drive reactive loads; the voltage and current are 90 degrees out of phase and all power gets dumped back into the amplifier. The same effect occurs with a 2.7 mH inductive load. Early transistor amps would probably blow up on this "test". The DB Systems, Hafler, Levinson ML-2, McIntosh MC2105, and SAE X25A handled this load with aplomb.

The effect of a 2 uFd load on frequency response was checked at 7.5 and 15 kHz. Usually amplifiers have an inductor in series with the output to ensure stability (Ha!)

and the resonance of this with the load capacitance causes a high-frequency rise. This might be responsible for at least some of the audible differences between amplifiers driving electrostatic speakers. The rise ranged from 0 to 1.2 dB at 15 kHz.

The final test was for crossover notch distortion. A 20kHz, 10 volt signal was driven into 8 ohms. A twin-T passive notch filter was used to eliminate the fundamental signal and the residual waveform was observed on the oscilloscope. This is like putting a magnifying glass to the waveform; any deviations from linearity become easily visible. Most amplifiers exhibit two glitches per cycle -- a spike is generated each time the signal is transferred between the two halves of the output stage. Class A operation eliminates this switching action at the expense of high quiescent dissipation. However, the measurements show that some Class AB designs can perform equally well.

Crossover distortion tends not to show on conventional distortion measurements since the spikes are of short duration and get swamped out in an averaging measurement. However, the ear may not integrate the spikes but rather respond to the peak value. Consequently for the table the peak-to-peak value of the spike is expressed as a percentage of the peak total waveform.

It is interesting to note that one amplifier's "Hypersonic Class A" design did not keep it from having measureable crossover distortion (though it had good performance for a high powered amplifier). The Carver amp had additional notches, apparently due to its switching power supplies; however, the magnitude of the spikes was small.

Subjective tests as to the audibility of crossover distortion have shown that surprisingly large values (1 to 2%) can be present without audible effects on musical material. However as with other forms of distortion, pure tones and special waveforms allow for audibility at much lower levels. Also it is worth remembering that the signal is passing through many pieces of equipment before it becomes a sound. It is important the individual units in the chain be much better than the audibility threshold of distortion so that the cumulative effect is not audible.

In conclusion, the most distressing result was that many amplifiers were unstable under capacitive loads. In most circumstances this will not cause a problem since typical loads are inductive. However, it reminds me of a time bomb waiting to go off. To my mind a good amplifier should be limited only by its power output -- it shouldn't have to be restricted to a particular kind of load. The overload behavior of all the amplifiers was acceptable. The small amount of sticking would probably not be audible on normal program material since the mid-range would be heavily clipping before the high frequencies started to clip. With perhaps one exception all amplifiers had well controlled crossover distortion and several units performed as well as the Class A unit.

LISTENING TESTS by Marl< F. Davis

The amplifier listening test section of the BAS amplifier clinic was intended to allow participants to hear for themselves the difference in sound quality (or absence thereof) between amplifiers compared pairwise. As members are no doubt aware, the debate on the ultimate requirements for an audibly "perfect" amplifier remains open in the minds of some, with certain segments of the audio industry claiming to be able to hear all manner of differences between otherwise competently designed power amplifiers, including such subjective perceptions as "hardness", "granularity", imperfect imaging, altered ambience, lack of "inner detail", et al.

Others have found that power amplifiers are sonically indistinguishable provided that their operating levels and frequency responses are closely matched. A tolerance of 0.1 dB or better is usually considered sufficient, providing the amplifiers are not clipping or otherwise producing gross distortion(s). (Although differences on the order of 0.1 dB or better are required for critical A-B tests in order to rule out frequency response as a factor, deviations of under 1 dB are considered unimportant in normal everyday use.) This author believes that amplifiers reached a state of perceptual perfection at least 30 years ago, although the intervening period has probably seen its share of incompetently designed amplifiers that did sound inferior.

The listening tests performed at the BAS clinic were not intended to prove or disprove any particular theory, but rather to allow members to hear diverse amplifiers A-B compared in their raw, "un-tweaked" state. Although output levels were matched before each listening comparison, there was no attempt to equalize frequency responses or correct for possible phase inversion. Nor were distortion, noise level, or other specifications checked in situ (with the amp connected to the loudspeakers). Psychophysically valid experimental procedures, such as "double-blind" listening, were not followed. Sweeping conclusions based on the results which were obtained are therefore not encouraged.

Equipment. Three signal sources were used for the tests:

A) a dbx-encoded quarter-track 7.5 ips tape of a variety of well recorded classical, jazz and pop music, mostly direct cut or dbx-encoded disks, dubbed with a Shure V15-IV cartridge, BIC 940 turntable, Davis-Brinton preamp, and Sony 377 tape deck, modified for flat response to beyond 25 kHz;

B) a dbx-encoded half-track 15 ips tape of some choral music and an ungodly thunderclap, both made by Brad Meyer, and played on a Studer deck loaned by Apt Corp;

C) a variety of high quality discs played on a Linn- Sondek turntable with an EMT-250 cartridge, via a pair of Levinson mono preamps; this equipment having been supplied by Alan Goodwin's stereo shop in Boston.

The heart of the equipment setup consisted of an amplifier A-B box constructed with Al Foster's help. The low level section of the box, which was entirely passive, accepted a stereo line-level signal from one of the signal sources and fed each input channel to the hot side of a pair of 20K ohm level-setting pots, whose wipers fed the inputs of the amplifiers under test via shielded cables. There was no switching of the low level signal; both amplifiers were fed simultaneously. A common ground was used throughout the low level section.

The ouputs of the amplifiers were fed via heavy-duty speaker cables to a listener-operated switch which selected the outputs of one of the amps to be fed to a pair of KEF 105 loudspeakers. No common grounds were used anywhere in the high-level section.

Procedure. After being connected, the two amplifiers being compared were level-matched by feeding the line level inputs of the A-B box from a Hewlett-Packard sinewave oscillator set to 1 kHz, and monitoring the power amp outputs (after the listener switch) with an H-P AC voltmeter. All four 20K ohm pots were advanced to maximum, then the signal levels to the amplifier with the greater gain were reduced until output levels from the two amplifiers matched. The levels were then rechecked at a few other frequencies from 20 Hz to 20 kHz, to assess relative frequency responses of the amplifiers while loaded by the KEF loudspeakers.

The amplifiers were then A-B compared using a variety of source material, usually over a period of about 15-25 minutes. The listener (usually the owner of one of the amplifiers) was allowed to switch at will, and was given his choice of source material; sections were repeated if requested. Listeners could, of course, listen for whatever perceptual characteristics they chose -- level, frequency response, imaging, ambience, etc. The tests were done single-blind; the listener wasn't told which amp was which, but the tester knew.

At the end of the listening period, the listener announced whether he felt he heard a difference. If there was a claimed difference, a few short additional trials were run with the same amplifiers to see if the listener could reliably identify one of the pair on the basis of sound quality. Otherwise, the next amplifier or pair of amplifiers were connected and the process repeated.

Results. Time constraints allowed only ten pairs of comparisons to be run. The results, which are tabulated in Table 3, support the hypothesis that power amplifiers are sonically indistinguishable except under conditions of frequency response difference (Nikko/ Levinson, McIntosh/Apt), clipping (Nikko/Levinson, Leach/Apt,Marantz 15/Apt), and obvious misbehavior (Marantz 9 birdies). As noted earlier, the listening tests cannot be taken as proof of anything.

In general, people were surprised at how minuscule the differences were, (given that there were claimed differences). The most oft-repeated comment was, ''If I hadn't heard it (the lack of a difference) myself, I wouldn't have believed it."

TABLE 3

Amp A	Amp B	Any Difference Heard?	Comments
Yamaha CR-620	Apt 1	No	
Nikko Alpha 3	Dunlap-Clarke 1000	No	
Nikko Alpha 3	Levinson ML-3	Yes	A, B
SAE Mk 31B	Phase Linear 700	No	
Marantz 9	Phase Linear 700	Yes	C
Integral. Sys.200	Phase Linear 700	No	
Dyna 150 mod.	Phase Linear 700	No	
McIntosh MC2105	Apt 1	Yes	D
Leach II kit	Apt 1	Yes	E, A
Marantz 15	Apt 1	Yes	A

COMMENTS

A: Audible difference in thunderclap, due to clipping of amplifier A.
B: Difference heard at high frequencies. Measured response of Levinson amp exhibited gradual high-frequency rolloff, -1.5 dB at 20kHz.
C: The Marantz 9 had its original tubes (!), produced obvious "birdies" on some material as if bothered by RFI. (The test location had much RFI.)
D: Difference heard in bass range. Measured response under load showed Mac amp down 0.5 dB at 100 Hz relative to the Apt, and up 0.3 dB at 50 Hz.
E. Mono comparison using left channel only, speakers paralleled.

An additional test. The people from Goodwin's store (Mr. Goodwin is a Mark Levinson dealer) were, not surprisingly, unhappy about the smallness of the differences heard, and claimed that the reason for this was the high level section of the A-B switch box, which they suggested could have an audible effect that would mask differences in power amplifiers (N.B.: the high level section of the box consisted of some speaker wire, a switch, and some Pomona [G-R] connectors).

It is certainly possible that introducing such a box into the signal chain could have an audible effect; the extra 20 feet or so of speaker cable plus the minute contact resistance of the switch could conceivably amount to a few tenths of an ohm which could attenuate the signal by a few tenths of a dB, as compared to the same setup without the switch box. If this level difference were not compensated, it might indeed be audible. However, this slight attenuation would in no way mask amplifier differences, if they existed. Whatever extra inductance or capacitance the box added would be many orders of magnitude too small to have any effect on amplifier performance at the impedance levels (8 ohms) in question. (Ex: If the box added, say, 100 pF of capacitance, it would resonate with 8 ohms at a frequency of 0.2 gigahertz.) In any case the box was common to both amplifiers, and levels were matched.

In the spirit of the academic search for truth, the people from Goodwin's ran a special listening test in the evening to try to establish whether the A-B box was audible. Four 5-minute listening sessions were conducted with about 5 listeners, using the record playing equipment described as source "C" above, and a Phase Linear 700 amplifier. No A-B switching took place during any of the four sessions; in two of the four sessions, the switch box was present, and in the other two, a direct connection was made between power amplifier and loudspeakers (the KEFs) using 7 feet of very heavy duty speaker cable. Between each session, the listeners left the room while the folks from Goodwin's made changes. After the four sessions (which used the same section of the same record each time), the listeners were asked to grade the sessions in order of quality, best to worst. The general consensus (not unanimous) was:

(session 1, session 3, session 4, session 2)

with session 1 clearly the best, session 2 clearly the worst, and sessions 3 and 4 pretty close. This corresponded to:

(presence, absence, presence, absence)

of the switch box. This would seem to support the notion that the switch box did not have an audible effect. The Goodwin people, however, discounted sessions 1 and 2 on the basis of "mis-set levels", and claimed that the test, on the basis of the remaining sessions 3 and 4, showed that the presence of the switch box degraded the sound. All agreed that the data from this brief test was sparse.

<u>Conclusion.</u> While the results of the listening tests are not conclusive, they are consistent with the notion that frequency response and power output are the two primary factors mediating amplifier quality: in every case in which a difference was heard, there was either obvious clipping or other misbehavior, or frequency response differences much greater than 0.1 dB <u>in the frequency range indicated by the listener.</u> Differences in measured distortion and noise levels, slew rates, protection circuitry, design philosophy, et al, seemed to have no audible correlates.

Further, the presence of clipping should be considered in practical terms. Brad Meyer's thunderclap recording was the only source that caused any clipping during the tests (reproducing a thunderclap over KEF's at realistic levels takes a bit of power). If you're not into thunder in a big way, you may be able to get away with less than gargantuan amplifier power.

In short, it does not seem from these listening tests as if power amplifiers are, in any practical audible sense, limiting the state of the art in sound reproduction. The cheapest and surest way to bridge the gap between hifi and the concert hall continues to be the purchase of a ticket rather than another amplifier.