THE B.A.S. SPEAKER

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In This Issue

Just a few years ago, most audiophiles would have laughed in the face of anyone trying to sell a $3600 amplifier. Nowadays there's less laughing and more check writing. Are the people who lay out small fortunes for audio equipment getting their money's worth? We can't say, but we do suggest that you hold onto your wallet until you've read this issue, which is in many ways (though perhaps not obviously) about thrift.

Dan Shanefield and John Sprague explore the pitfalls of subjective testing and lay the foundation for a healthy skepticism. Their conclusions should give pause to even the most ardent of reviewers. Some of Sprague's and Shanefield's ideas find practical application in a review by Al Foster and Ira Leonard of the Supex SD900/E cartridge. For about $380 you can buy a Supex in full battle array—about $320 more than you would spend for a Shure V15-III. Does that $320 buy a more natural sound, or just a different sound? Al and Ira think they know.

Bob Graham, who's assembled his own speaker systems, reviews a couple of exotic drivers that have been around since the fifties—the Janszen 130 and the Ionovac. His article points the way to state-of-the-art speakers for less than you may have thought possible. (And if you'd bought them when they first became available, they'd have cost even less.)

Finally, there's more on the Holman preamp tests (some of it from Tom Holman himself) and a description by Al Foster of a quick, simple, and definitive subjective test for preamps.

BAS Membership Directory

A new BAS membership directory is available to members only. Copies may be obtained at BAS meetings or by sending a stamped, self-addressed envelope (regular business size, please) along with your request to P.O. Box 7 (Attention: Joyce Brinton).

Please remember that the directory is for BAS members only and that it contains only the names and telephone numbers of those members who indicated they were willing to be listed.

For out-of-state members, the area codes should help you identify other members living in your area in case you want to begin your own group.

Meeting Cancellation

We had to cancel last month's BAS meeting because of snow. Since we had no prior mechanism for this, about six of us tried to call each local member to notify him of the cancellation—about 250 calls. No one wants to go through that again. So . . . in the future, meeting cancellations will be announced on WBUR and WGBH and on as many other stations as possible (probably WEEI, WCRB, and WHET). So in case of bad weather, tune in to see if the meeting has been cancelled.

Overseas Record-Buying Service

In the December BAS Speaker we reported that Dr. Brian Leeming is organizing a record-buying service. We should have added that all record orders must be prepaid and all deliveries will be made at BAS meetings—no mail orders are possible.

For Sale

- Decca International tonearm, new, never used, $100. Jerry Johnson, 542 N.W. 34th St., Oklahoma City, Okla. 73118, or (403) 524-7233.
- Pioneer SE-700 electret headphones, new with warranty card; Dayton-Wright XG8's with SG8/2 walnut cabinets plus ST-300, superb condition; Revox A77, Mk. 4, high-speed, two-track, never used (purchased late September 1975); Bib Groov Kleen 42, new; "Stereo Record Guides," Ivan March, Vols. 9, 7, 6, 5, 2; Sony ST-5000FW with walnut case; new Preeners; new Parastat. Ross Robinson, 8888 Riverside Drive, Apt. 1707, Windsor, Ontario, Canada, or (519) 945-8486.
- Supex SD-900E cartridge, $70; Vestigal arm, $65; both for $125. (212) 454-3205.
- Pioneer SX-828 receiver (has facilities for two tape decks, two phono inputs, two headphone jacks, two mike jacks, FM tuner section; can accept three pairs of speakers—run one, or any two simultaneously. Recently brought up to specifications. $300. Richard Gage, 9 Gowing Road, Wilmington, Mass. 01887, or (617) 658-3527.

Wanted

- Otari MX-5050-2SH tape deck; Quad AM11 or AM3 tuner; Quad 22 preamp; Quad 11 tube amp; Russound QT-1 patching control box; Soundcraftsmen RP2212; Sheffield S-9, Vol. 1; "Stereo Record Guide," Vol. 8. Ross Robinson, 8888 Riverside Drive, Apt. 1707, Windsor, Ontario, Canada, or (519) 945-8486.

SpeakerArticles for the *Speaker*

We have been receiving more and more mail on the general topic of loudspeakers—bare drivers, substituting new drivers in existing systems, range-extending units like super-tweeters and infrawoofers—with most correspondents wanting fairly specific suggestions as to which units are good or bad, evaluations of electronic crossovers, etc. Unfortunately the BAS members who most often contribute to the Speaker appear to be less interested in this end of things than in others, although from the equipment information on application forms, many members have what might be called "tailored" speaker systems.

With this in mind, we would like to request descriptions of and information on the pros and cons of the varied "roll-your-own" speaker systems owned by the membership. We are especially interested in pitfalls or outstanding successes, but we won't turn our back on a system capable of simple satisfaction—many of us would settle for that gladly. Contributions can be brief, but they must be thorough; for example, at this end of the hi-fi chain, a description of the listening room often can be as important as that of the system itself.

Having asked you to contribute, we begin fulfilling our half of the bargain by presenting this month what we feel is an excellent review of a fine pair of mid- and high-frequency speakers—the Janszen Model 130 electrostatic and the Ionovac—by Bob Graham.

— Jim Brinton

Spring Reverb Revisited . . . Visited . . . Visited

[Dan Shanefield's spring reverb note in the November 1975 issue was edited to improve the English, but some technical changes crept in as a result. To avoid confusion, Dan presents the following clarifications.—Ed.]

The edited article said that the Lafayette $34.95 reverb device is a "stereo unit," but a reader who buys it might be disappointed to find that there is only one, monophonic, output circuit. The Lafayette catalog describes it as a "two-channel reverb unit." Two channels in, but only one out . . . cheap, cheap! But it is effective anyhow.

The Madsen tube that I used for non-reverberant delay was similar to the type described by Duane Cooper in <u>Audio</u> (April and May 1971), except that the walls were made of an unusual material: vinyl plastic loaded with lead metal powder (Coustifab CC-488C-UF18A, obtained from The Ferro Corp., Norwalk, Conn.). Otherwise there might have been considerable reveration. In fact, Cooper-Madsen chambers often are used <u>for</u> reverberation. By the way, I also tried other "dry" systems, like tape recorders; the spring sounded better.

For classical music, some recording studios use "plate" reverbs (which are a bit like Chinese gongs) or air chambers (like Cooper-Madsen, etc.) more often than spring reverbs. However, fairly expensive professional spring units such as the AKG Model BX-20E are used to some degree, now that their sounds have been improved (for a review of various methods, see <u>db</u>, Nov. 1975, p. 34).

— Dan Shanefield (New Jersey)

$/dB at 4 Ohms

Owners of four-ohm loudspeakers who want to use the Mashey cost-effectiveness analysis should check the low-impedance output capabilities of the amplifiers in which they're interested. Four-ohm loads draw more power from most transistor amps than do eight-ohm loads, but the proportional increase varies from amp to amp. At least one that I know of, the Harman/Kardon Citation 12, puts out slightly <u>less</u> power into four ohms than it does into eight. Obviously, its ranking in Mashey's list would take a tumble if the table were redone for four-ohm loads.

— Michael Riggs (Massachusetts)

The Continuing Saga of *Sound Advice*

What once was the staff of <u>Sound Advice</u> is now publishing <u>International Audio Review</u>. <u>Sound Advice</u> has not folded, however, and plans to continue publication, presumably with a new troop of reviewers. Apparently the publishers of <u>SA</u> kept the money and the subscription list. Those who want to receive <u>IAR</u> should send $10 for four issues ($12 for first-class mailing, $13.50 foreign, and a $2 discount for charter subscribers to <u>Sound Advice</u>) to International Audio Review, 2449 Dwight Way, Berkeley, California 94704.

— Michael Riggs

Preamplifier Testing

With the advent of the newly opened discussions on phonograph preamplifier performance has come a useful exchange of information among designers. New facets of preamplifier design are being examined, and old ones are being reaffirmed. The purpose of the paper which I wrote for the Audio Engineering Society and reported to the BAS was to open new ground on the topic—not to run down conventional tests, but to propose new areas of testing which could lead to better sounding designs. Some have responded as if the set of proposed tests were meant to be definitive: this was not my intent. The tests were proposed as <u>additions</u> to those normally available.

The ranking of the various kinds of tests according to subjective importance barely has been mentioned. The area is subject to a range of difficulties, not the least of which is variation in individual perception. Another difficulty is to hold constant all variables except the one under test—frequently an impossible challenge. We believe that the new tests are important because they do rank designs in the same order as subjective tests. But older, more standardized tests also should be scrutinized for their subjective importance.

What is needed for further development of the state of the art is a comprehensive catalog of tests for phonograph preamplifiers with standardized test procedures. Without going into details of test conditions, etc., the following tests are proposed for examining all areas of performance currently known to be important. Reader comments are solicited.

Frequency-Response Tests

A. Audible Range Voltage Source Response: Frequency response conformance to the RIAA curve between 20 Hz and 20 kHz in ±dB.*
B. Infrasonic Response: Frequency response with respect to the RIAA curve between 1 Hz and 20 Hz. Include measurements as necessitated by "rumble" filters which may be switchable.*
C. Ultrasonic Response: Frequency response with respect to the RIAA curve from 20 kHz to measurement limits. Include response of any AM, FM, CB, or TV filters that may be present.*
D. Cartridge Impedance-Input Impedance Interaction: As previously detailed in <u>The BAS Speaker</u> in ±dB from 20 Hz to 20 kHz.*

H. Noise Tests

A. "Flicker," "Shot," and "1/F" Noise: These forms of noise, manifested as clicks, pops, and thumps, become more important in equalized preamplifiers due to use of heavy bass boost for compensation. The best test would involve simultaneous use of an oscilloscope and a high-quality loudspeaker to look and listen for this kind of noise.
B. Unweighted Noise in the Audible Band: Using a shorted input, a 20-Hz to 20-kHz bandpass filter, t and a true rms meter, measure the output noise and reference in dB below a 10-mV rms, 1-kHz input voltage. Mainly a test for hum.
C. Weighted Noise Test: Using an ASA "A" weighting filter and a level correction factor for the filter as needed, and a true rms meter, measure the output noise and reference in dB below a 10-mV rms, 1-kHz input voltage. Measure with a shorted input and with a cartridge and give both measurements.

III. Distortion Tests

A. Total Harmonic Distortion: Give the THD vs. output level at low, medium, and high audible frequencies. Give the THD vs. frequency at various output levels. Describe harmonic content.
B. SMPTE IM Distortion: Give the IMD vs. level with 60-Hz and 7000-Hz input tones mixed 4:1.
C. CCITT IM Distortion: Use two high-frequency tones separated by a small amount and measure the resultant low-frequency difference tone. This test becomes especially important in equalized amplifiers since the small distortion likely to be present is

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* Assume linear operation.
t Rectangular 20-Hz to 20-kHz bandwidth.

effectively multiplied by the equalization (by nearly 100 times in the worst case). Also since the information recorded on the disc is heavily pre-emphasized, distortion of this form is more likely to be important in RIAA equalized amplifiers.

D. The "Holman" test for symmetry and asymmetrical performance as previously detailed.

E. TIM Distortion: This may be most easily instrumented and measured by means of a slew rate test, which is easily made at the same time as the "Holman" test for symmetry.

F. Sine Wave Input Overload: In mV rms at 1 kHz. The input overload level should vary with frequency with the RIAA equalization—any deviation should be noted.

G. Recovery from Overload: Time needed for the amplifier to come out of saturation from a specified overload condition.

IV. RFI, EMI Susceptibility

A. Radio Frequency Interference Susceptibility: Topic is covered above under frequency response, ultrasonic response for RFI coming in through the inputs. A test needs to be developed for RFI for inputs from RF-noisy sources.

B. Electromagnetic Interference Susceptibility: A test needs to developed for susceptibility to electromagnetic fields.

V. Environmental

A. Specify the range of environmental conditions over which the unit should meet the specifications above.

— Tomlinson Holman (Massachusetts)

Questioning the Holman Test

I fear that some of the figures for the cartridge interaction tests in Table 1 on page 2 of the Foster/Davis report (November 1975 Speaker) may be in error. For example, the Quad 33 is quoted as having a 1 to 2 dB rise above 10 kHz. But was the fact that this preamp has a phono input impedance of 68,000 ohms, instead of the usual 47,000, taken into account? Unless the FET input resistor in Fig. 1c on page 3 was changed to 68,000 ohms (or the Quad input resistor to 47,000) to achieve a proper match, the high-frequency rise found by the testers is almost certainly the result of this change in the loading of the dummy cartridge, with or without the FET buffer in the circuit. Quad quote for their preamp a 68,000-ohm input impedance, purely resistive, within 5 degrees, up to 20 kHz. I therefore doubt very much that the measured rise is because of cartridge-preamplifier interaction.

I am also somewhat wary about the significance of the Holman square-wave tests, although I cannot comment on their possible correlation with phono preamplifier sound, which may indeed be valid for reasons that are still obscure. Specifically, referring to Fig. 4 on page 5 of the report, no disc cutter has a risetime of 6 µsec, let alone 1 µsec. Hence, no such transient signals will be present on any disc and so will not be presented to any phono cartridge (even assuming that any phono cartridge had such a fast risetime, which none has). Nevertheless, differences in phono preamps are audible when playing discs. If the reason lies in the behavior of the preamp in response to such transient signals, the explanation is far from clear.

If, as Holman suggests, the audible effects may be due to slew-rate limiting, then surely the actual level of the transient peak fed to the preamp will have a very marked effect on its performance in the test. This conclusion seems to be borne out by the performance of the AR receiver (Table 2, page 7) on the square-wave test. Halving the input sensitivity has an enormous effect on its behavior. In the tests, however, a constant input signal level is maintained, irrespective of the sensitivity or overload margins of individual phono preamps. And, as mentioned in the report, the amount of even-harmonic distortion did seem to depend only on the peak input signal level. But, as is also mentioned in the article, no cartridge used in a high-quality system will produce 600-mV peaks from discs.

This test seems dependent upon driving the phono preamp stage into clipping, probably because of slew-rate limiting resulting from the very short risetime of the applied signal. Accepting that there may indeed be a correlation between a preamp's performance under these conditions and its sound on (what, in the nature of things, must be much less severe conditions when playing) discs, what would be interesting to know is the manner in which this even-harmonic distortion varies with peak input signal level and how this relates to the preamp's sine-wave overload performance (if, indeed, such a relationship exists). Referring again to the Quad 33 (with which I am familiar), its disc adapter board offers two input sensitivity settings: 2 mV (overload at 40 mV, 1 kHz) and 5.6 mV (overload at 120 mV, 1 kHz). Which was used for the distortion test results quoted in Table 2 on page 7?

And what, by the way, happened to reference 6 on page 23 of the Holman report? It is referred to on line 9 of page 22.

To end this somewhat lengthy letter, let me congratulate you on the pioneering nature of your efforts. Your journal is always stimulating and excellently produced and seems to be showing the way to many of our less knowledgeable and meticulous manufacturers. My own experience, comparing the Quad 33 with the Dynaco PAT-5, has been that the latter shows an improvement in handling high-level, high-frequency massed voice or string passages on discs—i.e., its sound is more open and less congested. In fact, much of what I had put down to tracing distortion appears now to have been preamp distortion of some kind. The Quad 33 is, however, still relatively good in this respect.

— S. P. Lipshitz (Ontario)

I talked to Mark Davis about Mr. Lipshitz's letter. Mark reports that he doesn't know which sensitivity setting was used on the Quad preamp they tested. He concedes that the high-frequency rise they measured could have resulted from the Quad's nonstandard input resistance but points out that it shouldn t be 68,000 ohms to begin with: wrong is wrong, no matter how you cut it. At the moment, no one knows why the Holman square-wave test works. The test conditions are admittedly unrealistic. The results do, however (with tube and bipolar transistor preamps), correlate roughly with what we hear. Mark suggests that further research needs to be done.

Our thanks to Mr. Lipshitz for his kind words.

— Michael Riggs

How To Identify Edgy and Good-Sounding Preamps

In the November 1975 issue of the Speaker, I briefly described how to identify a poor or edgy-sounding preamplifier. I also pointed out that my subjective ratings of preamplifiers correlate well (except for the FET designs) with the results of the Holman test. After years of listening to a variety of preamps, I have come to rely on three easily obtainable records that instantly reveal the sonic character of a preamp. The listening sessions take just minutes, and there is no urgent need to employ a switching box, as the differences are obvious. Those for whom I've demonstrated the test agree that differences are easily discernible, even though it has often required as much as five minutes to complete the necessary rewiring. A preamp either passes or it fails: there is no gray area.

Test results are also relatively independent of the quality of the associated equipment. I have obtained identical results on a pair of $50 ADC 404 six-inch bookshelf speakers. To ensure that the test is not amplifier-dependent, I've used a Phase Linear 700, a Dynaco 400, and a Marantz 500. Switching cartridges also seems not to have any effect; the Shure V15-III, the Supex SD900/E, the ADC-26, and the Audio-Technica AT-11 all yield the same results. Similarly, differences in RIAA equalization appear to have little effect, although it is considered good practice to examine the response. Most quality preamps are within +0.5 dB of the standard.

To successfully demonstrate the test, one must obtain two preamplifiers, one rated good and the other edgy-sounding according to Table 2, "Square Wave Test Results," in the November Speaker. One can't complete the test without alternating between the good and the bad preamp, as a reference is required. The idea is to detect sonic differences; listening to only one preamp won't show up the aberrations for which one should be learning to listen.

All the non-FET preamps, including the Dynaco PAT-5, that precede the Marantz 7C in Table 2 have failed my subjective test. The rating order is not, however, absolute. For example, the Pioneer SC-700 sounds smoother and less edgy than either of the two new PAT-5's I've sampled. But, in general, the rankings hold up fairly well, the PAT-5 being better than the PAT-4, etc. On the other hand, all the preamps following the Marantz 7C in the list are slightly smoother sounding than the 7C, as are the FET designs, but the differences are minute. Subjectively, three well-defined categories can be established: (1) preamps failing the test, edgy, (2) preamps passing the test, good, and (3) excellent. All of the preamplifiers in the third category sound very much alike. The differences that do exist are, I believe, largely a function of their RIAA compensations, which are never quite identical.

To separate good preamps from excellent ones, prolonged listening sessions with a variety of records, good switching facilities, volume compensation devices, and a high-quality playback system are necessary. Whether a preamplifier passes the three-record test is obvious, but most people will not be able to distinguish preamps in the excellent category, simply because they lack the time or the necessary testing apparatus. The differences do not correlate with price or with RIAA equalization accuracy, but they are there.

The records I use for my test are as follows (others that bring out the same differences may be substituted):

• Janis Ian, Between the Lines, on Columbia

The aberration to listen for on this album is accentuated sibilants. The particular cut I use is the first three minutes of "At Seventeen." Listen to the first time she pronounces the word "seventeen." On an edgy preamp, the S's and T's sound more like white noise than like air passing between the teeth. In addition, poor sounding preamps produce twice as many sibilants as the good ones.

• Robbinsdale Spring Band Concert, a Fulton Ark recording, $5.95 from Roman's Audio Classics, 8012 Cedar Avenue So., Bloomington, Minnesota 55420

On this record, it's the quality of the very prominent cymbal attack for which one should listen. I use the opening passages of "Wall Street Rag," in which the cymbal is recorded very close up. On an edgy preamplifier, the sound of the attack transient, when the drummer strikes the cymbal, is a loud click followed by a poorly detailed shimmer. The sound is similar to amplifier clipping and is obvious. A good preamplifier greatly diminishes the click; the cymbal loses the heavy coloration associated with a poor preamp and gains detail, body, and a shimmer that instantly identify it as a cymbal.

• Handel's Water Music, Raymond Leppard, Philips 6500 047

The first few minutes of the "Suite in F" is the most definitive test. This selection contains a large number of violins, which do not blend with the orchestra when heard over a poor-sounding preamp. They sound edgy, harsh, and unnatural and lack ambience or "air," which results in a flat, strident sound. The violins lack homogeneity and warmth and are overly bright and zippy.

In conclusion, we can say that compared to good preamplifiers, edgy preamps sound brighter and less detailed. They lack "air," transparency, and naturalness. Good preamps are nonfatiguing and much more pleasant sounding. Because of their ability to elicit more hall ambience, they sound more three dimensional. To appreciate these differences, you should try the test yourself.

— Alvin Foster (Massachusetts)

The Rectilinear III Revisited

Recently I wrote Rectilinear about the midrange driver of their model III loudspeaker, and they answered that they have a new type midrange available for about $20 postpaid, instructions included. They claim that the improvement is very definite. I am sending a check to them for two units, to see what happens.

The discarded units will be used as full-range extension speakers, but I lack knowledge of how to build an enclosure for them. Perhaps another reader has had experience with this driver?

— Carlos E. Bauza (Puerto Rico)

KEF 104 Loudspeakers

I won't stake my sacred honor on it, but the KEF 104 may be, overall, the best speaker I've ever heard. I'm reticent mainly because I haven't heard it in familiar surroundings. A friend bought a pair recently, and, for me, it was love at first listen. If you have $600 and a yen for new speakers, you should check these out at your local emporium (in Boston, Suffolk Audio).

— Michael Riggs

More on Headphone Amps

[Editor's Note: John Sprague comments below on material presented by Peter Mitchell in his article, "Making a Compact Headphone Amplifier (or Two)," which appeared in the June 1975 issue of the BAS Speaker. Rather than recapping that prior material, we refer you to Volume 3, Number 9. For new members without access to Volume 3, a complete set of that volume can be obtained by sending $12 to P.O. Box 7.]

Sight unseen, it certainly seems as if the Radio Shack 0- to 18-volt, 1-amp variable power supply (catalog number 277-112) with current limiting and dual tracking capabilities (possibly options) would be most suitable for the 380 IC headphone amp. The project board is not in their stores here yet.

I agree that the 741 is the better amp, but my phones are the 18-ohm Superex ST-F. My amps have no provision for phones, and I don't want to go the route of resistors on the speaker outputs because of lowered damping. I also don't like the possibility of blowing out either the phones or my ears. SWTP's class-A amp looks good but is reported by The Audio Amateur to run hot and is expensive. Although reported specifications are not quite comparable, Peter Mitchell's 380 looks as good for roughly half the price.

As for the uncertain impedance of the Koss Pro 4AA, it should be possible to measure this just as it's done for loudspeakers. How to Build Speaker Enclosures by Don Davis and Alexis Badmaleff gives the method on page 136. The 300-μF capacitor Mitchell used is certainly intended for 25 ohms, not 250. Some have suggested using an emitter follower transistor with the 741 to enable it to drive low-impedance phones.

Besides a volume (dual input level) control, a balance control and a variable blend control (continuous, as on the Dyna DSC-1, or stepped, as on their later equipment) are useful additions. For the Superex ST-F (discontinued, but available for about $15), which, according to Consumers Union, rolls off in the bass, one could add loudness compensation to correct the response at all levels.

The Radio Shack power supply's current limiting feature should offer additional protection, as the 50-mA dc peak Peter suggests for a 380 IC can be readily measured and adjusted for using most volt-ohmmeters. It should be more than adequate for four channels, if one has low-impedance quad phones (as most listed in Audio's latest product directory are).

The construction article in the June 1975 Speaker contains a diagram that may confuse some readers. It shows the board layout from the component side, but the solder pads, which are on the other side, are also shown. For physical strength, holes should be drilled through the pads (the board comes with holes for the IC or its socket). One-sixteenth-inch holes are a tight fit for two no. 20 wires, such as most parts are likely to have. Some capacitors, however, use smaller wires. Two holes or one larger one are needed for pads with three or four connections. Pads 3, 4, 5, 7, 10, 11, and 12 are all grounded, and it is easier to bridge across the socket pin solder joints between pads 3, 4, and 5 and 10, 11, and 12 than to run jumpers between the pads, if one is careful to bridge at the right places.

Using an enclosed socket for the headphone jack (Radio Shack 274-382 instead of 274-312, open contact) should help prevent accumulation of dirt and pollution-caused corrosion inside the box. The terminals of this unit are not specified, and I haven't checked mine out yet.

If the power supply is mounted in the same box, putting the transformer on the box rather than on the circuit board is desirable to prevent breakage. If hum is a problem, a partition to shield the transformer should help, or the transformer can be mounted outside, if it won't be subject to mechanical abuse, with holes (use grommets in them) for its leads. Radio Shack has several sizes of 24-volt transformers. Even if the board doesn't call for the largest, moving up will provide a more conservative rating and, presumably, let it run cooler (another good reason to put it outside).

Even if you pull plugs, as I do, for lightning protection, you may want a switch on one or both sides of the power line, a fuse on the power line, and perhaps even a pilot light. A spike suppressor can be installed at the switch. Don't forget fuse holders, to make replacement easier for both power line and output channels. Fuses for 1/2 watt (ST-F's rated capacity) at 18 ohms or 1/6 amp should be 1/32 AGX fast-acting instrument fuses, which allow up to about eleven times as much transient peak current (see Popular Electronics, July 1972, pp. 43-46), or about 0.35 amp or 2.2 watts. The ST-F's rating will not be exceeded for more than about 0.3 second if these fuses are employed. A Superex engineer told me, without checking, that he thought 10 to 15 milliwatts to be about the normal level for the ST-F. Similar calculations could be made for whatever phones anyone is interested in using.

— John F. Sprague (New Jersey)

Disc Exporters

Any reader wanting a capsule comment on the quality, reliability, etc., of British record exporters should send a self-addressed stamped envelope to: Ross Robinson, 8888 Riverside Drive East, Apt. 1707, Windsor, Ontario, N8S 1H2 Canada (U.S. postage rates apply). In my opinion, some of the heaviest advertisers in the Gramophone are among the most dubious operators. One operator of the opposite type is Maildisc (17 Red Lion Square, London W.C.1),who remains an incredibly honest, reliable, obliging LP exporter. They actually looked after a private hi-fi purchase transaction for me after I left London late last summer. There was no profit in it for them either.

— Ross Robinson (Ontario)

The Shanefield Report on Cable Capacitance

The tonearm wiring capacitance list in the June Speaker didn't include the total of lead and internal wiring capacitances for the Thorens and SME arms because the manufacturers failed to provide complete data. To measure these totals, I removed the timing capacitor from an audio oscillator and used in its place the lead and internal wiring of the arm under test (without cartridge). The oscillator was adjusted to about 6 kHz using a frequency meter. I then removed the arm and its associated cables and substituted capacitors of known values until the same frequency resulted. Values arrived at by this measuring technique appear in the table below.

	Capacitance, pF	
	Measured	Calculated
Thorens TP-16 arm (in TD-160C turntable)	300	
SME 3009-II Improved arm	150	
Radio Shack 3-foot patch cord (catalog number 42-2309)	220	200
Nakamichi 3-foot patch cord	100	
Radio Shack 6-inch Y-connector (catalog number 42-2436)	50	
RG-59A/U TV coaxial cable, 3-foot length	60	63
Typical component internal wiring from tape monitor input jack to selector switch	800 to 2,300	

A check of the literature on the Thorens TP-16 arm showed that Stereo Review, in its April 1974 review of the TD-160C (p. 30) reports a capacitance of 260 pF for "the integral signal cables." Their review of the Orotfon M15E cartridge in the January 1973 issue includes the statement that a TP-16 arm in a TD-125 Mk. II "had a total capacitance, including cables supplied, of 360 pF." And in the February 1974 Audio the TP-16 is said to have a "lead capacitance" of 270 pF.

The oscillator method was also used to measure the capacitances of various patch cords. I bought the TV coax from Lafayette and attached phono plugs to make low loss cords. These are not, however, as flexible as CD-4 turntable cables of similar capacitance, such as Lafayette catalog number 21R68292 (p. 29 of catalog 760).

In a couple of cases, I also calculated the cable capacitances with the formulas for cylindrical capacitors in the fourth edition of ITT's Reference Data for Radio Engineers (available through Sams Books), p. 134. I measured the wire diameters with a micrometer and, as a double check, calculated them, using the characteristic impedance formula for transmission lines (p. 588 of the ITT book) and the standard value of 75 ohms for the TV cable (p. 608). The dielectric constant of pure polyethylene is 2.3 (p. 66), but a plasticizer would increase this value.

This oscillator timing method works only with essentially pure capacitances. Therefore, the effective input capacitances of preamps are not measurable by this means, as there is usually a 47-kohm parallel resistance when the selector is on phono and a shorted line for other selector positions. I was, however, able to examine tape-monitor inputs with switches in their "out" positions and with compensation for cable capacitance. The table shows the input capacitance range for the preamp sections of the Lafayette LA-975, the Dynaco PAS-3x, and the McIntosh 1900 as well as for the BSR Metrotec FEW-1, Soundcraftsmen 20-12, and Bose 901 equalizers. These high input capacitances are surprising and somewhat disturbing. A look at the guts of some typical preamps indicates that phono input wiring is probably just about as capacitive as the tape monitor wiring I actually tested.

Now, let's suppose preamp designers compensate for it to achieve good RIAA equalization, using a few well-known cartridges to construct their test curves. Cartridge reviews show that some pickups are unusually sensitive to load capacitance and that some others are extraordinarily insensitive. Extreme cartridges will respond differently from the usual ones, thus giving some distorted RIAA curves. Is this one reason why there is such disagreement between otherwise careful audiophiles when cross-auditioning cartridges and preamps?

I suggest (respectfully, of course) that preamp manufacturers get together on input capacitance, as they already have with the 47-kohm standard input resistance. Capacitance may turn out to be more important than resistance. How does 1500 pF sound as a standard?

— Dan Shanefield

Chrome Cassette Tapes: Are They Really All the Same?

[Editor's Note: Member David Satz is not only one of the Society's most avid and professional tape recordists (he tapes the weekly Bach cantatas and many other Boston musical events using Dolby A, superb microphones, and the unbelievable battery-powered Nagra recorder), but he is also a fine musician and, more to the point, fluent in the German language. He subscribes to German hi-fi magazines, and has offered to summarize articles from them whenever there is enough interest. Below is a summary of a review of chrome cassette tapes appearing in Fono Forum, September 1975.

It should be noted that Fono Forum, at least in some cases, tested more than one sample of a given manufacturer's product, unlike Julian Hirsch and other American reviewers, who have generally been content to check only one sample. While three or four samples still probably don't constitute a statistically significant sample, it is a step forward. I recently tested 20 BASF C-90's, presumably from the same production run, and found the response at 10 kHz varied by up to 2 dB at different spots on a given cassette, especially in the left channel of course. The variation over the 20 samples tested was about 4 dB. These results tend to confirm the tests conducted by Fono Forum. But the question remains: What is a significant variation? I've never really been bothered or even aware of that 2 dB variation along and across the tapes in listening to music recorded on them.— Bob Borden]

The selection of appropriate cassette tape is important because, unlike the situation with open-reel tape recorders operating at 7½ or 15 ips, the possible differences in cassette sound quality are immediately audible, owing mainly to the slower speed, narrower track width, and thinner oxide coatings of cassettes. The mechanical characteristics of the cassette package also affect the sound quality: the pressure pad,which should be centered and perpendicular to the tape (many were found crooked, too high, or too low), and the guide rollers. These both can cause the tape to run crooked, which disturbs the output level from the tape, primarily in the outer (left) channel, or can produce different levels depending on which direction the tape is moving. These effects depend on what recorder is being used because of differing head geometry, etc. [The left channel also is more vulnerable to physical damage due to defective cassette mechanisms because it is at the edge of the tape.—Ed.] Noise during fast winding is no index of ease of tape movement.

Manufacturers are constantly trying to improve the mechanics of their tapes and thus usually nothing is left alone for more than half a year's production. So no two cassettes are truly comparable even if they are the "same" model from a given manufacturer.

Chrome cassettes have better high-frequency characteristics than ferric oxide cassettes, but their sensitivity at lower and middle frequencies is lower and the distortion is higher. It is virtually impossible to meaningfully compare the two types of tape since performance is so recorder-dependent.

Few manufacturers manage to make C-90's with the same quality as their own C-60's. In general, the C-90's, with their more limited mechanical stability, have more distortion and/or lower sensitivity at lower and middle frequencies, and modulation levels must be more carefully set for them; but at high frequencies, they mostly do better, not only relative to their low- and mid-frequency performance but also on an absolute basis. As a rule, C-90's sound somewhat brighter than the C-60's of the same manufacturer. These tendencies are even more strongly valid for C-120's, which should be used only with the greatest caution—and especially chrome C-120's are to advised against: several manufacturers don't even offer them and with one exception they are not considered in this report. If such a 2 x 60-minute playing time is desired, a good ferric cassette is to be recommended, since with most recorders the sensitivity and distortion figures in the bass region are significantly better anyway.

All cassettes were measured on a Nakamichi 1000, then on a Telefunken C-3300, and a Uher CG-360. All three machines were rebiased prior to testing. Frequency response was determined at 333 Hz and 10 kHz relative to the DIN reference blank tape. (This DIN tape is made to very close tolerances, is very expensive, and is primarily for laboratory use.) Results are shown in the following tables.

Frequency Response for C-60's Relative to DIN Reference Tape

	10 kHz	333 Hz
Advent	+1.5 dB	+0.5 dB
Agfa	-1	-0.2
BASF	-1.5	-0.8
Maxell	+ 1.2	0
Nakamichi	0	-0.5
Philips	-1	+0.8
Sony	+ 1.8	0
TDK	0	0

Frequency Response for C-90's Relative to DIN Reference Tape

	10 kHz	333 Hz
Agfa	-1.4 dB	-0.8 dB
BASF (1)	+0.5	-1.5
BASF (2)	-1	-1.5
BASF C-120	-2	-1.5
Philips	-1.2	-0.8
Scotch	-0.5	0
Sony (1)	+2	0
Sony (2)	+0.5	-0.4
TDK	+0.5	+0.5

Overall, the TDK (especially C-90) and Sony cassettes set the mark for the high-frequency region, and give above average results for the midrange and lower frequencies. Other above-average CrO_2 cassettes are rare on the German market, either lacking official representation (Advent, Memorex) or else are overshadowed by the same firm's ferric cassettes (Maxell).

Among moderately priced cassettes, Philips can be recommended, despite a tendency to "darker" sound quality. Agfa and BASF, all in all, didn't achieve the quality of the DIN reference tape. Agfa is generally somewhat more sensitive; BASF can be modulated more heavily and has less noise (C-60 only, four samples of each checked). Scotch chrome cassettes must be content with one of the last places, on both electrical and mechanical grounds, completely at odds with the reputation of this company and with the price usually charged for this tape.

[Capsule summaries of the chrome cassettes available in the U.S. are given below. Dynamic range and noise figures are relative to the DIN reference tape. It should be noted that many of the tapes tested were of European manufacture and are not available in the U.S. Also, certain tapes that were tested and are available in the U.S. are not readily available in Europe (e.g., Advent, Memorex, Nakamichi) and so only one sample of these was tested.—Ed.]

<u>Advent</u>. The sample tested was a demonstration cassette that came with an Advent 201, dating from 1972. This was the only sample available. The DuPont "Crolyn" is of standard-setting quality.

Overall dynamic range: +2.5 (C-60)   Noise level: +0.5 dB

<u>Agfa</u>. Among the domestic [German] offerings, the Agfa deviates least from the reference tape. Both C-60's and C-90's exhibited decent quality, with little variation among samples and with dropouts held within bounds. Recent samples have been fitted with an SM mechanism; older versions had been criticized for very loud noise during winding.

Overall dynamic range: -1.5 (C-60)   Noise: +0.5 dB
-2.5 (C-90)

BASF. The sensitivity is generally unsatisfactory, but distortion and noise are in order. The C-60's tested, including one from 1972, varied only slightly. One C-90 had excellent response in the high-frequency region, but a second C-90, like one C-120, had only average high-frequency sensitivity. Dropout occurred with average frequency. The high-frequency output level varied as much as 2 dB with some samples from measurement to measurement and depending on the direction of the tape. The cassettes were equipped with the SM mechanism.

Overall dynamic range: -2 (C-60, C-90 sample 1) Noise: -0.5 dB
-3.5 (C-90 sample 2)
-6.5 (C-120)

Maxell. [As in the U.S.] you can't get Maxell chrome in stores. Stores seem to favor UD and UD-XL, which is a shame because the Maxell chrome is clearly one of the really good chrome cassettes. Its overall characteristics can be convincing with a mild high-frequency emphasis. It has splendid constancy of output, no dropouts of any kind, and is therefore well suited for measurement applications.

Overall dynamic range: +1.5 (C-60) Noise level: -0.5 dB

Memorex. Again the only sample available was a demo cassette supplied with a Teac A-350 and dated from 1972. This cassette was top grade. Because of its high-frequency recording level, this cassette was our "champion" for years, although the Sony cassette here narrowly outdoes it.

Overall dynamic range: +2 (C-60) Noise level: +0.5 dB

Nakamichi. The sample tested came with a Nakamichi 1000 recorder. The results weren't bad (especially in the upper frequencies) but also weren't overwhelming, since the distortion was somewhat more than usual. Not what one would expect from the name Nakamichi, but maybe this was only a bad sample.

Overall dynamic range: -0.5 (C-60) Noise level: 0 dB

Philips. Since Philips licenses the whole compact-cassette system, one would expect them to offer cassettes that draw the optimum from the system. These expectations were not disappointed in respect to sensitivity and distortion in the bass region—here the Philips C-60 (older version, silver packaging) gives absolutely first-class results, which were otherwise achieved only by the Advent and TDK C-90. In the high end, its characteristics aren't significantly better than, for example, Agfa or BASF, so that there is a tendency toward full but slightly dark sound. Another C-60 sample was as good in the bass and about 2 dB better in the high end, putting it in the absolute top class alongside the Advent, but current production samples (black packaging) didn't entirely fulfill the expectations raised. The C-90 sample was well above average but displayed output variations, especially in the left channel.

Recently Philips has begun fitting their cassettes with "floating foil security," referring to a second foil between the "glide foil" paper and the cassette housing. This foil provides a kind of spring action which is evenly distributed by the glide foil onto the tape so that under all conditions good tape guidance is attainable.

Overall dynamic range: 0 (C-60) Noise level: +0.5 dB
-1.5 (C-90)

Scotch. These cassettes are welded and therefore practically irretrievable if the tape jams. Another fault is the pressure pad, which, instead of the usual spring mounting, is simply glued to a piece of foam. Two samples of C-60's (an older series labeled "Made in USA" and a new series labeled "Made in U.K.") were very similar and showed, considering their price, too limited a high-frequency sensitivity. In the best case, a new series C-90 was very good (exceeded only by the Sony C-90 and TDK C-90), but various measurement showed significant-to-great variations in output level, even in the right channel, possibly as a result of the pressure pad. This problem was more pronounced in the Telefunken recorder than in the Nakamichi and Uher recorders. The new C-60 was constant in output level; the older C-60 showed deviations, but less than the C-90.

Overall dynamic range:	-2 (C-60)	Noise level:	0 dB (C-60)
	-0.5 (C-90, best case)		+0.5 dB (C-90)

Sony. This is the only manufacturer, to our tester's knowledge, that prints a serial or batch number on its cassettes. Two samples with the same number were completely identical and displayed excellent sensitivity and maximum recording level at high frequencies. A C-90 (different batch) was scarcely worse in the bass than the C-60 and in the highs was a trace better and therefore the most brilliant chrome cassette overall. With that, one buys a slightly elevated noise level, which can be taken care of with Dolby.

Overall dynamic range:	+2 (C-60)	Noise level:	+1 dB (C-60)
	+2.5 (C-90 sample 1)		+1.5 dB (C-90 sample 1)
	+0.5 (C-90 sample 2)		+0.5 dB (C-90 sample 2)

TDK. Two samples of TDK C-60's exactly matched the reference tape, but one sample showed periodic output variations of about 2 dB over the whole length and breadth of the tape. The C-90 was not only better in the highs than the TDK C-60 but was better in the bass than any other C-90 in the tests. The nearly unbelieveable results were verified by a second sample, probably from another production series. The C-90 did tend to mildly varying high-frequency output levels in the left channel.

Overall dynamic range:	0 (C-60)	Noise level:	0 dB (C-60)
	+1.5 (C-90)		+0.5 dB (C-90)

— Translated by David Satz

Reeling Them In

Frustrated? Angry? Generally tense when trying to start the tape leader winding properly on the hub of a reel-to-reel machine? Try Doctor Dan's simple remedy—it's almost guarantted. Fold the first quarter inch of the tape over and pinch it until it is creased. Stick this "hook" into the slot in the hub. It will hold just tightly enough to start winding smoothly but not so tightly as to really yank the tape when it is pulled out at the end of rewind. — Dan Shanefield

An Amplifier for Quad ESL's

Audio T (190 West End Lane, London) is said to have conducted many tests trying to find the "best" amp for Quad ESL's. Their results indicate that the Sugden P.51 amp/C.51 preamp do the best job—better than the Quad 33/303. Audio T—John Bartlett, proprietor—appears to be a reliable source of equipment from and within the United Kingdom, and has a "no-spiff" retailing approach. —Ross Robinson

Feedback Damper

Try Micro Seiki MSB-1 Microsorbers as an at least partial cure for acoustic feedback at your turntable. A package of four sells in the United Kingdom for about £5.50, or just about $11 U.S. at recent exchange rates. This may be the best add-on device yet developed to deal with this problem. —Ross Robinson

Book Reviews Solicited

Ever felt frustrated by your inability to find useful information on matters audiophilic? Most of us have. The BAS is doing what it can to combat the problem. One of our strategies is book reviews. If you've stumbled upon a gem, why not share the wealth? Send us a review. It doesn't have to be long, and it doesn't have to read like Hemingway. And, though we prefer a review already written, if you can't manage it, send the title, the author's name, and the publisher to me care of Box 7.

— Michael Riggs

In the Literature

[A major contribution this month comes from Dana Craig.]

Audio, Jan. 1976

- A skinny issue (as are many in January) with a short piece on reverbs, soon to be outdated by upcoming products (p. 18); a review of tuner specification laws and Audio's test methods (p. 38); another supertuner (LUX T-310) at an almost reasonable price (p. 48); and a nice commentary on the Advent table radio (p. 52).

Audio Amateur, 2/75

This is an especially good issue for the truly experimentally inclined audiophile.

- A Variable Frequency Equalizer: A build-it project, with kit to be announced, for a device with a small number of bands, each able to be centered at will within a portion of the audio spectrum. (p. 3)
- How to Form an Audio Group, by Jim Brinton (BAS) and Charles Repka (NY Audio Society): Good information for those of you interested in forming local groups apart from the BAS. (p. 8)
- Ampzilla: An already out-dated report on this deleted kit, but still good reading, if perhaps a future source of controversy. (p. 11)
- Speaker Evaluation: Ear or Machine?, Part 2: A good conclusion to the less interesting first installment. (p. 20)
- Active Filters: A good companion for the construction project in this issue. (p. 28)

Audio Engineering Society, Journal of the, Oct. 1975

- Design Criteria for a Universal Compander for the Elimination of Audible Noise in Tape, Disc, and Broadcast Systems: Master's thesis that compares Dolby A and B, Burwen Noise Eliminator, and dbx systems against the criteria necessary for an ideal compander. Conclusion: "It was shown that current noise reduction systems fail to meet several of these constraints."
- Improvements in Cutting Styli for CD-4 Discs: Discussion of CD-4 cutting styli by JVC engineers and John Eargle. Has two photomicrographs of CD-4 grooves—How can any cartridge possibly track them?
- The Sound of One Hand Clapping: Richard Heyser suggests that the impulse produced by a "clap stick" in old sound tracks may be used to "decode" them to obtain the original sound by appropriate computer processing of the signal.

db, Nov. 1975

- Echo and Reverb: John Woram discusses various techniques of simulating acoustic delays—tape delay, digital delay lines, acoustic delay lines (Cooper Time Cube), spring reverb, etc. (p. 34)
- Multi-Track Phasing Errors: Deals with tape head misalignments. (p. 40)
- Balanced and Unbalanced Lines: Reprint article. (p. 43)

db, Dec. 1975

- The CBS Technology Center: Discusses activities at CBS Technology Center (formerly CBS Laboratories) including digital audio technology, SQ decoders and encoders, and those strange looking DVX speakers. (p. 22)
- The Signal Path: A long article on function generators using IC's by Walter Jung.

Dymek News

- This is an occasional newsletter published by Mckay Dymek (675 N. Park Avenue, Pomona, Calif. 91766), the only manufacturer of hi-fi AM gear. Mostly advertising, but it's free, and may be of interest to hams, dx'ers, etc.

Electronic Design, Dec. 6, 1975

- A very short piece on the question of crossovers in multi-speaker systems and a way to smooth out response in the region where drivers overlap. (p. 96)

Electronics, Oct. 16, 1975

- Planar Vertical FET Process Improves Efficiency, Reduces Production Costs: Toshiba has two prototype VFET's for audio, one for audio output and one for audio driver stage. (p. 53)

Electronics, Nov. 27, 1975

- International Electronic Devices Meeting: Discussion of new power devices: 1) Hitachi has a power MOSFET that can handle 20 amperes, has breakdown of 85 volts, and cutoff frequency of 1.5 MHz; 2) Toshiba has a new JFET for audio application that can deliver 50 watts into 8 ohms when used push-pull, with voltage gain of 5, breakdown of 60 volts source to gate and 200 volts drain to gate; 3) Sony has new bipolar power transistors with breakdown voltage to 10 kV! (p. 100)

Electronics, Dec. 11, 1975

- Koss's Bastiaans Believes the "Tailored Sound" Will Sell: Bastiaans is new engineering VP, formerly chief engineer. Interesting quote: "We'll also go more to active electrical controls and we're developing things that work on a rechargeable nicad battery, for example."

Electronotes, Aug. 1975

- Special issue devoted to analog delay techniques, including CCD's, BBD's, digital filters, and correlation. Not for the neophyte, but interesting nevertheless.

Electronotes, Sept. 1975

- Selection of Op-Amps for Power Bandwidth: Includes discussion of 741 and slew rates.

For those not familiar with Electronotes, it is a one-man enterprise, run by Bernie Hutchins, a graduate EE student at Cornell. Its full title is: Electronotes Newsletter of the Musical Engineering Group. It resembles the Speaker in format, with bibliographies, classified ads, readers' circuits, letters, theory, new ideas for designs, etc. Hutchins also makes group purchases of hard-to-get special devices. There is also a new handbook on music engineering as well as subscription services to bibliographies and a lending library of reprint articles. For those interested in music synthesizers, this is an absolute must, and there is also much of interest to audiophiles.

Popular Electronics, Jan. 1976

- A Wireless Audio System for Remote Speakers: Not hi-fi, but very convenient and inexpensive in "kit" form. This device is essentially a good quality wireless intercom with an added low-power amplifier. Distortion is high, but if some of this is in the amp, the transmitter/receiver sections could be of some interest. (p. 35)

Radio-Electronics, Jan. 1976

- Review of the Lux L-100 Amplifier. (p. 42)
- Turntable Drive Systems: (p. 49)

Stereophile, Autumn (3) 1975

- With unusual rapidity, we have another issue, this one a bit skinny, to put the publisher back in sync with the cover dates. Major reviews include the huge Fulton FMI J-modular and the Infinity Servo-Static 1A systems, with the results coming out about "equal." The issue is rounded out with a short preview of the inexpensive ($300 complete) FMI/Pro Musica record playing system, a possibly important editorial about attempts to improve the sound coming from commercial record companies, and the expected replies to Mr. Gammon after his inept defense of the Vestigal arm technical design.

Stereo Review, Jan. 1976

- Can You Really Hear Those Hi-Fi Specs?: A rather strange explanation of some psychoacoustic effects. Informative if a bit lacking in completeness of content. Perhaps a BAS member might do better in a coming issue. (p. 52)
- Review of the Accuphase T-100 points up a winner, rated number one (in its price range) by the BAS's best informed tuner reviewer. (p. 25)
- The Survival of the Avant Garde: Good reading for the less conservative members. (p. 66)

Wireless World, Nov. 1975

- Audio Fair Preview: This preview, review stuff is getting more boring than the election deluge.
- Crossover Networks and Phase Response: B&O present their current philosophical hangups, which do have some credibility.

The Ride of the New Jersey "Epistemologiphiles"

Daniel Shanefield and John Sprague

[Editor's Note: The appearance of the "Epistemology Department" in the October 1975 issue of the BAS Speaker proved too much for the regional champion of "hands-on" epistemology, "Jersey Dan" Shanefield, who herewith weighs in with some notes on that topic derived from his numerous and well-reported experiments in A-B comparisons of hi-fi components. Ditto for "Jersey John" Sprague, who (more in sorrow than in pedantry) points out a number of factors in A-B situations almost impossible to control. Shanefield's notes are, in the end, more optimistic than Sprague's from the audiophile's point of view, but both his and Sprague's are important enough to strike terror (nay, even caution!) into any subjective hi-fi reviewer, so don't stop reading this one until the end. And look for more "Epi-tomes" from Les "Winnipeg Wild Man" Leventhal now that the Canadian mails are moving once again.

On a more serious note, BAS members are learning more about accurate comparisons of hi-fi components than we had imagined possible as the epistemology question is pursued. Leventhal, Sprague, and Shanefield are telling us important things, because it seems that especially in the case of hi-fi, the matter of how we learn, how we sense, and how we form opinions has a great deal to do with the components we like, and therefore buy. And also a great deal to do with our resulting enjoyment of the hobby and of music.—Jim Brinton]

CONFESSIONS OF AN EXPERIMENTAL EPISTEMOLOGIST

Daniel Shanefield

Even though I love to argue, I have to agree with the theoretical ideas presented in the October BAS Speaker by fellow "epistemologiphiles" Les Leventhal and John Sprague. But now I would like to throw a few experimental logs on the fire; I think you'll find that the experiments bear heavily on the philosophy.

In order to run A-B tests on amplifiers, I have assembled a chain of components from a list recommended by The Absolute Sound. These components also got good reviews in other magazines. While they are not necessarily the very best, they are universally well thought of, so it is unlikely that I am masking a bad component by having a seriously weak link somewhere in the chain. But just to double check, I also repeated these tests with two other, completely independent, chains of components.

The "reference system" consists of a Thorens TD-160, SME 3009 with non-detachable head shell, ADC XLM, Dyna PAS-3X, Soundcraftsmen 20-12, Dynaco ST-400, and Magnapan MG-II speakers.

I've interposed several other power amplifiers in place of the Dyna 400, including Lafayette cheapies, and middle-aged middle-class McIntoshes. Listening juries included males and females from 6 to 45 years old, and tests were double blind.

There are audible differences between power amps when one is simply plugged in to replace another. We can divide these differences into three categories: 1) frequency response, 2) distortion, and 3) other horrors. I find that there are random differences in frequency response among units of the same model (as measured with a sweep generator and meter). I further find that these differences are audible. Further still, I find that they can be carefully equalized out to the point where any residual differences are not then audible in double-blind tests. Other BAS members said much the same thing in the November 1974 issue of the BAS Speaker.

I have not used the very best components in my chain, and the very best pairs of ears have not ventured into my parlor. But I am confident that the whole thing qualifies as an "audiophile-grade" experiment, at least, so let us suppose that these results are statistically typical and significant. (You don't have to believe this yet; please just assume it's so and bear with me to the end. I think you'll see that I get logical leverage out of it.)

If these results are significant, what does this tell the hi-fi philosopher? For one thing, it tells him that listening tests of power amps by Mr. X probably are meaningless for Mr. Y, because there is random variation in something which is audible, namely frequency response. It also tells him that since this variation can be equalized out, we can forget about the distortion and other horrors because they are not audible in generally accepted modern power amps, assuming we are operating well below clipping levels. [Would some become audible once the frequency response variations were nulled out? —Devil's Advocate]

Now, how about Mr. X testing a given unit, pronouncing it fit, and then passing it on to Mr. Y. Well, it turns out from my experiments that putting a given unit in another system can alter frequency response to an audible degree. In one system, amplifier A can sound duller than amplifier B, but in another system, they might sound the same.

The cause of such differences sometimes is referred to as compatibility (see Julian Hirsch's "Technical Talk," Stereo Review, Jan. 1974, p. 61), sometimes as impedance matching, and sometimes as loading. I'll call it a "loading problem" and give three examples of the many things that can cause it.

Example One. Consider a tube-type preamp with a 1000-ohm output impedance. We have long cables coming from the tape output jacks and going to a recorder's input, to an equalizer, an expander, etc., all of which are in parallel. Also, we have more than one power amplifier being driven as, after all, we are doing A-B tests. Even though the tape-monitor switch is not actuated, and the tape input jacks therefore are not hooked in, it is important to realize that the tape output cables are, and that they and all those inputs are providing a good deal of capacitive loading. The 1000-ohm series output impedance of the preamp plus all that capacitance in parallel constitutes an R-C filter that acts to roll off treble response. This effect can be strong at 20,000 Hz, weak at 10,000 Hz, and minimal at 3000 Hz—but it is distinctly audible.

A partial solution to this problem is to interpose a component with a very high input impedance and a very low output impedance, such as a good graphic equalizer with the controls set to the flat position. But in many cases there still will be a slight effect. (Unfortunately, you can't look at a circuit diagram and easily calculate the magnitudes of effects like these. For one thing, dynamic resistance is often not the simple value of the output resistor. And remember that we are looking for response differences as small as 0.1 dB, which, voltage-wise, is only 1%.)

At any rate, one should minimize the number of parallel capacitance "hangers-on" during A-B testing.

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Example Two. Consider a transistorized preamp with a series-coupling capacitor at its output. This, say, drives a power amp such as the Phase Linear 400, which has a pretty low input impedance—10,000 ohms. Or maybe it drives an amp with a level-control pot that can cause the input impedance to be quite low at some settings. For example, at high-level settings, the input sometimes is directed right into the base of a transistor having an emitter that is nearly at ground potential. The series capacitance plus the low parallel resistance constitutes an R-C filter that will act to cut bass response. (Also, if the capacitor loads up on each current swing, the phase will be greatly shifted, but that's another matter.)

Different components with differing input and output circuits will interact differently than those above. Once again, there are few absolutes (if any) that carry over from system to system.

Example Three. Let's say a power amplifier has a fairly low damping factor and therefore an output impedance that is fairly close to 8 ohms. In the old days of power transformers, we wanted to match impedances to get maximum power into the speakers. Nowadays, it is clipping that limits power, so there's less need to equalize impedances. But let's say that the amplifier's output impedance and the speaker's impedance are nearly equal. Then in the bass region of the response curve, where there usually is an impedance peak for a dynamic speaker due to resonance, there will be a slight drop in the fraction of the power dissipated in the speaker and thus a slight decrease in bass response. (Again, see Julian Hirsch's "Technical Talk," Stereo Review, Sept. 1975, p. 32.)

For a pure electrostatic or a dynamic speaker with a highly capacitive crossover network, there is an impedance drop in the treble range, or elsewhere in the response curve. Again, since matched impedances give maximum power with such a setup, response changes slightly when things are unmatched.

With modern power amps, it's more realistic not to expect matched impedances. So, being unmatched to begin with, further unmatching (depending on its magnitude) might have little effect. But again, we are looking for 1% effects, and they may well be audible.

In this case, maybe the best thing to do is to put a 16-ohm resistor in series with each speaker and really unmatch things, thus flattening out all curves. This would also protect power amps from deep impedance dips with some electrostatics. You would need power to spare, but only 4.8 dB worth.

Explanation. The above is intended to show how power amps might sound different, depending on which one is in use, and with which other components. In other words, it's intended to convince you that I might be telling you a significant truth when I say that my jury does not hear differences when I carefully equalize. This is in spite of other testers who do hear differences. Those who don't equalize also are telling a truth, but it is not in conflict with my truth. But their truth is not a significant one, because it varies randomly due to the above causes.

You can see how the experimental fact of randomness weighs heavily upon the meaning. (I suppose it would be hip to call it "pheomenological" rather than "experimental" judging from what one reads in The Physical Review Letters these days; or how about "behavioristic?")

We're just less interested in catching a bad-sounding amp if we know that its badness is random, or pseudo-random, but not easily predictable. [Maybe we should take reports of good-sounding amplifiers with a grain of salt for similar reasons.—Ed.]

Now it is time for me to face the glaring lights and admit that I might be factually wrong. Maybe a pair of Golden Ears will eventually enter my parlor and burst my balloon by easily telling the difference between power amps in a below-clipping, equalized, double-blind test. Maybe.

(Similar, though not identical, tests have been reported by other BAS members in the BAS Speaker, Nov. 1974, pp. 1 and 3, and by Julian Hirsch in Stereo Review, Nov. 1975, p. 27. So far, all balloons appear intact.)

But all you have to do is believe that I have a reasonable probability of being correct, and you'll never again fully trust those who don't equalize and who don't use double-blind tests. Once you believe that I might be right, the burden of proof falls upon them. That's the source of my logical leverage.

And About Records. Now let us imagine that Mr. Y tests several power amps in his own system and picks the one he likes best under these specific conditions—instead of just trusting Mr. X. Suppose that he uses the same reference recordings that I used, a set that consists of seven chosen from The Absolute Sound's recommended list. Suppose the equalizer is switched out or flat, and that the preamp's tone controls are flat or bypassed (as in most of my recent tests).

Having chosen a certain amplifier, for his specific situation, and with one of these recordings, Mr. Y's trouble now is that other recordings from his collection would lead him to chose a different amplifier!

For example, try choosing a power amp using the London Phase-4 recording of "Pictures at an Exhibition," SPC-21006. Then try again with another disc from the same firm: "Scheherazade," SPC-21005. Totally different sound and results; these records must have been made by different engineers, with different studio equalization settings, mike positions, etc.

Conclusions. Then how do we ever get meaningful test results? Here's what I recommend: Equalize by ear to make one overall system sound good, with the recording included as part of the system. Then using an oscillator and meter, note the response curve of just the equalizer and power amp. Use another equalizer and power amp, setting them to yield the same response curve. Switch back and forth between them, with double-blind control, and you won't be able to tell the difference.

Do you have to re-equalize extensively for each recording? No, in most cases, a touch at the high- and low-frequency extremes is enough to make a satisfying correction to the sound. But little touches make big differences—so big that they overwhelm the "apparent" differences between power amps.

When carefully considered and carried out, this procedure is fully satisfying to both the philosophical mind and audiophile ears.

IT'S MORE COMPLEX THAN YOU THOUGHT . . .

John Sprague

In the October 1975 BAS Speaker, Dr. Les Leventhal stated that the amplifier that must be adjusted louder (by meter) to sound as good as the reference amplifier is the one with the higher distortion. On first reading, I thought—"No, he has it backward." But it isn't as simple as that.

Changes in sound level (or amplifier drive voltage) as measured don't correlate linearly with perceived sound level. There also are some interesting nonlinear equipment effects; we'll approach these first.

Equipment Nonlinearities. Amplifiers with output transformers often exhibit rolloff at both ends of their frequency range, and often within the audible spectrum. Old IHFM standards encouraged this by allowing ratings of X watts to be specified with 0.5 x X watts (-3 dB) at 20 Hz and 20,000 Hz for a specified combination of output power and distortion. The turnover points for

a

these rolloffs might vary among different designs for the same rated power, and presumably any such amplifier run at more than half its rated power would have poor distortion and nonlinear power output (i.e., drooping output at the band extremes). The distortion components also could vary among designs, especially when an amplifier was overdriven (typical clinic procedure), as could the parts of the audible band affected. Not just increased harmonic distortion could result, but increased intermodulation as well. The Audio League Reports have data on this.

Loudspeakers vary considerably in their distortion at different drive levels and frequencies, even for second and third harmonics, as recent test reports have shown. These often are non-linear too. It seems likely that their frequency/power response(s) also varies at different drive levels, whether adjusted for distortion or not. Woofers, especially, tend to "run out of breath" and start doubling at low frequencies as power is increased.

Phase shift? It is rampant throughout the record and playback chains, and needs an article to itself written by someone willing to defend his views against different ones.

And Then There's the Bloody Human Ear. Thanks to Fletcher and Munson, we know (approximately) that average hearing below 1000 Hz is more nearly linear above 90 dB SPL than at lower levels. More complex changes in apparent hearing sensitivity occur above 1000 Hz with variations in SPL, but often are neglected. Too bad, as this includes the so-called presence range.

And what about distortion originating at higher frequencies? Most likely it is rarely noticed, but can it be dismissed?

We all remember what the Fletcher-Munson curves look like and what they mean. When playing signals back more softly than the perceived level of the original performance, the bass should be boosted to restore the illusion of frequency balance. The greater the difference in the two levels, the greater the needed boost. [And as for the loudness controls on hi-fi equipment—good luck.—Ed.]

But so far as apparent sound quality is concerned, these nonlinear effects at" the higher end of the band may be just as important, or more important, than those in the bass range. The usual (and Fletcher and Munson's) reference frequency is 1000 Hz. The ear's range of greatest sensitivity, though, is between 3000 and 4000 Hz, and it shifts in frequency as SPL increases.

At 60 dB SPL, average sensitivity is within about + 1, -2 dB from 400 to 5000 Hz. At 90 dB SPL, it is within about +1, -2 dB from about 50 to 2000 Hz. At 100 dB SPL, it is within about +1, -2 dB from about 20 Hz (or below) to 1500 Hz.

There is a dip in sensitivity between 2000 and 5000 Hz which is very slight at 60 dB SPL, but more pronounced at both higher and lower levels. The sensitivity difference between 1000 and 2000 Hz is slight between 40 and 90 dB SPL, and greater at lower and higher levels. At 100 dB SPL, the ear's average sensitivity to 2000-Hz energy is about -4 dB relative to its 1000 Hz sensitivity, about -5 dB at 110 dB SPL, and about -3 dB at 120 dB SPL. Thus a 105 dB SPL tone at 2000 Hz may sound as loud as a 110 dB SPL tone at 1000 Hz.

The implications of this on subjective judgment in a playback situation are sobering. If you play music that would normally (in live performance) be heard at 60 to 70 dB SPL (and hopefully it was recorded that way), but do so at 90 to 100 dB SPL, you will experience a subjective boost of about 4 dB in output between 3000 and 5000 Hz. A subjective boost of about 5 dB in the treble should be apparent for 70 dB (real) SPL sounds reproduced at 100 dB SPL, but for 60 dB (real) SPL sounds reproduced at 90 dB SPL, the illusory boost would be only about 2 dB. For frequencies from 2000 Hz down to about 1000 Hz, there may be a 2 to 3 dB subjective boost if the original SPL was anywhere between 40 and 90 dB, and playback occurred at 100 to 120 dB SPL.

These subjective alterations to frequency balance, caused merely by playback at other than a "suitable" level (assuming perfect equipment and no studio equalization), may well be responsible for the common desire to reproduce sound at loud levels and for some of the differences of opinion among subjective reviewers.

These subjective variations in response affect the so-called presence range, and their effect thus can be confused with "naturalness" and "transparency" (so far as these can be defined). Even a difference of 10 dB in level at 1000 Hz may mean a subjective difference of 2 to 3 dB in the range about 2000 Hz.

Obviously, the relative location of these effects in the frequency band depends on the reference frequency. In this case, it's 1000 Hz. Perhaps someday (both for the benefit of subjective reviewers and audiophiles), reference-level tones will be included in recordings so that correct playback levels can be achieved at the listening position. Or maybe a future record jacket note will say something like, "For natural reproduction of the tonal balance of this recording, adjust volume so that the cymbal crash just before the end of band one on side two peaks at 98 dB SPL at the listener's position with a sound meter in its fast response position and aimed at the right-front loudspeaker."

A B.A.S. Test Report

Cartridges: Supex SD900/E Versus Shure V15-III

Alvin Foster and Ira Leonard

Audiophiles are constantly scanning hi-fi publications in an attempt to remain abreast of the latest fads and products. The penalty we pay for our myopic cultism is often out of our pockets. The latest fad or benchmark of progress is the reintroduction of the moving-coil cartridge. The Supex SD900/E is the first of this group to arouse the interest of audiophiles. Is it progress? A test which we were unable to perform must be undertaken to convince either of us that the Supex represents real progress.

In order to properly evaluate the two cartridges, identical and ideal conditions were maintained throughout the tests. The Shure and the Supex were both mounted in Decca tonearms on separate, high-quality manual turntables. Both cartridges fed a modified Marantz 7C, Phase Linear 700 amplifier, a pair of LST-1's, and a pair of LST-2's. The latter were mounted on top of the LST-1's, and both were equalized for flat frequency response. The Supex was fed into either the accompanying Supex SDT77 transformer or the newest Mark Levinson JC-1AC line powered head amp, with three gain settings, channel balancing, and selectable high-frequency compensation switches. The JC-1AC we tested was preset for maximum attenuation of the high-frequency resonance.

The list of tests included:

1. Listening test with records. The evaluations took place over an 8-hour period, and the listening test was performed first. A null switch and a mono record were used to balance the output of the system in order to prevent the effect of perhaps one cartridge sounding "better" than the other simply because it had more output into the speaker with the most nearly ideal placement in the room. A tape recording was made of each cartridge playing a particular record, and the results were used to verify our impressions after the actual listening test, without the tape recorder, was completed.

2. A frequency response plot for each cartridge. Separate plots of the Supex's response were completed with the transformer and with the pre-preamp.

3. Tracking ability. The Shure TTR-102 test record was used to adjust each cartridge to its proper tracking force and anti-skating setting. The Shure TTR-103 record served as the definitive test of tracking ability. Separation tests at 1 kHz were also completed for both cartridges.

4. Spectrum analysis. The Supex was analyzed separately with the transformer and with the pre-preamp.

5. A preamp improvement test. A record listening test was employed to see if the Supex transformer or the Mark Levinson pre-preamp were capable of making a poor or "edgy" sounding phono preamp sound "good." The edgy sounding preamps used in this test were the SAE Mk 1B and the Pioneer SC-700.

Listening Test

We listened first to the Supex with the transformer. Our immediate observations closely parallel those of other "super-fi" magazines. The Supex seemed to have far more "air" or ambience around the instruments than the Shure. Its added ambience contributed to the three dimensionality of the cartridge and gave it a greater sense of spaciousness and transparency on most records. Sibilants were heightened on all records, and transients were sharper and sometimes more musical. Perhaps the enhanced transients of the Supex account for its uncanny ability to localize instruments. Psychoacoustically, high frequencies supply the ear with directional clues. The Supex, which sounds brighter and more open than the Shure, probably supplies more of these clues. Being Shure owners, we were probably struck by the phenomenon known as "the halo effect": anything different sounds better at the first listening. The Supex also reproduced more record surface noise.

The low-end of the Supex was more detailed and tighter than the thick, muddy sound of the Shure, at least on some records. The Mark Levinson pre-preamp and Supex combination sounded more natural and realistic on a greater variety of records than did the Supex-transformer pair. The bass sounded slightly more full, and the extra brightness of the transformer was diminished. The Shure sounded more natural on a greater variety of records than the Supex combinations, but as with the Supex-Levinson combination, some transparency was lost. An excellent example of the latter phenomenon shows up on direct comparison of a speaker known to be dull, the AR-3A, with a brighter speaker, the ESS AMT-1. The ESS, particularly on rock, elicits more hall ambience, yielding a more open sound.

Frequency Response

After our listening tests were completed and our impressions confirmed, we plotted the frequency response of each cartridge with the CBS STR-100 test record. The results are given in Table 1 (variations within ±0.4 dB were not recorded). As indicated by the table, the Supex has a rising high end, and this probably accounts for the added transparency. Based solely on frequency response, the Supex would complement dull speakers while the Shure would sound best on speakers with a linear response, assuming, of course, that all records are recorded flat.

Table 1 also suggests that the rising response of the Supex accounts for its accentuation of surface noise on playback. The extra low-end punch with the pre-preamp can be explained easily by examining the slight bass rise below 60 Hz—good news for most speakers. The added low-end definition of the Supex-transformer combination was probably the result of the decrease in response around 250 Hz (mid bass) and below 30 Hz. In previous tests with the Allison speakers, we found this to be the case.

In conclusion, there is sufficient evidence to suggest that the differences we heard between the Shure and the Supex combinations were primarily a function of frequency response, not of operating principal. In order to make a definitive statement as to which is actually superior, one would have to equalize each cartridge and make intensive listening tests. Until this is done, we are of the opinion that you should use the cartridge that complements your system.

Tracking Ability

Both cartridges were super trackers, and their differences were not obvious until we used the Shure TTR-103 test record. To get ready for this "untrackability" test, we optimized each cartridge with the Shure TTR-102, which has a peak velocity of 27.1 cm/sec. The test passage

Table 1. Frequency Response

(Left Channel Only)

Frequency	Supex with Transformer	Supex with Pre-preamp	Shure V15-III
20 kHz	+7.8 dB	+3 dB	+3.5 dB
18	+4.4	+2.5	+1.5
16	+4.7	+1.2	0
14	+4.6	+1.6	0
12	+4.3	+1.8	0
10	+3.2	+1.7	0
8	+0.9	0	0
6	0	-0.9	0
5	-0.6	-0.8	0
4	-0.6	-0.6	0
3	0	0	0
2	0	0	0
1.5	0	0	0
1	0	0	0
800 Hz	0	0	0
600	0	0	0
500	0	0	0
400	0	0	0
300	-0.6	0	0
200	-0.7	0	0
150	0	0	0
100	0	0	0
80	0	0	0
60	0	0	0
50	0	+0.6	0
40	0	+1.2	0
30	-0.5	+1.7	0
25	-1	+1.9	0
20	-2.4	+1.2	0

simultaneously intermodulates 400 Hz and 12 kHz. (The 12 kHz is 12 dB down.) When mistracking occurs, a buzz is heard through the speakers and the scope displays a poor sine wave. This is also true for the TTR-103 record. The Shure required 1.25 grams and the Supex 1.64 grams to successfully pass this test. A lateral cut on the TTR-103 with a peak velocity of 31.5 cm/sec proved to be unmanageable for the Supex, but the Shure kept on tracking.

To test for separation, the CBS STR-100 1-kHz band was used. It has a sine wave recorded first in the left and then in the right channel. Each cartridge was adjusted laterally several times to achieve its optimum results: Shure -29 dB left channel, -19 dB right channel; Supex-33 dB left channel, -25 dB right channel.

Spectrum Analysis

Using a spectrum analyzer, we looked at the output of each cartridge while it was being driven by a 1-kHz sine wave recorded on the CBS STR-100 record. The scope of the spectrum analyzer was set to display the energy distribution across the frequency spectrum of 1 to 20 kHz (see Table 2).

Table 2. Decibles Below 0 Operating Level

(1-kHz fundamental)

	2nd Harmonic (2 kHz)	Percent Distortion	3rd Harmonic (3 kHz)	Percent Distortion
Shure V15-III				
Left channel	-20 dB	10%	-38 dB	1%
Right channel	-22 dB	6.2%	> -40 dB	
Supex with transformer				
Left channel	-24 dB	4.9%	> -40 dB	
Right channel	-20 dB	10%	-32 dB	1.9%
Supex with pre -preamp				
Left channel	-24 dB	4.9%	> -40 dB	
Right channel	-20 dB	10%	-32 dB	1.9%

Tonal balance cannot be demonstrated in this test because only one tone was used. However, because there were no measurable energy differences using the Supex with either the transformer or the pre-preamp, this test does suggest that the only sonic difference between the combination was the frequency response. More investigation with an equalizer is needed here also.

Because the third harmonic switched sides, from the left channel of the Shure to the right side of the Supex, we suspect the third harmonic was more a function of tracking geometry than of cartridge performance. The 1-kHz tone was recorded at "0" operating level, which is much higher than the average recording amplitude. This probably accounts for the high distortion figures. (Lateral distortion varies as the square of the signal amplitude, directly as the frequency, directly as the stylus radius, and directly as the groove velocity.) Also, we do not know the purity of the CBS 1-kHz tone. We did discover that a speck of dust on the stylus tip increased odd and even harmonic distortion dramatically. In view of the large amount of second-harmonic distortion found with both cartridges, it is reassuring to know that the ear is relatively insensitive to it.

Preamp Improvement Test

Unfortunately, neither the transformer nor the pre-preamp can compensate for a poor or "edgy" sounding preamp, i.e., a preamp that fails the Holman test and the three-record test. Using the three records and the testing procedure outlined elsewhere in this issue of the BAS Speaker, the poor preamp was never able to sound "good" or smooth.

Summary

The Shure lists for $77.50 and the Supex for $125 without the transformer, which costs an additional $100. The Mark Levinson pre-preamp costs $225, which means that you can pay from $225 to $380 for the Supex combinations.

The Shure is well shielded, but the Supex is relatively sensitive to extraneous hum fields. For example, if my ac VTVM was placed within four inches of the pre-preamp, an audible hum could be produced. With any of the Supex combinations, keep away from magnetic fields. The power to the JC-1AC is supplied by a separate shielded box, which must be kept away from the pre-preamp and the cartridge.

Which cartridge should you buy? Assuming unlimited resources, buy the cartridge that makes your system sound most natural.

A Publication of the BAS

Improving the High-End Sound of Your Loudspeaker System

Robert Graham

The Janszen 130 Electrostatic Midrange/Tweeter

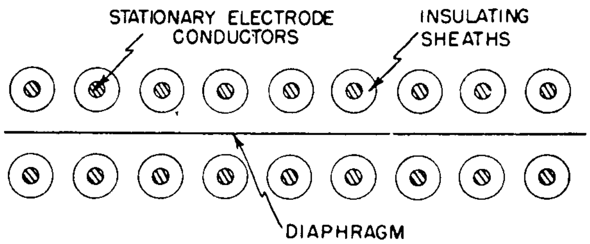
One of the popular "state-of-the-art" loudspeaker systems of the middle '50's was the AR-1W/Janszen 130 combination. The AR-1's 12-inch woofer had been recently introduced, with an impact that is now well known, and the Janszen was a newly designed electrostatic utilizing a true push-pull diaphragm. As a midrange/tweeter, the Janszen 130 was almost without equal in smoothness and detail, so the mating of it and the AR-1W was an ideal combination. The two manufacturers even had matching grill cloth.

Production of the Janszen tweeter shifted from its designer, Arthur Janszen in Cambridge, Massachusetts, to Neshaminy Electronics Corp. in Pennsylvania, where the basic design was continued in the original Model 130 add-on midrange/tweeter and also in several full-range systems utilizing dynamic woofers. A few years ago, production shifted again, this time to Electronic Industries, Inc. in Minneapolis.

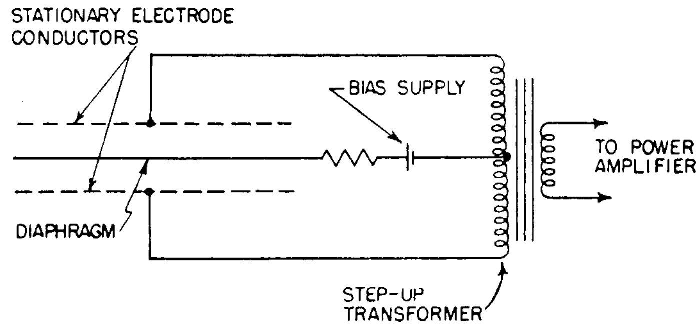
EI now has a broad line of add-on tweeters and full-range systems (using dynamic woofers for now with electrostatics coming), and the original Model 130 is still made. For those familiar with the older 130, you should know that the only similarity between the original 130 and the current design is in the distinctive cabinet shape, and even that is a bit smaller than the old model. The electrostatic elements and the power supply have been completely re-designed, and the improvement this makes is quite apparent.

The push-pull electrostatic works like this: each radiator consists of an array of insulated conductors spaced on each side of the diaphragm, which is a sheet of ultra-thin electrically conductive polyester film. The whole assembly is mounted in a plastic frame, and thus becomes a "sandwich" type of construction.

As shown in Fig. 1, each of the two sets of stationary conductors is connected to opposite ends of the secondary of a transformer. A high-voltage power supply applies -1100 V to the conductive diaphragm, and the "+" side of the supply is connected to the center tap of the transformer, so that a difference of potential is established between the diaphragm and the stationary conductors. Remember from high-school physics that "unlike charges attract and like charges repel." That is the condition that exists here, but until an audio signal is applied to the input transformer, the negatively charged diaphragm remains stationary between two equally positively charged grids. This condition makes the diaphragm receptive to any change in voltage between the conductors, and when an audio signal is coupled over to the secondary, one side of the transformer becomes more positive than the other, causing the diaphragm to be attracted toward one set of conductors and simultaneously repelled by the other set. As the audio signal swings the other way, so does the diaphragm. Thus there is a uniform push-pull effect operating on the diaphragm, and this principle is responsible for the very low distortion of the Model 130.



(a) Electrostatic element



(b) Schematic

Fig. 1. Janszen Model 130 electrostatic element and schematic

The current design of the electrostatic radiators incorporates improved materials and the alteration of the original "waffle-iron" plastic frame. Also (and this should settle anyone's fears about durability), the electrostatic elements of the current design are guaranteed unconditionally for 10 (yes, 10) years! So much for unreliability!

The power supply also is new, and is mounted on a sturdy cast aluminum plate for improved heat-sinking (what little is required). Also, whereas the old 130's level control would vary the signal's output level by only a few dB, the new version regulates the bias level itself, and can maintain control from full "on" to full "off." This greatly improves compatibility with various speaker systems.

Incidentally, the electrostatics should be used with relatively low-efficiency woofers for proper level matching. Now, as in the 1950's, a likely candidate is the AR-1W woofer, but any good low-end driver will do nicely. The Janszen 130 now crosses over at 800 Hz, so if you are using the AR woofer, I'd recommend using a good mid-bass driver to bridge the gap between the best range of the woofer (no higher than 500 Hz) and the Janszen. I'm using a 5-inch Philips mounted in a sealed sub-enclosure in the AR cabinet. (This is the same midrange as is used in the Rectilinear III, and is available from SpeakerLab, 5500 35th N.E., Seattle, Washington, for around $10.)

Janszen (EI) is currently offering six different add-on electrostatic units, ranging from a two-element Model 132 up to a large eight-element Model 138. All of the systems have internal high-pass filters; a few models contain a full crossover so that a separate woofer crossover is unnecessary. The Model 130 is normally supplied with just the high-pass filter, but it can be special-ordered from the factory with the full crossover by requesting the Model 134hp power supply instead of the 130 power supply. The Model 134hp has four radiators, but these are mounted two over two in a square box. Each radiator points somewhat sideways and up or down from its neighbor, and so provides equal vertical and horizontal dispersion. The Model 130 is the original four-element version with a broad horizontal array (120°). To me, this design looks better aesthetically when it's perched on top of an AR or similar bookshelf speaker, but the vertical dispersion may be somewhat less than the other models with vertically angled elements.

How does it sound? If I had to use an adjective, it would be "excellent." But, actually, the sound that the Janszen 130 produces is hard to describe, since it is utterly neutral and unassuming in its character. That type of phrase has become fashionable these days, using words like "uncolored," etc. No doubt many conventional speakers deserve such praise, but I have never heard midrange detail and clarity that is as easy to listen to as with this speaker. I could go on, saying it different ways, but it amounts to the same thing; this is a superb speaker.

The electrostatic, by design, <u>should</u> sound better than dynamic speakers. There is much, much less mass to move around, and the push-pull action of the elements should contribute to superior transient response and very low distortion. This is not to downgrade the better dynamic systems; there are many that I admire. But they all, to my ears, fall behind the Janszen Model 130 in overall smoothness and freedom from "speaker sound."

I have not tried to tax the power handling capability of the 130, but in normal use I have never heard the slightest hint of overload, and this with an amplifier capable of 140 watts/channel. On one occasion, while testing a new preamp, I accidentally jarred a ground wire loose, and the amplifier immediately sailed to full power. The speaker fuse opened after a couple of seconds, and the Janszen seems none the worse for the experience. I <u>won't</u> repeat the test, if I can help it.

There is one area where the better dome speakers would be better than the electrostatic, and that is dispersion. Because the electrostatic needs a relatively large, flat surface to be able to reproduce the midrange adequately, it necessarily tends to "beam" the very highest frequencies. This is noticeable if you move around in front of the Janszen. On axis, you'll hear everything there is with equal smoothness; off axis you'll notice a rolloff of the higher registers. While this is a deficiency, in theory, I find that in practice it is not. Since most people listen to music while sitting down and sitting still, normal room reflections fill in the gaps nicely. For "enthusiasts" who love to listen to music while running around the room to evaluate dispersion, this would not be a totally satisfying speaker. However, if you are content to sit quietly and listen to the music as intended, then you will be rewarded with clarity and detail that you may not have heard before.

While I have not compared the Janszen directly with full-range electrostatics like the Quad, KLH-9, etc., I feel sure that it is the equal of any of them <u>within its bandpass</u> (i.e., 800 Hz and above). I do recall, when listening to the KLH-9, a very noticeable "beam" of highs coming from the center of the panel. I notice no such intense beaming from the Model 130.

The practical advantage of the Janszen 130 (and its relatives) is that one can obtain the first-class performance of the electrostatic without going to the enormous cost of a full-range electrostatic. By relinquishing signals below 800 Hz to good dynamic speakers, you can enjoy much of the very real improvement that the electrostatic design has to offer at a cost far below that of full-range units. The standard Model 130 lists for $200. The addition of the woofer crossover power supply (134hp) adds perhaps $15. The speaker, however, is discountable. In the Boston area, K&L Sound carries the line. For literature, you may write to the manufacturer [mentioning the BAS.—Ed.] at: Electronic Industries, Inc., 7516 42nd Avenue North, Minneapolis, Minnesota 55427.

<u>Ionovac—The Tweeter With No Moving Parts</u>

In the preceding review of the Janszen Model 130 midrange/tweeter, I pointed out that its only real limitation is its lack of dispersion at the higher frequencies. While this is, in practice, not as much of a limitation as some would say, it would nevertheless be nice to have a tweeter with the clarity and detail of the electrostatic that would also provide good dispersion.

The Ionovac is such a speaker. This speaker is unique in that it uses absolutely no mechanical parts to move the air. If low-mass drivers are good, then a no-mass tweeter should be superb. This was the thinking that led Siegfried Klein to invent the ionic speaker in the early '50's.

How does it work ? The idea is to set up a small "cloud" of ionized air in the throat of a horn. Ionized air is still just air, but it can be modulated electrically and moved physically, thereby producing sound. In the Ionovac there is a horn assembly that contains a high-frequency oscillator needed to produce the ionization "cloud." The output of the oscillator is fed to an electrode located within a quartz cell. This cell is roughly the size of a pencil eraser, and is drilled out in the center (see Fig. 2). This quartz cell is seated inside a ring-type socket that is

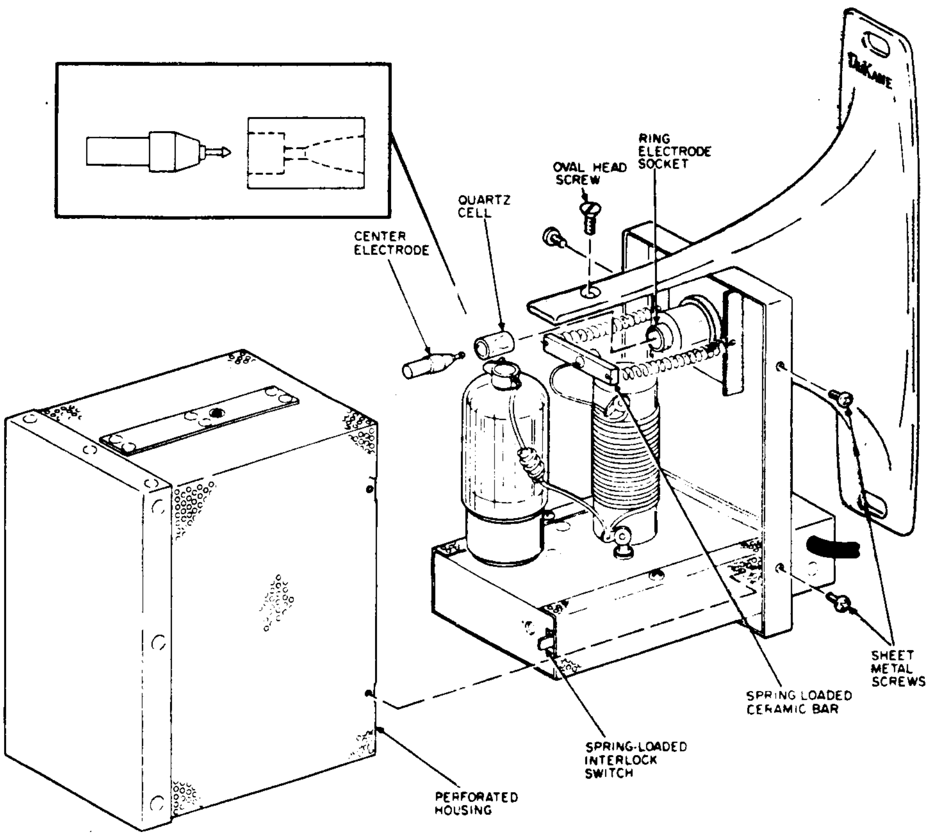


Fig. 2. Ionovac tweeter

connected to the other side of the oscillator circuit. With the application of the high-frequency (27 MHz) high-voltage oscillation, a corona of ionized air is established around the tip of the center electrode inside the quartz cell. Note that this ionized "cloud" looks out of the cell and into an exponential horn, which provides the acoustic transfer necessary to achieve good dispersion and adequate volume.

The other half of the Ionovac is the power supply. Audio is applied to an input step-up transformer, the secondary of which is in series with the screen grid of the oscillator tube. The audio thus changes the oscillation level and thereby the ionization level, producing sound.

The Ionovac originally appeared in this country in 1958 and was manufactured by Electro-Voice. The design was quite different at that time, and the unit was unreliable and short-lived. The quartz cell and electrode used by E-V usually failed after about 200 hours of use. The Ionovac disappeared relatively soon after introduction and did not re-emerge until a couple of years later when the DuKane Corp.— the United States licensee from the beginning—completely redesigned it. The new model completely cured the problems of the Electro-Voice version, but there are some people who are aware only of the early model "which didn't last." More about that later.

DuKane, a major supplier of electronics marketed under more familiar names, continued to produce the Ionovac by itself and in systems until the middle '60's , when they ceased production of the tweeter. Here is a good case of the public cheating itself of a clearly superior product. People just didn't want to bother with a speaker that required maintenance like the Ionovac does;

they didn't want to fool around with a speaker that needed to be plugged into the wall (electrostatics weren't doing too well at that time, either) and generally everyone was content to have a plain dynamic speaker that they could bring home, attach to their amplifier, and forget about. Fair enough, I suppose, but this attitude forced DuKane to shut down production, and it's really too bad, since I don't belive there has ever been a tweeter that is quite as good.

Some criticism is leveled at the Ionovac even now. Some say it requires maintenance. This is true, but the maintenance is extremely simple; just an occasional cleaning of the quartz cell and replacement of the 6GW6 oscillator tube perhaps once a year to maintain peak performance. Other people, usually with another product to push, say it has distortion near crossover. I don't believe this one. The horn shape and dimensions say that the tweeter can respond down to 1500 Hz. Since the crossover is at 3500 Hz, horn cutoff problems should be minimal. Since the actual radiating area of the ionization "cloud" is very small—about 1/8 inch in diameter—there is no need for the phasing plug required in dynamic horn tweeters. The "horn sound" sometimes experienced with such speakers is not evident with the Ionovac. The ionic process itself is not frequency-sensitive—the Ionovac could be a good midrange with the right electronics and (horn) coupling.

The Ionovac is blowout-proof, since there is no diaphragm, cone, or membrane to push around. (The Ionovac was subjected to the same preamp accident as the Janszen with no ill effects.) Transient response is just about perfect for the same reason. Even the very best and lightest dynamic and electrostatic speakers exhibit some mass, which means that some musical energy must be wasted just to overcome inertia before the speaker can begin to move. At very high frequencies, this situation can reach the point where more of the signal is being used to overcome the inertia of the speaker than to produce sound! Electrostatics, of course, reduce this problem by an order of magnitude; at the midrange frequencies, their diaphragms are so light that they behave as if they were massless. This virtue diminishes with increasing frequency, but not nearly as dramatically as with dynamic tweeters.

[Editor's Note: Realize that this inertia (and then some) must be overcome twice each cycle—at the peak and at the trough of each cycle. Graham, in viewing the situation from the point of view of the diaphragm in its at-rest position, presents it in a bit more favorable light than he might.

It takes more energy to stop and restart a moving diaphragm than simply to start one moving from its at-rest position since (viewed simply)

$$K.E. = MV^2$$

(Kinetic Energy = Mass x Velocity squared).

At higher frequencies, average diaphragm velocities are higher and because both dynamic and electrostatic diaphragms have noticeable mass (though the former much more than the latter), the multiplication by V^2 to find kinetic energy can yield some surprisingly high numbers. Because the system also must pump energy into the speaker to overcome kinetic energy—not just to move the diaphragm out of an at-rest position—the rate and direction of diaphragm movement will tend to lag any change in the input waveform by a small amount. It's physically impossible to get out acoustically exactly what you put in electronically even with the Ionovac, but much, much more so for dynamic drivers.

Because the mass of the ionic cloud at the throat of the Ionovac's horn approaches zero, the V^2 term is largely cancelled. Almost zero x V^2 still equals only almost zero. Thus, the Ionovac not only can do a better job of reproducing transients than the ear is capable of appreciating, perhaps, but, and maybe more importantly, there should be less distortion of more-or-less steady-state waveforms. This latter may account for the unit's uncanny clarity as much as any other factor.—Jim Brinton]

c

As with the electrostatic tweeter, the Ionovac relinquishes only one area to the dome tweeter—dispersion. However, the dispersion from the Ionovac is more than adequate, and much better than any electrostatic. The frequency response graph of the Ionovac (measured in a well-known Cambridge hi-fi manufacturer's anechoic chamber) shows the speaker's measured performance (see Fig. 3).

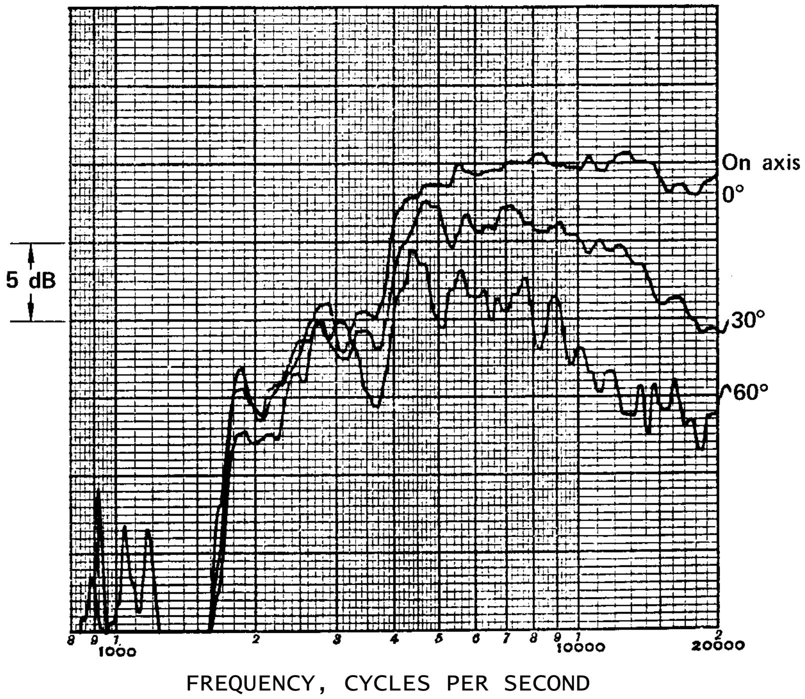


Fig. 3. Ionovac frequency response

The graphs, however, don't really tell the whole story. The Ionovac sounds much better than the graph would indicate, even though it is a smooth response. I suspect that it is the massless "diaphragm" that is responsible for this; none of the music is being lost.

As for reliability, I have had my Ionovacs since 1967, and the only component failure I've had was a diode in the power supply, and that was easily fixed. The quartz cell is the original one, and shows no sign of quitting. It is easily replaceable for around $12 through DuKane. I do find that by exercising some care in maintenance, it is possible to extend the life of the quartz cell beyond the advertised guarantee (1200 hours). Once or twice a year, I disassemble the horn unit and carefully clean out the quartz with a Q-Tip to get rid of accumulated debris from the ionization process. I also clean off the center electrode by gently burnishing the tip with no. 600 emery paper. The throat of the horn also collects some stuff from the ionization and this is cleaned out with another Q-Tip. Sometimes I wash the horn itself under running water, just to flush it really clean. (How many speakers can you do that to?) With this simple care, my Ionovacs have performed continuously for over eight years. While this housekeeping may seem like a chore, I can assure you that the sonic performance from this speaker makes it all worthwhile.

I have compared the Ionovac with many tweeters over the years, and so far not one has been as good. The Janszen comes very close, to be sure, but still does not quite make it. I once compared a straight Quad electrostatic to another Quad crossed over at 3500 Hz to the Ionovac and it was no contest. The Ionovac-equipped Quad beat the straight Quad hands down. A few dynamic tweeters do very well, but most are left far behind.

The Pair

The reason for writing about the Janszen electrostatic and the Ionovac in the same article is this: while either speaker by itself makes a profound improvement to any system it's added to, the combination of the Janszen and the Ionovac yields, to my ears at least, the very best mid-range and highs that I've ever heard. Period. This is not to discount the excellent speakers on the market, but I have yet to hear anything better.

So far as dispersion is concerned, the combination of the two speakers yields excellent imaging of instruments. It is possible to easily locate the source of solo instruments and the general relationship of the large orchestra is solid and well-defined, even off axis. That type of reproduction is indicative of good dispersion. Also, the separation of lows, middle, and highs is absolutely non-existent on either channel; that is, the music—all of it—blends smoothly together and "surrounds" the speaker system.

As with the Janszen, the sound of the Ionovac is hard to describe. Strings sound like strings and brass sounds like brass. Percussion is startling. And so on. The speaker does not intrude upon the music. It is possible to listen for hours without tiring.

I am presently attempting to locate a source of Ionovac tweeters so that interested BAS members can perhaps obtain a pair. There is one possible supplier who may have a very limited number available for (sigh!) $175 each. Inflation has even hit out-of -production items. When I purchased mine, they were $80 apiece! Any success in locating a supplier will be reported in a future issue of the *Speaker*. Occasionally, they appear on the used market, although I don't know why! (I shall never sell my Ionovacs or Janszens; in fact, they are constantly guarded by two mad dogs and three laser beams . . .)

There is a new midrange/tweeter that may have the low-mass virtues of the electrostatic and the dispersion of the horn. That is the British Fane model 701 ribbon tweeter, with a crossover at 1700 Hz. If I can obtain one to evaluate, I'll report on it in a future issue.