

THE B.A.S. SPEAKER

EDITOR-IN-CHIEF: Michael Riggs
COORDINATING EDITOR: Bernard Gregoire
STAFF: Henry Belot, Robert Borden, Joyce Brinton,
Dana Craig, Frank Farlow, Robert Graham, Lawrence
Kaufman, James Lindquist, Peter Mitchell, John
Schlafer, Jack Stevens, James Topali, Peter Watters

THE BOSTON AUDIO SOCIETY
P.O. BOX 7
BOSTON, MASSACHUSETTS 02215

PUBLISHER: James Brinton, President, BAS

VOLUME 5, NUMBER 2
NOVEMBER 1976

THE BOSTON AUDIO SOCIETY DOES NOT ENDORSE OR CRITICIZE PRODUCTS, DEALERS,
OR SERVICES. OPINIONS EXPRESSED HEREIN REFLECT THE VIEWS OF THEIR AUTHORS
AND ARE FOR THE INFORMATION OF THE MEMBERS.

In This Issue

This month's issue contains another potpourri of equipment reviews, record reviews, reader responses and reports on the two BAS meetings held during October.

For those interested in speaker construction, member Jim Nichol provides useful analyses of a goodly number of raw drivers. Peruvian member H. Gallegos reviews five damped unipivot tonearms, including the new Grace 704. Also included are James Mitchell's follow-up on Al Foster's preamp tests, a Mark Davis modification of the Pioneer RG-1 dynamic range extender, Steve Feinstein's extensive review of jazz recordings, more on Sound Guard, a short review of the new Sonex speakers and more. As a bonus, there are two meeting summaries, covering Harold Beveridge's lecture on his fascinating new full-range electrostatic speaker system and an exciting colloquy between Dr. Matti Ojala, Mark Davis and a distinguished panel on the controversial subject of transient intermodulation distortion.

But before moving on to all that good stuff, we want to thank those who have stepped forward to help run the Society and produce the Speaker. The response has been gratifying. Don't stop.

Membership dues are \$14 per year (October 1 to September 30) or portion thereof. Dues include a one-year subscription to the BAS Speaker. (Note that almost the full amount of dues is allocated to production of the Speaker. The local activities of the BAS are strictly self-supporting.) For further information and application form, write to: The Boston Audio Society, P.O. Box 7, Kenmore Square Station, Boston, Mass. 02215.

Feedback on the BSR FEW—III Equalizer

In the April issue, Harry Zwicker has some serious criticisms of the BSR FEW-III equalizer, the main one being that the nominal zero settings of the controls do not produce flat response. I own two, and I have never heard any difference between the normal and bypass positions. A review in Popular Electronics of the FEW-I indicated that setting the controls at their center positions resulted in flat response from about 10 to 100k Hz. Curious, I decided to measure one of my units.

I used a SWTPC function generator and the meters of my Pioneer SD-1100 audio scope. With the FEW-III in EQ-BYPASS, the meter's readings were flat from 10 Hz to about 30 kHz. I set the pots to their zero positions carefully, but without a lot of fussing. With the equalizer section switched in, the maximum deviation from flatness was now $\pm 1/2$ dB over the same frequency range. I can only conclude that Mr. Zwicker owns a defective unit. [See reply below --Ed.]

The gain of the FEW-III, controls centered, is nominally +0, -1 dB. If you have a unit with slightly less than unity gain, this loss of gain could sound like veiling when a direct comparison is made. The best situation is where you buy really flat speakers and use the equalizer primarily for correcting poor program material, so the sound quality of the equalizer may not be of major concern to most users, anyhow.

Mr. Zwicker complains about the lack of over-all gain pots for each channel. These would only be useful for comparing the equalized response with "flat", since there is no way that equalizing two channels for flat response could cause them to become unbalanced with respect to each other. [See reply below -- Ed.] The meter sensitivity pots have only short, partially-recessed plastic extensions on them, and I agree with Mr. Zwicker that they are hard to turn. But the service manual tells how to align the meters, and with a little bit of work you can get them to read within 1 dB of absolute over the 24 dB range, which is pretty good. They are also better than $\pm 1/2$ dB from 20 to 20K Hz.

Lest you think that I am totally enamored of the FEW-III, I must tell you that the circuit boards are garbage. The base material is very brittle, and the plating is thin and poorly attached. Any soldering must be done carefully, or you will vaporize the copper (I'm not kidding). On the other hand, BSR's service is outstanding. I was sent a service manual at no cost, and a defective transistor was replaced in a few days with no quibbling. Inquiries are answered promptly. There is a modification which eliminates the turn-on transients, but I don't know whether or not it reduces the BYPASS transient.

The BSR FEW-III has two more bands and all the switching flexibility of the Soundcrafts - men at half the price. I would think twice before passing over the BSR.

It would seem that a product which is not "average" is more likely to be defective than superior, so perhaps the Speaker should have a policy of not publishing very favorable or unfavorable reviews unless the reviewer has had a chance to audition at least two samples of the product. A lot of extra trouble, yes, but it would increase your reliability, and maybe save a lot of money and/or anguish.

I wish I had joined the BAS a long time ago; the Speaker is the most interesting audio publication in the United States!

-- William Sommerwerck (Maryland)

Reply: Good comments. The FEW-III reported on in the April issue was one of the first off the production line, and perhaps the tolerance on the pots has been upgraded. I do find, when balancing speakers in non-symmetric room locations, that channel rebalancing is required, and for \$2.00 BSR certainly could have added gain controls. The better-made Soundcrafts-men is still probably a safer, if not better, buy.

-- H. Zwicker

For Sale

- *2-SWTPC Universal Tiger amps mounted in a single custom-made walnut cabinet. Output transformers are installed on special extra massive heat sinks. \$130 or best offer. Call Ken (617) 646-3427 anytime.
- *KEF 104's (African Walnut) 6 mos. old, \$475 per pair. Harmon Kardon CITATION 12, \$175. Gary Rancourt, 369-1949 evenings.
- *Quad current-dumping stereo amplifier. 100 W/ch. Less than six months old. Must sell in order to meet prior financial commitment. Asking \$330, but will consider any reasonable offer. William Bell, (202)265-6229 evenings and weekends.
- *Audio Research D-76A, tuned to peak performance, \$950. B&W DM2a speakers with stands, \$580. Both mint and one year old. Alan Van Savage, (201)783-9488.
- *Unused Dual 1225 turntables complete with base, cover, and M91E pickup, half price (\$90) while the supply lasts. Call Larry at 964-3988 evenings and weekends.

WBUR Friends

An organization has recently been formed, known as the "WBUR Friends Group, Inc.," for the purpose of spreading the word about the fine programming available at 90.9 MHz. (In addition to their extensive jazz, classical and public affairs programming, WBUR broadcasts "Shop Talk.") Meetings are held about once a month at the station's Kenmore Square studios, and anyone interested in joining or in further details is advised to call Rich Akell at (617)767-2842. [Ed. note: "Shop Talk" is an audio miscellanea co-hosted by BAS members Richard Goldwater and Peter Mitchell. B. F.G.]

George Szell Concerts Wanted

I would like to contact members who have good-sounding tapes of George Szell's concerts with the Cleveland Orchestra or New York Philharmonic, as well as other historical material.
-- Michael Miller, (617)661-3331 (Massachusetts)

Errata

In the June issue of the Speaker, in the note about the Lahti Switchbox, on Point #4 you made a mistake in saying that "all grounds on the low-level side of the box are tied together and floating." It should read "high-level side."
-- David Sherwood (New Jersey)

Modification of the Pioneer RG-1 Dynamic Range Enhancer

The Pioneer RG-1 appears to be one of the better products of this type on the market, in part because of its novel ability to adjust its effective timing characteristics automatically in response to the dynamics of the program material. However, its performance can be improved somewhat by the following two modifications. The modifications should present no problem to anyone with a little experience in circuitry and soldering, especially as Pioneer is good enough to include a circuit diagram in the instruction book.

The first modification affects the filter in the level circuit, which allows frequencies primarily between 1.2 and 3 kHz to affect the level circuitry. The rationale given for this is that it allows the unit to be sensitive to changes in the harmonic content of the program material. Unfortunately, harmonic content appears to exert only a secondary effect on perceived loudness. As a result, the expander unnaturally emphasizes vocal sibilants and instruments with strong high frequency content. This can be corrected by adding a pair of 0.005 pF capacitors, one across C7

(left channel) and one across C8 (right channel). This permits the level circuit sensitivity to more closely approximate the sensitivity of the human ear for the levels likely to be encountered, with a primary range of 200 Hz to 3 kHz and gentle rolloffs above and below these points.

The second modification changes the fast attack time, which happens to be quite a bit shorter than it need be for any musical transients. The unhappy result is that record ticks that might otherwise go unnoticed are expanded into major pops. The solution is to lift the emitters of transistors Q13 (left) and Q14 (right) and to insert a 4700 ohm resistor in series with each. This yields a 5 ms fast attack time, which is rapid enough to follow any sudden musical transients but slow enough to ignore record ticks.

-- Mark Davis (Massachusetts)

The Compleat Acoustic Suspension

I have verified, by experiment, one of the design principles incorporated into the Cizek speaker. Extra care has been taken to seal the cabinet and drivers so that the original acoustic suspension concept has really been adhered to. In free air, the woofer cone is very compliant and cannot tolerate much low-frequency high-power music-- it's simply not damped enough. In the cabinet, however, the enclosed air supplies the necessary restoring force. It is Roy Cizek's contention that many so-called acoustic suspension systems fail to completely utilize the principle, and that at lower frequencies, many of them behave almost as if they were ported enclosure designs, but without the benefit of having a woofer designed for use in a ported cabinet -- in other words, a compromise of design.

In the Cizek speaker, the cone is treated to prevent air from seeping through the cone surface; likewise, the edge suspension is treated with a thin vinyl coating. What does this accomplish? I made a silicone rubber solution and painted a thin layer on the edge suspension of my older AR-1. Prior to doing this, the cone had a return rate of about three seconds: that is, if I depressed the cone to its limit, it would take three seconds to return to center. After applying the silicone rubber treatment, the spring had increased very noticeably, and the cone now has almost a seven-second return rate. This means that the Q of the woofer has been increased, the frequency of resonance is unaffected, and the susceptibility to infrasonic garbage has been reduced. If the amplifier has the capability to amplify low-frequency interference, then the woofer will try to reproduce it. Even record warp, which might be as slow as 2 Hz, will attempt to get through. If the woofer's return rate is slower than this kind of rumble, the sound will be cleaner. Even with amplifiers that do not reproduce infrasonic signals, this is a desirable way to have a woofer behave. My AR-1W's really do sound better to me.

-- Bob Graham (Massachusetts)

TIM

Regarding Holman's statement (BAS Speaker, April 1976) that slew rate limiting is an abrupt phenomenon, I must disagree that T.I.M. can be described in the same way in all circumstances. Slew rate limiting is often caused by input stage overload and in non-inverting amplifier stages (virtually all phono stages) the input stage is effectively outside the feedback loop. Therefore distortion generated by the input stage before overload can be significant. About 0.15% distortion is generated by the input stage at 1/10 of slew rate limiting. An inverting input would enclose the input stage within the feedback loop but is excessively noisy for most phono stages. This situation can also cause a discrepancy in distortion vs. slew rate in power amplifiers. This is perhaps why the Marantz 500, with 11 V/ μ s slew rate and inverting input, tended to sound better than some other amplifiers with higher slew rate but with non-inverting inputs.

-- John Curl (London)

Licenses to Record in the UK

A most interesting matter has come to my attention and may be of interest to Society members. In reading a Sansui advertisement in the June issue of Hi-Fi News, p. 108, I noted: "UK listeners now have some of the finest FM stereo transmissions available anywhere. All you need

to capture them for your own private enjoyment is an Amateur Recording Licence and a good quality cassette deck. For more information contact: MCPS-Elgar House, 380 Streatham High Road, London SW 16 6HR. " I wrote for this information and received the following letter in reply:

Whilst we, as a Society, represent 95% of the World's music repertoire, our territory is restricted to the United Kingdom, the British Commonwealth, excluding Canada and Australia, but including the Republic of Ireland; our own British repertoire being controlled in the States by the New York-based Harry Fox Agency, Inc. , office. From this, you will gather that the Licence that we issue to members of the public in the United Kingdom would have no standing in the States, but for your consideration, I am enclosing a copy of our specimen Licence.

Over the past few years, there has been a considerable increase in the sale of new sophisticated reproducing equipment and, of course, a corresponding increase in the sale of blank cartridges and cassettes. The United Kingdom Music Industry, together with the Record Industry, has for some time expressed concern about the domestic reproduction from existing gramophone records or television and radio. Both this Society and the Industry have made representation to our Parliament for a royalty to be levied on either the sale of new equipment or the sale of blank cartridges. This matter is being studied at this time by a special Government Committee. In the meantime, we have advertised our existing Amateur Recording Licence to ensure that members of the public are aware that the Music Industry generally has a right under the existing Copyright Act to obtain payment of royalty on domestic recordings. At this time, several thousand private individuals have applied for our Licence.

In the Federal Republic of Germany some three years ago the Government passed Legislation placing a royalty on the sale of all new equipment. This royalty is calculated at 5% of the recommended selling price, and is paid to our sister Society in Germany, who distributes the monies collected to their own Members, the record industry and a small part to the German Literary Society.

In conclusion, I must confess that I am not sure whether the American Copyright Legislation does extend to private recordings, but if you do take steps to establish any scheme on private recordings through your Boston Audio Society, I would be happy to hear of any results you may obtain.

-- John M. Edwards, General Manager, Mechanical-Copyright Protection Society Limited, 380 Streatham High Road, London SW16 6HR, August 4, 1976

This is an intriguing idea, and I wonder what others think about trying to have the same actions effected here. I imagine that the large controlling interests such as ASCAP, etc. , for popular music, and the various recording companies who own the copyrights to music and recordings would be interested in endorsing for application here the system that is being attempted in Europe. This would take a large effort on our part to crusade for adoption and ultimate government support of collecting royalties on, for example, the sale of raw tape. This money could then go to the organizations who now control all recorded music, as well as to the musicians, who at this time do not receive any royalties from copying that is now being done illegally.

It appears to me that adoption in the U.S. of a policy such as that suggested by the MCPS might eliminate opposition to tape recording of records and, especially of live performances, as revenue would accrue from the sale of the tape.

This matter is presented for discussion by the members and possible adoption of some positive action to bring a desired change to the home recording scene. I am open to suggestions on how to proceed with a letter writing campaign to the "powers that be." A list of all "these" would also be appreciated. However, direct action from the BAS might be more effective.

-- Nathan Garfinkle (California)

[Editor's note: Further discussion of "licensing" to allow recording for private use from broadcasts or commercial recordings would seem to be academic at least in the United States. The courts have recognized the impossibility of enforcing laws which would prevent casual home-type

copying. The so-called "pirating" of this material for reproduction and resale is another matter. The courts have recently dealt dramatically with several pirating cases. The amateur recording of live performances is currently widely forbidden, with the exception being performances by non-professional musical groups of public domain material. If some arrangement for financial remuneration for composers and performers could be worked out, one can imagine the growth possibilities for the first class tape equipment market. On the other hand, it's hard to imagine the appearance of a major concert hall with the microphones of several hundred recordists hung from the ceiling. Perhaps there are alternatives. Reader comments are welcome. B. F. G.

Preamplifier Testing and Evaluation

Congratulations to Tom Holman and Alvin Foster on their phono preamplifier research and on their new ideas for correlating measurements with listening tests. Foster's overload vs. frequency test is an interesting one and requires only a sine wave generator, a vacuum tube voltmeter and an oscilloscope. Since I have three preamps on hand, I decided to compare listening tests with the overload test.

The three preamps are the Audio Research SP-3A-1, the DB Systems Precision Preamp, and the Marantz 7T. The SP-3A-1 is a tube unit widely recognized for its smooth musical sound. The DB Systems unit is a modern transistor design by David Hadaway with very low harmonic distortion and noise. The Marantz 7T is an early transistor design, generally regarded as being somewhat edgy.

In my listening opinion, both the SP-3A-1 and the DB are excellent. The 7T is noticeably less clear in the high frequencies. Violins and voice sometimes sound grainy. Although the SP-3A-1 and the DB are both excellent, they do not sound identical. The differences are subtle, with the SP-3A-1 sounding somewhat more musical and lucid in the important middle range. The DB seems to have more detail in transient highs. On very good records with lots of bass impact and zingy highs the DB may be preferred, but on complex symphonic or voice recordings the SP-3A-1 is favored.

The frequency vs. overload tests (made at the tape outputs) are as follows:

Preamp	Frequency (V Overload)			Overload Ratio 50 kHz/10 kHz	Subjective Audio Quality
	1 kHz	10 kHz	50 kHz		
Marantz 7T	12 V	11 V	5.2 V	0.43	Edgy
DB Systems	8 V	8 V	7 V	0.88	Excellent
SP-3A-1	56 V	22 V	5 V	0.09	Excellent

The 7T and the DB measurements seem to confirm Foster's thesis. However, the SP-3A-1 results are somewhat surprising. It has a low overload ratio mostly because of its very high overload capability at middle frequencies.

I think it is logical to compare the input voltages at the clipping points, as it is the preamp's ability to handle the pickup signal that we are investigating. The inputs at overload in the same test are as follows:

Preamp	Frequency Input Voltage at Overload			Subjective Audio Quality
	1 kHz	10 kHz	50 kHz	
Marantz 7T	90 mV	410 mV	770 mV	Edgy
DB Systems	150 mV	760 mV	2500 mV	Excellent
SP-3A-1	1100 mV	2400 mV	1800 mV	Excellent

Here the much higher overload capability of both the SP-3A-1 and the DB are evident. The mid-range overload level of the SP-3A-1 is phenomenal, and perhaps that is related to its mid-range clarity.

When the DB preamp clips at 50 kHz, it is evidently not slew rate limited but is simply protected against input overload by its diode protected inputs. The SP-3A-1 overloads gradually and smoothly.

I hope this will encourage others to compare listening quality with preamp measurements so we will come to understand these mysteries better. -- James A. Mitchell (Tennessee)

(Alvin Foster comments: Since I have never tested the DB or the SP-3A-1, your information is indeed interesting. The Marantz is indeed edgy, but I wonder if the subjective audio quality you ascribe to the SP-3A-1 fills the "good" category I described in the June issue of The Speaker. The "good" category includes the so-called "soft" sound, etc. I also mention that one may prefer the "good" category sound, but I do believe that preamps in the "excellent" category are the most accurate. All excellent preamps sound almost identical.)

And in response Foster suggests that the subjective audio quality of the Audio Research SP-3A-1 might be classified as the "soft" sound, which he categorizes as "good." His thesis that when preamplifier designs surpass some performance characteristic, which is indicated by the white noise test or the ratio of high frequency clipping outputs, they all sound excellent and identical is most interesting. I have not, however, compared a sufficient number of preamps to confirm or deny this thesis.

I have conducted some more carefully controlled listening comparisons between the DB Systems preamp and the SP-3A-1. I have constructed an A-B preamp switching device similar to that described in The BAS Speaker but using remote controlled switching relays instead of a manual switch. I, too, have found it necessary to precisely set the output levels of the two preamps under comparison to exactly the same output level in both channels and also to keep the same capacitive load in the phono cartridge circuits. (I used an ADC-XLM and a Sonus Blue in a Grace 707 arm as the signal source.)

Although I still rate both of these preamps as excellent, I have concluded that the DB unit is more accurate and more detailed, especially in the high end. It is very clean and open in its sound and is exceedingly quiet. One could describe the sound of the SP-3A-1 as being softer or more liquid. It is certainly very musical and on many records (perhaps the less perfect ones) would be preferred.

Incidentally, my SP-3A-1 incorporates the changes in circuitry recommended by Audio Research as of July 9, 1976. This change noticeably improved the detail and accuracy of its sound, so there may be varying degrees of "softness." -- James A. Mitchell (Tennessee)

Another Case for Room Equalization

I've just been doing some experimenting with room environment and sound. So far I've taken my trusty little sound meter into six different living rooms and compared the sound at the speaker versus the sound at the listening position. The meter is calibrated and the records used are CBS's, Stereo Review's (a bummer actually) and Soundcraftsmen's. Each gives pink noise segments throughout the frequency extremes (although Stereo Review's uses only test tones, helpful only to the ear, and then not much except as a backup test. At any rate, if these rooms are typical, the situation is grim. In no case have I measured frequency deviations of less than ± 6 dB. Under such conditions, nothing less than a 10-band equalizer is of much help. And the worst part is that in no case did both speakers measure the same at the listening position.

The speakers tested were FMI J's, FMI 80's, Double Advents, Dyna A-25XL's, Avid 103's and JBL 100's. I know that every audiophile friend of mine tells me I'm crazy to use an equalizer --destroys depth, hurts transient response, adds distortion, etc., but I've only heard benefits. Switching my equalizer in and then out on my J's, for example, convinces everyone who listens.

At the moment, and until I'm convinced otherwise, I don't believe any single addition to a good stereo system can improve it as much as a good equalizer judiciously used.

-- John Puccio (California)

Sound Guard Revisited

I must disagree with Christopher Gupta's review of Sound Guard, the record preservative. My findings are consistent with findings of friends and of the consumer magazines. On the following pieces of music, which have demanding high frequency transients, I could detect no alterations in the frequency response: Band 4, Side 1 of AR ENY-1, "Corpus Christi en Sevilla;" Band 5, Side 2 of AR ENY-1, "Farruca;" "Sonatas for Harpsichord," Domenico Scarlatti, Archiv 2533 072; and the vibraphone solo on "Knots" by Gentle Grant on their album "Octopus." The equipment used in testing was a B&O 3000 turntable with B&O SP-12 cartridge, Sansui AU-7500 integrated amplifier and a pair of B&W DM-4 loudspeakers. One must make sure, though, to burnish the record very hard or else the Sound Guard will clump up, thus producing surface noise.

The best thing about Sound Guard is the price. Although you can pay \$6.99 to Ball Corporation every time you need a refill, don't. DuPont sells the liquid, trichlorofluoroethane in Freon 113 spray form, for 55 to 60 cents per pound. Get in touch with DuPont if you'd like to save money.

-- David Sherwood (New Jersey)

Various Observations

Bryston Pro-3 power amp. This is a very smooth amplifier with sweet high end and low distortion. I compared this amp briefly with an Epicure One and with a Quad 405. Considering the good reviews that I have read on the 405, in particular, I was surprised that the Bryston sounded less distorted than the Quad. In addition, the Bryston sounded less colored than the Epicure. Rather than continue with my subjective impressions, I recommend that members listen to this amplifier for themselves.

Grace 707 tonearm. I had problems with the headshell on my first unit. I couldn't rotate the headshell to the vertical position; the headshell was not flat in one plane -- warped from one corner to its diagonal; and the metal plate on top of the headshell was not properly aligned, so I pried it loose in order to align the cartridge parallel to the arm. In addition the hydraulic lift on the first arm was slow in lowering the tonearm, although it did lower the arm exactly on cue. I recommend that members examine the headshell on their new units and try the lift before mounting their arms nonstop as I did. I am awaiting a replacement for the above defective arm.

Audio Scene (Canada) Magazine. If you don't read it already, I recommend that you start. Every other issue has a good technical article, and some of their equipment reviews are interesting, as they often comment on their impressions of sound reproduction by the equipment being tested. The August issue has an interesting article on measuring headphones (by Dr. Floyd Toole), five headphone reviews, and a special section on the Japanese audio scene, which reveals that Sony is working on a class D amplifier (as if their V-FET weren't enough).

-- Roger Foster (Winnipeg, Canada)

Sonex Loudspeaker

I don't think anything in hi-fi is perfect; but a few things, such as the Sonex, stand out. They're flawed, but good. I heard the Sonex recently in the home of Paul Maury, a gentleman who contributes to Sound Advice. The speakers are rather common in appearance, standing about three feet high and about a foot and a half wide. They were spaced about four feet apart. Related equipment of Paul's included DB Systems, Trevor Lees and (I think) AGI preamps, Quad 405 power amp, another specially built amp which Paul claimed sounds better than the Electro-Research, Fidelity Research cartridge and Sony turntable.

First the bad news on the Sonex. Their size limits the bigness of their sound: they present a quite diminutive sound front when compared with the big sound of Fultons, Maggies, or infinities. The imaging is only moderately good; at times the speakers appear as point sources. Depth illusion, too, is only moderate, but Paul blames this on the recordings. The extreme bottom end is quite weak; response appears to drop off rapidly below 50 Hz. And, finally, the top end is not particularly noteworthy. The treble is quite flat and extended, but somewhat soft and sweet sounding. It lacks the crispness of a good electrostatic.

But, ah, the good news: it is extremely flat throughout most of the audible range, producing a beautifully balanced, natural sound. Its phase coherency is exceptional. I've not heard a better blending of drivers -- certainly the equal of the Quads in this respect and better than the Dahlquists. But the final point is the big one: they possess the best mid-bass to mid-midrange transient response I think I've ever heard from any speaker, bar none. The sensation of realism from recorded drums and bass fiddles is uncanny -- they literally punch you in the stomach.

Yes, but could I live with them as an ultimate speaker? Certainly at \$500 each they beat everything in their price range hands down. I guess it's a matter of which compromises one can most easily live with. For twice the price the Fultons and new Infinity Quantums do more things well than the Sonex and probably present a better over-all illusion of reality. But, then, the Sonex may be the nucleus of a state-of-the-art system of the future.

My main reaction to (owning) the Sonex would be to experiment with them, e. g. , to elevate them about a foot off the floor and angle them slightly inward to produce a bigger, more realistic sound front (who wants to hear a two foot saxophone player -- it destroys the whole illusion). Then I'd triamp, using RTR ESR-6 electrostatic tweeters crossed over at 8-12 kHz and subwoofers at maybe 40-50 Hz. Of course, all that tinkering might screw up the response and phase linearity so delicately engineered into the system. Who knows? But it would be fun playing.

In any event, the Sonex are something to listen for, but they may be hard to find. They are made a few at a time by Ed Long in Oakland. Supposedly there will be only a handful of dealerships nationwide, which is a shame. -- John J. Puccio (California)

Record Reviews--Jazz

Chick Corea - Return to Forever (ECM 1022) and Light as a Feather (Polydor PD 5525). Chick Corea is one of today's major forces in jazz piano. Over the years, he has led or has been a member of some of the most interesting groups in jazz. These two records, from the 1970-1972 period, catch Chick in a Latin-flavored phase of his career (prior to the unfortunate rock he's doing today). The music is fresh, inventive and rhythmic. Especially noteworthy is Stan Clarke on acoustic bass. Singer Flora Purim will either excite you or make you cringe with her unique singing style. (Personally, I find her the most creative singer in jazz today, and her albums on Milestone are worth running out to purchase.) Overall, these records are quite good. The recordings are clean and razor-sharp, with good definition in the mid-bass and extended highs. Flora's voice reproduced on a speaker with a midrange peak will drive you out of the room.

Tenor saxophonist Stanley Turrentine has a powerfully big sound and a driving, straight-ahead style. Sugar (CTI 6005) and Cherry (CTI 6017) are two of his very best. These records feature outstanding performances by such well-known players as trumpeter Freddie Hubbard, bassist Ron Carter (on Sugar), Milt Jackson on vibes, and drummer Billy Cobham (on Cherry). This is excellent, swinging, small-combo jazz. The recordings are very close -miked and very crisp, not unlike the sound of a group close up in a small nightclub.

Straight Life (CTI 6007) finds Freddie Hubbard and an all-star cast in top form as they smoke through two sides of pulsating funky jazz. The album is nicely topped off by a mellow ballad rendition of "Here's That Rainy-Day," in which Hubbard is accompanied only by bassist Ron Carter and guitarist George Benson. Again, as with most early CTI's, the recording is excellent. The instruments have a close-up quality, but the sound is extremely realistic. There is a lot of crisp percussion work on this record. Straight Life has long been one of my favorites.

Alive (Mercury SRM 1 650), by the Chuck Mangione Quartet, is one of the best live recordings in my collection. The group is inspired, loose and obviously having a good time. The music, a blend of light funk and jazz, is thoroughly enjoyable. Unlike the CTI's, Alive has a distant, middle-row sound to it. The recording captures the "live" feel of a large hall as well as any I've heard. The tonal balance is very good, surface noise on my copy is quite low, and the sense of air around the instruments is really something. This record will not overwhelm you with its recorded sound, but in its own slightly understated way, it's one of the most natural sounding records I know of.

Joe Farrell is one of the most talented and versatile reed players in jazz. Moon Gems (CTI 6023) is perhaps the best recorded example of his capabilities. Together with Stan Clarke on bass, Herbie Hancock on piano and drummer Jack DeJohnette, he produces 40 minutes of really first-rate contemporary jazz. Musically, this album comes highly recommended, although for neophyte jazz listeners it may be just a shade on the avant-garde side. The recording is especially interesting, but not for the expected reasons. It seems that on this day CTI engineer Rudy Van Gelder had a heavy hand on the treble equalization. The very uppermost octave is quite exaggerated. Because the midrange is quite smooth, the record is not offensive in the usual presence-peaked manner, but even my Shure V15/III struggles to make it through DeJohnette's cymbals -- a real test.

The wonderful Thad Jones-Mel Lewis band hardly needs an introduction. It is simply the best big band around anywhere. From the standpoint of recorded sound, Potpourri (Philadelphia International KZ 33152) is unquestionably the best album they've made. The separation of individual instruments, low surface noise and sense of depth make for a very lifelike sound. But it's such a treat to listen to this band that you don't really care how good the recording is. My favorite of theirs is Central Park North (Solid State 5S18058). The band really moves!

Azar Lawrence - Summer Solstice (Prestige P-10097). Azar Lawrence is a relative newcomer to jazz. He is just starting to establish a reputation as a fiery, energetic tenor and soprano sax player with strong Coltrane influences in his work. There are five cuts on Summer Solstice, and the album really exemplifies much of what is good in modern jazz. The recording does the music justice. It is well balanced, clean and well defined. Some of the percussive effects, notably a cowbell on the second song of the first side ("Novo Ano"), are strikingly realistic. A voice-trombone-soprano sax unison on the same song is a really difficult test for mid-range distinction. This is a good record, well worth getting.

Weather Report is consistently one of the most innovative bands to be heard anywhere. Led by keyboardist Joe Zawinul and saxophonist Wayne Shorter, the quintet blends superb percussion, tasteful electronics and amazing group interplay to create a sound that is at once rhythmically dynamic and emotionally moving. Yet, despite their talent, they are never guilty of egotistical self-indulgence. Tale Spinnin' (Columbia PC 33417) is their fifth record (they've since come out with another, Black Market, also on Columbia), and like all of their previous efforts, it is excellent. I strongly suggest buying all six Weather Report records and listening to them in chronological order to appreciate the development of this remarkable group. Tale Spinnin' is a simply fantastic recording, and I don't use that adjective lightly. The clarity and impact of the percussion is incredible. If you are looking for a record to show off your system to your friends, this is the one.

-- Steve Feinstein (Massachusetts)

Record Reviews--Classical

Kleiber's DGG recording of Beethoven's Fifth Symphony is the best sounding disc I've ever heard. If this is how DGG is going to produce records in the future, there are a lot of great discs in the offing. The only thing about this record that exceeds the excellence of its sound quality is the splendid performance.

The score is now California 1, Elsewhere 2. I agree with Collins Beagle that there is nothing exceptional about 12 Fantasies for Flute on Columbia-Odyssey. [See below. -- Ed.] The flute comes across as having an unnaturally sharp "air," similar to the kind of sound heard in a public lavatory.

-- David Sherwood (New Jersey)

I disagree about the Kleiber Fifth. Both the disc and the master tape (heard in Boston courtesy of Victor Campos' "Adventures in Sound") exhibit an unnaturally zippy string sound due to excessive high-frequency equalization. Compare it with the superbly natural, well-blended orchestral timbres recorded by Tom Mowrey for DGG in the Ozawa/BSO disc of Ravel's "Ma Mere L'Oye."
-- Peter Mitchell (Massachusetts)

Still More on the PCM Disc

In reference to the comments by Mr. Collins Beagle on p. 16 of the August 1976 issue, I might add more on the PCM disc. Though I have changed my mind as to the merits of the Odyssey PCM disc that I was so impressed with at the time, I feel that Mr. Lipshitz' comments in the July issue of The Speaker are in agreement with my information as to the background noise heard on the disc. I have read that this noise is air conditioner and studio noise as well as traffic noise. As I stated in my original comments, my copy is very quiet between selections, and Mr. Lipshitz seems to agree. I still feel that the record is superb in its musical content.

We will have to wait for more PCM discs to pass final judgment on the method of recording. After all, this was produced by Columbia on their low priced label, and they have informed me that they do not plan to issue any more made with this process, unless they find one that would do well in the present market.
-- Nathan M. Garfinkle (California)

Both Mr. Canby (Audio, October 1976) and Collins Beagle (Speaker, August/September 1976) may have missed the primary reason for producing Odyssey 33200 with the PCM recording technique, admittedly potential overkill for a mere flute.

A solo flute, however, has a very simple (predominantly sinusoidal) sound pattern, and any anomalies of a step-function (digital) process would be easily noticeable if present. Also, the recording had considerable ambient information with occasional low frequency traffic noises as Stanley Lipshitz (Speaker, July) pointed out. The nature of the traffic noises, the ambience, and Mr. Rampal's breathing were about what one would expect in a large quiet church or well-designed classical music studio. This ambience, low frequency, low level sounds and all, was rather perfectly restored with no hiss pumping or loss of definition of the low frequency noises. Again, both are potential areas of problems for a digital process system. Small sounds (details) would be closer to the "size" of the discontinuities of the switching process, as would ambient information. In short, for what I suspect Nippon Columbia was trying to demonstrate, the record is an engineering tour-de-force.

As to Mr. Beagle's dislike of ambient information, I can only disagree and wish that more producers would try to capture a natural ambience rather than making attempts to filter it out, often only to later add false reverb and echo in the mastering process. For natural ambience in orchestral sound, Kertesz's performances of Dvorak symphonies on London are among my favorites.
-- Scott Kent (Massachusetts)

In the Literature

[Sharp-eyed readers may have noticed something amiss in last month's "In the Literature." The error is easy to correct. Just remove the binding staple, flip over page 9 and restaple. The page sequence is now 8-10-9-11, but it does read properly. --Ed.]

Audio, November 1976

*Build the Latest CD-4 Demodulator: JVC kit. (p. 36)

*Lots of tests: Sonab C-500 cassette deck, Spectro Acoustics 210 equalizer, McIntosh C-28 pre-amp (!), Yamaha HP-1 headphones, Heath AA-1640 power amp, Dynaco A-25XL speaker; also White 140 spectrum analyzer and Crown M600 power amp. (p. 52)

AES journal, September 1976

- *Twin-Tone Tape Testing: A new method for evaluating magnetic tape. (p. 542)
- *The Louisiana Superdome Sound System: by J. Jacek Figwer of Bolt Beranek and Newman, Cambridge. (p. 554)
- *Detection of Phase Shifts in Harmonically Related Tones: by researchers at Rensselaer. Conclusions: "The results, both quantitative and qualitative, correlate well with those of previous researchers. This raises the question of its audibility compared to the more familiar forms of distortion." (p. 568)

AES Journal, October 1976

- *Record Warps and System Playback Performance: by Shure engineers. "There exists a great variation in the ability of current high-quality tonearm-phonograph cartridge combinations to successfully track warped records. In this paper the optimum design parameters of the tonearm system are developed which are necessary to cope with record warps without sacrificing other important performance aspects. An extensive survey of warps found on commercially available phonograph records and an analysis of the dynamics of the playback system are included in this study. The analysis extends the concept of trackability into the subaudible, or warp, region." (p. 630)
- *Ultimate Performance of Wide-Range High-Frequency Compression Drivers: Altec engineer introduces two new performance variables: bandwidth ratio (BWR) and bandwidth-efficiency product (BEP). (p. 639)
- *Equivalent Circuit Analysis of Mechano-Acoustic Structures: Reprint of 1954 article by Benjamin Bauer of SQ fame, with accompanying biographical sketch; Bauer's record (pun) is quite impressive. (p. 643)
- *Magnetic-Head Relapping Techniques: Nortronics VP describes techniques for a "professionally relapped head in approximately 5 minutes." (p. 656)
- *Perspectives in Audio Analysis: Changing the Frame of Reference, Part I: Dick Heyser's further theoretical ventures. (p. 660)
- *An advertisement for what must be the ultimate in noncommercial perfectionist audio magazines: The Audio Critic, one year (six issues), \$28 (that's right, \$28.00), first class mail only. This ad also appeared in Audio. It sounds interesting. Address: Box 392, Bronxville, New York 10708. (p. 678)

Audio Handbook from National Semiconductor Corporation, \$4, 177 pages

- *Published in April 1976, this is one volume in their data bookshelf. It looks like a pretty good book. From their brochure: "Thorough explanations, complete with real-world design examples, make clear several audio areas never before available to the general public ... emphasis [is] placed on intuition rather than rigor, favoring the practical over the theoretical." Two other National volumes of interest: Linear Applications, vol. I, II, each \$4. Available from NSC, 2900 Semiconductor Drive, Santa Clara, California 95051 or from their local distributors, e.g., Wilshire Electronics. (Note: If there is enough interest, I may write a list of relevant manufacturer's data sheets, handbooks, application notes, design aids, etc. from Motorola, RCA, Fairchild and the like, for those experimenting with audio. --D. Craig)

Audiogram, April/May 1976

- *A combined issue, the last received to date. (The fate of this magazine is still unknown.) It includes reviews of Sound Guard (favorable), the Rogers LS3/5A loudspeaker, the DB and Berning preamps, the Jensens modified PAT-5 (FET-5 Mk. II), the Formula 4 arm, the Magneplanar Tympani I-C, the ARC SP-3A-1, the RTR ESR 6 and the Paoli 60M, along with a list of recommended components. These folks are especially good about reviewing generally overlooked British equipment.

db, October 1976

- *Don Davis elaborates on his article on biamping in an exchange of letters. (p. 2)
- *Ad for Dynaco Mark VI mono tube amplifier: 120W into 4, 8, 16 ohms; meter reads (adjustable) bias or output; \$425 kit, \$649 assembled; also new dual 10-band equalizer \$249 kit, \$349 assembled. (p. 9)

- . Theory and Practice column on loudspeaker impedance. (p. 10)
- *Broadcast Sound column on routine recording maintenance. (p. 24)
- *The Why and How of Equalization. (p. 39)
- *The New Breed of VU Meters: On LED meters. (P. 44)
- *Test report on UREI Model 200 automatic response plotting system. (p. 46)

Electronic Design, September 27, 1976

- *FCC Swamped with Complaints as Sources of RFI/EMI Increase: Special report on radio interference. (p. 24)
- *Focus on Digital Voltmeters: An excellent article on digital multimeter specifications. DMM prices have fallen much like calculators, digital watches, et al, but before you buy one, read this. Audio is not the only field plagued by specsmanship; you might wonder if DMM manufacturers are also used car salesmen. New ANSI standards to take effect soon may clear up the problems. (p. 62)

FM Guide, October 1976

- *Profiles in High Fidelity: The Superscope and Electro-Voice stories. (p. 19)
- *Feldman Lab Reports on Jennings Research Contrara Vector One, Comsette Corporation "Century" C-90 cassettes, Allison Acoustics Allison:One speaker, Sherwood S-9910 receiver. (p. 36)

The Gramophone, August 1976

- *John Borwick meets Matsushita Electric, and Robert Layton presents the "Quarterly Retrospective" on recordings. Reviews of the HK-2000 cassette deck and the Goldring G 900 Super E cartridge, and the record reviews, which are this magazine's particular glory.

Hi-Fi News and Record Review, any issue 1976

- *Whatever issue you happen to pick up, one item will be of universal interest -- the advertised prices for British audio equipment; B&W, KEF, Celestion, Transcriptors, Quad, and others can be purchased -- if you are willing to put up with the hassle, delay, and repair uncertainty -- at really bargain prices. With the exchange rate near \$1.60 per Pound, a pair of Celestion 66's list for less than \$350; compare this with \$1000 locally and you get an idea of the savings. Hard to find items such as the Formula Four tonearm can be obtained easily. Also of interest are the prices of American and Japanese goods in Britain -- witness AR 11's at \$462 list, including VAT. It may not pay for your trip to the island, but a trip to the foreign news exchange will make for interesting reading.

High Fidelity - Musical America, November 1976

- *Normally ignoring the Boston music scene, Musical America this month deigns to visit the Tanglewood performance of Beethoven's Ninth by Tennstedt. Heaps of praise for Klaus, but "It must be said that the Boston Symphony did not play up to its best." (p. MA-24)

The IMF Newsletter #12

- *The always provocative Irving M. Fried promotes his new "Signature Series" loudspeakers and discusses their design. This issue hints at a turn to ribbon tweeters, perhaps even a revival of the ionic driver. Interesting reading, if only for the personality. Free from I. M. Fried Products, 7616 City Line Avenue, Philadelphia, Pennsylvania 19150.

Popular Electronics, November 1976

- *Stereo Scene: Mods and Modifiers: Jensens Stereo Shop modifications of Dynaco Stereo 400, PAT-5; David Shreve's Rabco modification. (p. 22)
- *Professional vs. Consumer Tape: Larry Zide compares Scotch Classic, 250; Maxell UD-50; Ampex Grand Master 456. (p. 66)
- *Test reports on Spectro Acoustics 210 graphic equalizer, Pickering xv-15/625E phono cartridge; also a different type of audio kit, the Schober Theater Organ kit. (p. 78)

Radio-Electronics, November 1976

- *All About Digital Multimeters, Part I: Good article by Heath engineer. (p. 45)
- *Create Sinewaves Using Digital IC's: Don Lancaster explains digital synthesis of sinewaves. (p. 59)
- *Improved Noise-Reduction for Tapes: Len Feldman analyzes JVC's new Super ANRS. (p. 76)
- *Test reports on Crown IC-150A preamp, AKG P8E phono cartridge. (p. 78)

Sound Advice, Vol. 1, No. 3

*Australian BAS member Trevor Lees makes his American debut with advice on how anyone can achieve the much-sought-after (by SA, at least) quality of "intertransient silence," and with two tube designs, one a pre-preamp, the other a modified Dyna PAS-3, both of which receive raves. Other equipment reviewed includes the Magneplanar Tympani I-C, the new McIntosh MC 2205, the Threshold Class A amp, the Ace Audio, Paragon, DB and AGI 511 preamps, the Supex 900 Super, Fidelity Research FR-1 Mk. III and Sonus Blue and Red Label cartridges, the Luxman PD-121 turntable, the Grace 707 and 940 tonearms and three step-up devices for moving coil cartridges.

Stereo Review, November 1976

- *Equipment tested: Nakamichi 600 cassette deck, with its low distortion and the dirt-cheap Pioneer TX6500 tuner, with fine test results, as far as they went. (p. 35)
- *Using FM Hiss: A dirt-cheap test instrument. (p. 74)

Wireless World, September 1976

- *Projection Television: The first of a two-part review of large-screen television products. (p. 47)
- *Surround-sound decoders, Part 4. (p. 53)
- *Letter from America: Review of the CES show; mentions the Advent receiver, Nakamichi power amplifier, Elcaset, dbx, and several loudspeakers, including the British exports. (p. 67)
- *Magnetic Pickup Preamplifier: Uses a dual 747 IC with a flat-response buffer-input and an RIAA-equalized output stage. (p. 81)

Wireless World, October 1976

- *Digital Filter Design: "Programming a microprocessor to act as a digital filter." (p. 47)
- *Variable Pre-emphasis in FM Broadcasting: A Dolby-like method of noise reduction being tested in Britain. (p. 65)
- *Projection Television, II. (p. 67)
 - Dana Craig (with contributions from Harry Zwicker and Michael Riggs)

October 17th BAS Meeting

Business Meeting

The first of two October BAS meetings convened in the Conference Auditorium of BU's Sherman Union. The treasurer's report was not yet available and was therefore waived. Volunteers were sought for a Speaker-stuffing party October 30. Members are reminded that with a sufficient number attending these occasions can be reasonably brief and quite enjoyable.

Scott Kent offered his usual piano and harpsichord records for sale, along with a new recording of the Methuen organ, which he said is excellent. Al Foster will be taking orders shortly for the new Sheffield disc of the Harry James band and reminded those interested to order early to get copies made early in the life of the stampers. He also had some Maxell tape for sale. He reports that out-of-state membership renewals are pouring in; people don't seem to be objecting to the dues increase.

Dr. Brian Leeming apologized to members for his part in the delays in imported record shipments and added that he is also having trouble getting Maildisc in Britain to ship promptly.

Dick Lewis brought to the meeting one of the electret cardioid microphones he has for sale. Manufactured for an American firm (CMI) by AKG, they are bargain-priced at \$62 to \$67 per unit (depending on impedance).

Finally, Jim Brinton announced the second October meeting, to be held a week later on the 24th, in which TIM maven Dr. Matti Ojala has agreed to subject himself to "the piranhas of the Boston Audio Society." That meeting, advertised in advance as "bare-knuckles engineering," is chronicled elsewhere in this issue.

Meeting Feature - Harold Beveridge

Mr. Beveridge is an engineer, primarily a microwave specialist, with a long-standing interest in audio. He "retired" about three years ago to begin manufacturing a new speaker system, which is intended as a state-of-the-art device. The system, which sells for \$4000 per pair, consists of a vacuum-tube amplifier direct-coupled to a full-range electrostatic transducer, whose rearward radiation is contained in its cabinet and whose forward radiation is directed through an acoustic lens which ends in a narrow slot designed to produce a cylindrical wavefront six feet high (see Fig. 1). The greater part of Mr. Beveridge's talk consisted of theoretical support for the design of the lens/cabinet combination, as follows.

The most important part of the performance of a speaker system is determined by the first arrival (at the listening position) of transient information. This first arrival should have flat frequency response, should be as evenly distributed as possible throughout the room and should be unencumbered by subsequent arrivals (reflections) less than two milliseconds later than the original. Such early reflections are a little-recognized but important source of unpleasantness because, he stated, the brain has trouble deciding whether the reflection is a separate sound or part of the original. More specifically, the brain is unable to localize the first arrival properly if the path length of the reflection is less than about two feet longer than the direct path, and subjective tests indicate that a difference of four milliseconds, or about four feet in path length, is even better. [This may help explain the nine-foot-wide solo violins so well-known to the owners of Bose or similar designs. --EBM.] Furthermore, these reflections set up interference patterns which degrade frequency response as well.

If we consider one popular concept of the ideal loudspeaker, the pulsating sphere, and locate it in the usual place near a wall, the wall behind it will create an acoustical mirror-image, which will cause all of the aforementioned problems. If we place it near a corner, there will be three such virtual images, all interacting with the original and each other (think of what you see in two mirrors meeting at right angles). Besides these, there will be two more troublesome virtual images, one reflected from the ceiling and one from the floor. Clearly, this is not the way to go.

Now, if instead we imagine a pulsating cylinder the height of the room, we immediately see

some improvements. The cylinder radiates almost totally horizontally, so floor and ceiling reflections are largely eliminated. And because the cylinder is in effect a line source, rather than a point source, the intensity of the radiation from it falls off only 3 dB for each doubling of distance from the transducer, instead of 6 dB for the pulsating sphere. Therefore sound pressure levels will be more evenly distributed throughout the room. [Remember that we are discussing only the first arrival from the transducer and a few early reflections; the reverberant field in the room is not being considered. There are, of course, differences of opinion on the validity of this assumption. --EBM.] The other virtual images, created by reflections from the walls, would remain, however.

All these difficulties would be laid to rest if we could place a pulsating half-cylinder flush with the wall and far from the corners. The sound would radiate evenly through 180 degrees and would be evenly distributed from floor to ceiling. The frequency response of first arrival information would be the same everywhere in the room, and there would be no bothersome early reflections. The Beveridge speaker system approximates this model when the speakers are placed on the side walls of the room, two feet or more out from the corners. The ideal listening position is back at a point where the angle between the speakers subtends about sixty degrees, but due to the wide dispersion and the improved intensity gradient of the sound field, listening position is far less critical than with other types of speakers. Absorption of the backward-directed sound from the electrostatic screen eliminates reflection problems and rear-wave cancellation at low frequencies that plague other electrostatic designs, and the use of a single transducer for the entire frequency range eliminates the phase-coherency imperfections that accompany multiple transducers and crossover networks.

During the question period Beveridge provided some interesting information about the construction details and the design history of his devices. The transducer, which with its power amp was basically designed in the late 1950's and early 60's, has several interesting features. The diaphragm is aluminized mylar, whose good conductivity helps eliminate problems caused by static electricity in other electrostatics. The electrodes are 250 thousandths of an inch thick, with a conductive coating on the outside, and are made of epoxy with barium titanate mixed in. This material has a dielectric constant of 100, so that even though the air gap between the diaphragm and the inner side of the electrode (80 mils) is less than 25% of the total distance to the outer conductor, almost 97% of the voltage drop between the two occurs across the air gap, right where it's wanted. And because the inner side of the electrode is non-conductive, if the diaphragm is overdriven and touches it no harm is done to the system. [That the speakers are almost literally bomb-proof was demonstrated to me the following week when someone played a recording of a certain well-known orchestral piece with cannon accompaniment; the diaphragms were absolutely plastered against the electrodes. The distortion was extreme but the speakers were undamaged. --EBM.] The speakers will generate sound-level-meter readings of 100 dB out in the room with orchestral material, and according to Beveridge their good articulation makes very high listening levels less necessary.

The acoustic lens material is made of paper, cardboard, and enamel paint. The cabinets are 1/4" plywood with ribbing and bracing for added strength; 3/4" wood in a cabinet that large would have been too heavy, he says. In response to questions about making a smaller version or issuing the speaker as a kit for people who can't afford it, Beveridge said that if BAS members would all buy lots of speakers, the resulting large production runs would allow the price to drop! (The line forms on the right, folks.) For those interested, the speakers may be heard at Natural Sound, 401 Worcester Road (Route 9), Framingham, telephone (617) 879-3556.

A Brief Listening Impression of the Beveridge Loudspeaker

I auditioned the Beveridges for about two hours at Natural Sound the Sunday after the meeting. My overall impression was very favorable. The speakers sound very smooth from about 150 Hz up to 15 kHz, the present upper limit of my hearing, alas. There is a slight "hooded" effect on male voices (compared as best I can remember with Koss ESP-9 headphones), but less than I hear on KLH Nines or Infinity Servo-Static I's (I haven't heard the IA's). Stereo imaging on large ensembles was pleasant and airy but somewhat vague. Both the overall sound and the stereo image were remarkably consistent throughout the room: Defects in the program material are revealed but not glaringly spotlighted. On a 15 ips, two-track, B-Dolby master tape of a choral concert, they sounded absolutely gorgeous and beautifully transparent. A BSO broadcast of Copeland's Ap-

palachian Spring done in the same format gave them some difficulty when played very loud. There was some distortion from the massed brass sforzandi, and the bass drum lacked authority compared to the best I have heard. (Beveridge says that the screens in their cabinets resonate at 80 Hz until the lens is installed, after which the improved coupling to the room effectively brings the resonance down to around 40 Hz.) In my opinion, real bass freaks are still going to want to add a sub-woofer, and die-hard dynamic range freaks should still stay with cones. If the Larger Advent is used as a standard for overall frequency balance, this speaker is an example of the bright-high-end school of design, a family of speaker I generally do not prefer. Nevertheless, I found them extremely pleasant to listen to and not in the least fatiguing. This to me speaks well for their smoothness in the upper mid-range and high end. All in all, it is a successful design, worthy of a serious audition by anyone seeking the current state of the art. -- Brad Meyer

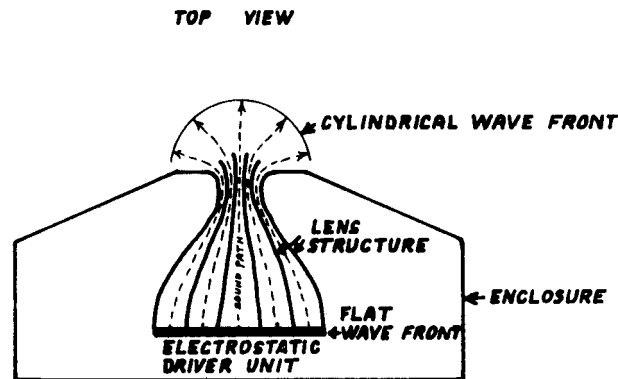


Fig. 1. Beveridge Loudspeaker System

(Drawing Courtesy of Harold Beveridge, Inc.)

October 24th BAS Meeting

Business Meeting

The second October meeting was held at Boston University. President Jim Brinton mentioned that Bob Tucker of Dynaco had called regarding the results of the Tuner Clinic. Dynaco has offered to fix defective Dyna FM-5 and AF-6 tuners at no charge. Al Foster said that Michael Riggs had returned his unit twice and was still not too pleased with the results but we now gather that Dyna has a handle on the problem and members with truly defective units should write directly to Tucker.

Peter Schnell announced that his loudspeaker would be available at a special BAS discount of about 15% at Natural Sound in Framingham. The offer is good through at least December 15, 1976, and Schnell will attempt to extend the deadline for those members who receive late news via this publication.

Al Southwick, whose MacIntosh MR71 was the subject of a Tuner Clinic report, has discussed the revamping of that unit including adding a phase-lock loop with MacIntosh. Scott Kent said that he had revised MR67 and MR71 tuners with both PLL and filter modifications.

Southwick also mentioned that special two-record sets of E. Power Biggs at the Methuen organ were available at the hall in Methuen, autographed by Mr. Biggs at \$5.00 which sounds like a bargain. Dick Lewis indicated that the AKG microphone offer is still open. These microphones are approximately equivalent to the AKG CE5. Dick Goldwater announced that he would like to have an offer on his eight KLH-9 speakers and would find something in the vicinity of \$4,000 entertaining. Scott Kent would like to purchase a used Pioneer TX7500.

Jim Topali had zero response in his quest for volunteers to handle refreshments for the next

and subsequent meetings. This, coupled with a generally cavalier attitude toward the 25¢ charge for those supplied at this meeting may result in a barren intermission at subsequent meetings.

Ira Leonard and Scott Kent had their regular assortment of records for sale.

Meeting Feature

Matti Otala, Mark Davis and panel on the why's and where-is-its of TIM. The technical tyros of the BAS were treated to an exceptionally rich serving of presentation and discussion on the elusive transient intermodulation distortion. The session was broken into three parts. Dr. Otala presented the illustrated paper prepared for presentation later that month at the 55th AES Convention in New York. This was followed by a detailed analysis and refutation by Mark Davis of Dr. Otala's 1973 paper on TIM. After a very short intermission, the discussion finished with a panel made up of Dr. Otala, Mark Davis, Rene Jaeger, Chief Engineer of dbx, Scott Kent of BKM Associates and Dick Burwen, President of Burwen Labs.

Because of the highly technical content of the entire presentation and discussion, we would like to offer the following layman language condensation for the benefit of non-engineering members. It is fair to state that in raw form, Dr. Otala's papers are open to misinterpretation by even the most trained.

(Because of the interesting and technical nature of the material covered in this meeting, the usual meeting summary approach may be insufficient. Also, much of the discussion was free-form and therefore difficult to capture on paper. For this reason the BAS has commissioned a recording of the meeting; it will be available on request to "TIM Tape" c/o Box 7 for \$15. The total time of the recording which is offered on C-90 cassettes is somewhat more than four hours. Members incapable of using the cassette format must make their own arrangements and should enclose a self-addressed, stamped envelope with their request. --JBB.)

In essence, Dr. Otala claims that all amplifiers with substantial amounts of overall feedback (greater than 40 dB) are susceptible to and many can be shown to exhibit transient intermodulation distortion. The basic mechanism is short-term overload of the second stage by the first stage output due to the time lag of the feedback signal which is derived from the final output. Under steady state conditions the input signal to the second stage is quite small being the sum of the output signal from the first stage and the feedback signal from the output which is inverted. If the original signal changes quite rapidly as with a transient, one can conceive that due to lower high-frequency response in the second and last stages, the final output will lag slightly behind the first stage output signal and the feedback will arrive late. Instantaneously, this will result in an abnormally high voltage at the input to the second stage. It is Dr. Otala's contention that this voltage will temporarily overload the second stage. In extreme cases, the second stage will saturate, clip and take a finite time to recover. This is transient intermodulation distortion. He further contends that TIM may occur before the second stage actually saturates but as it is driven into a non-linear operating region.

In an amplifier with no feedback this cannot occur since the second stage is designed to operate from the full output of the first stage.

Dr. Otala claims that sensitive listeners are able to detect TIM distortion at levels of 0.2 to 0.3 per cent. A 0.5 to 1 μ s of clipping results in TIM levels of 1 to 3 per cent. It is his further contention that significant amounts of TIM can exist and be detected audibly in amplifiers with excellent harmonic and intermodulation distortion specifications. Although there is a direct inverse relationship between TIM and slew rate limiting, he claims that TIM is noticeable at as low as one tenth of a circuit's slew rate limit.

The presentation of Dr. Otala's AES paper was preceded and ended with a demonstration tape recording of simulated TIM distortion including a female vocal soloist, a folk dance played primarily on stringed instruments and a Mozart song with a baritone and full orchestra. While the distortion was clearly heard by all, in answer to a subsequent question from Dick Burwen, Dr. Otala indicated that the average induced distortion in the "bad" sections was 26 to 35 per cent at average levels and substantially higher on the peaks. Dr. Otala then attempted to describe sonic effects of TIM in that low levels of TIM would impart a throaty sound to a female vocalist, would

make cymbals sound hard and shattery and a mixed voice chorus would contain paper tearing sounds much like that caused by tracking errors with a phonograph cartridge.

Otala's new paper, titled "Method for Measuring Transient Intermodulation Distortion," describes a test method and results with various amplifiers. The test signal consists of a 15 kHz sine wave superimposed on a 3.18 kHz square wave which has passed through a 30 or 100 kHz low pass filter. The output is observed on a spectrum analyzer. The amplitude of all the intermodulation products is measured and summed. The static intermodulation is then measured by replacing the square wave with a triangle of identical peak to peak amplitude. That can then be subtracted from the first, resulting in the TIM distortion figure. The results on operational amplifiers 709, 739, 741, 301 and 1456 showed high TIM at low output voltages. His conclusion being that the 741 should be completely excluded from audio circuits and the others used only with gains greater than 20 dB and less than a few volts output. High slew rate types such as the 318 and 2505 showed very little or no TIM.

Eight power amplifiers were tested. These were Sansui 771, Marantz 2270, Harmon Kardon 230A, Sony TA5650, Pioneer SX535, Tandberg TR2075, Salora 2000 and Dux TA4000. Test results showed TIM in all amplifiers. In the worst case showing up at as low as 10 watts in an amplifier rated for 70 watts under conditions of no low pass filtering of the input square wave on the premise that a properly designed preamp should accomplish this limiting. With 10 kHz filtering of the square wave most of the amplifiers showed barely discernable TIM with total distortion running just slightly over the normal intermodulation distortion level. He went on to compare the results of the proposed TIM test with other standard distortion tests presenting results only for the operational amplifiers which seemed to bear out his contention that other methods did not detect this type of distortion. His conclusion was "many amplifiers having excellent THD and SMPTE-IM data show high values of distortion as measured with the proposed method."

Mark Davis of MIT then delivered an illustrated dissertation on the results of his analysis and duplicated experimental results of Dr. Otala's 1973 paper on TIM.

Dr. Otala is quite sensitive to arguments against statements erroneously attributed to him in some of the more spirited discussions of TIM. Mark Davis was especially careful to list the statements with which he disagreed and these were taken directly from Dr. Otala's paper. In summary, Otala's statements are that transient testing will reveal distortions not seen in sine wave testing; that an amplifier which passes sine wave distortion tests may have audible distortion with music signals due to transient effects; that the higher the amount of feedback, the greater will be the TIM distortion so that it is desirable to build amplifiers with small amounts of feedback. Further, the open loop band width of each stage in an amplifier should be greater than the closed loop band width of the previous stage and that transient effects exclude the use of operational amplifiers, such as the 741 in audio amplifiers.

Key points made in Davis' discussion were that although the second stage input signal was instantaneously larger than its final value, due to the lag of the feedback signal, it could never exceed the original signal level contrary to the impression given by the curves in Dr. Otala's 1973 paper which were normalized so as to have a final value of 1. Thus in Dr. Otala's curves of amplifiers with high feedback ratios, the input signals apparently instantaneously far exceeded the available supply voltages. In reality, Davis pointed out, the final value is not 1 but approximately $1/B$ where B is the feedback ratio. Therefore, the peak will never be higher than 1 diminishing to the final very low level due to the feedback.

Clipping can only occur if the original signal exceeds the supply voltages or if it is large and fast enough to cause slew rate limiting. Otala's oscilloscope photographs of the wave forms at various points through the test amplifier show identical input and output wave forms even when non-linearities showed up at the intermediate stages, indicating a lack of distortion effects even though the circuits were supposedly exhibiting TIM distortion.

In order to demonstrate the TIM effect, the paper notes, diode clamps were introduced at the input to the second stage to artificially induce the clipping effect. The final scope photo illustrating the distortion of a sine wave by an injected square wave was apparently achieved under unrealistic conditions. Davis' analysis indicated that the sine wave was of the order of 250 kHz to 1.3 MHz, that the preamp stage had treble boost which increased its band width to 400 kHz, that the

power amplifier was already slew rate limited at the high sinusoidal frequency and that diode clamps still had to be added to generate the effect.

Davis briefly commented on Dr. Otala's new AES paper and specifically the statement that "modern amplifiers often measuring under 0.01 per cent total harmonic distortion at 1 kHz or below 0.1 per cent intermodulation distortion as measured with the SMPTE method, may sound directly unacceptable." indicating that there was no supporting data.

Davis' contention is that any amplifier may be driven into slew rate distortion if the input signal is large enough in amplitude and high enough in frequency but, even when properly applied, Dr. Otala's transient test far exceeds the possibilities of any music signal. In an attempt to prove this Davis (and Alvin Foster) went so far as to scratch deep radial cuts on a record surface and did not find significant signal levels above 20 kHz. Both FM transmissions and tape sources cut off not much higher than 15 kHz and normal music signals contain the highest frequencies at substantially lower than average levels. Dr. Otala's test transient is a full scale square wave passed through a 30 kHz filter. This tremendously exceeds normal or even abnormal signal conditions and may drive very fine performing amplifiers into slew rate limiting which is a previously well understood phenomenon.

Panel Discussion

Dr. Otala led off the panel discussion by stressing the point that the saturation which caused TIM did not occur at the input of the second stage but at its output and that discussions about the actual overshoot levels at the input were unimportant. He further claimed that Davis' references were to a 1969 paper which was the first published on TIM wherein many of the levels were speculated rather than measured. He then presented Davis with a copy of the later work which we suspect may have removed several points of contention. Otala claimed that the lack of distortion in the output photos from that early paper were correct since there was no intermediate stage clipping shown. He essentially agreed that sine waves could be used for analysis but that square waves were simpler to interpret. He again stressed that clipping occurred at the output of the second stage and that the peak amplitude was proportional to the feedback.

Davis agreed that it was good practice to have compensation at the earliest possible stage and that square wave testing was ok but should be band pass limited at 10 kHz but not 30.

Rene Jaeger, after reading Otala's earlier paper, did a number of tests with integrated circuit amplifiers. A compensated 301 type amplifier band limited to 24 kHz exhibited one -half per cent distortion but a newer type 4558 was 10 to 20 dB lower in distortion. Moreover, specific vendors' 4558 amplifiers were 10 to 30 dB better than that, or completely inaudible. His further contention was that a phonograph cartridge constituted a 20 kHz low pass filter rolling off at 12 dB per octave. His testing indicated the absolute worst preamp level output when the needle was dropped onto the record was 0.5 volts peak-to-peak and less than 0.5 volts per μ s. His contention was that the problem would be practically non-existent at the preamplifier level where integrated circuit amplifiers are used. He then tested two high power amplifiers. The Ampzilla with a 15 V/ μ s slew rate and a Phase Linear 700 with an 8 V/ μ s slew rate. With a 60 kHz band limit on the square wave, the Ampzilla's distortion was down more than 80 dB with 80 volts peak-to-peak output. A capacitive load of 1 mfd raised the distortion to -60 dB. Note that while the capacitor was intended to simulate the loading of a physical crossover network, it would, in practice, have some resistance in series with it which we suspect would significantly lower its effect on the distortion. The Phase Linear amplifier exhibited extremely high distortion with any capacitive load. Jaeger's conclusion is that TIM is almost entirely a power amplifier problem. Further, he would like to see some psycho-acoustic data comparing the short time bursts of TIM distortion to normal intermodulation.

Scott Kent discussed a number of low level listening tests which indicated to him that TIM did not show up below a certain power level. He could only hear a difference with a version of the Harmon Kardon Citation 12 which he purposely slowed down from a 2 μ s to 10 μ s rise time and then only occasionally. The Leach "TIM-free" amplifier was the only one he found capable of driving a capacitive load without bad distortion. The fastest natural rise time he could find was measured directly at a capacitor microphone, picking up the highest notes on a harpsichord when placed within one foot of the strings. This was of the magnitude of 20 to 40 μ s. The fastest signal

he could measure from tape had a 30 μ s rise time.

Dick Burwen stated that he has over 2000 op amps in his system and is sure he could hear the problem if it existed at that level when he switched in and out 25 or more of them at a time. He felt that very high slew rate input signals were only achieved with abnormally close miking and a lot of high frequency emphasis. From his own experience he uses IC op amps at very low gains therefore with a lot of feedback and contends that the amount of feedback is not as important as how it is applied.

A statement from the floor that at least one reviewer uses 10 to 20 kHz square waves to test amplifiers brought more spirited discussion on its pros and cons. Otala claims that the ideal amplifier should be able to take DC to infinite frequencies without harm. Scott Kent suggested that applying a square wave and increasing its level until the output rise time increased was a good way to check for internal state limiting. The better amplifiers should retain a constant rise time up to full output.

Otala restated that TIM was observable at as low as one-tenth of the specified slew rate limit since slew rate limiting occurred gradually. Davis disagreed, indicating his tests showed under .01 per cent TIM at 90 per cent of the slew rate limit with the 741 amplifier. Scott Kent suggested that some discrepancy in results might be due to the big difference in actual slew rate between the 741 amplifiers of different manufacturers.

In answer to a question, Otala indicated a possible relationship between TIM and listener fatigue.

In answer to a request to identify good and bad measuring amplifiers, Otala listed all of the types that he tested but refused to discriminate.

Al Foster stated that the question seemed to boil down to whether Otala's test was realistic. Otala indicated that the rise time of his test square wave was 10 μ s after passing through the 30 kHz filter. Foster asked Davis if 10 μ s were a realistic rise time. Davis replied the level also would have to be known in order to determine slew rate. He went on to say that RIAA limits for recording would result in a 50 cm/sec acceleration at the outer edge of a record which would result in a rise time of 6 μ s but with 30 dB gain in the preamp, would only give a 0.032 V/ μ s slew rate. Foster then asked Otala if he accepted Davis' premise on velocity and slew rate. Otala thought he might be off by a factor of 2 or 3 which Foster pointed out still wouldn't approach the limit for a 741 op amp.

Jim Brinton announced the results of a discussion with Rene Jaeger to the effect that a particular cutter could put 105 cm/sec onto a disc. Otala claimed that the playback limit is controlled by the compliance of the vinyl sidewall and Davis discussed slope and curvature overload which caused gross distortion at that high a cutting level and wondered about the necessity for an amplifier to reproduce that distortion accurately. A voice of reason from the crowd pointed out that Shure figures that 30 cm/sec is the best tracking they can guarantee.

Otala reflected that he was trying to measure things that could be heard and that discussions of slew rate limiting or power bandwidth limiting were not themselves audible. There was more general discussion of wave forms, actual and contrived.

Al Foster asked for a clarification on the relationship of TIM and slew rate limiting. Otala indicated a direct relationship but up to a factor of 10 to 1 in the operating point at which they become effective. Foster stated that he understood a basic disagreement between Otala and Davis was whether clipping went up with increasing or decreasing amounts of feedback. Davis stated more feedback was better and a 741 with a gain of 20 had more TIM than at a gain of 1. Otala claimed the opposite was true. In the ensuing spirited discussion it appeared the basic reason for their disagreement was Otala's assumption of constant output voltage versus Davis' assumption of constant input voltage. Davis agreed that if the output voltage is held constant and the feedback is increased, the amplifier is more susceptible to TIM. Otala further indicated that if the input were held constant and the gain changed the compensation should also change. Davis disagreed vehemently and at that point, the meeting was adjourned.

-- Joel Cohen

The Boston Audio Society does not endorse or criticize products, dealers, or services. Opinions expressed herein reflect the views of their authors and are for the information of the members.

A Survey of Speaker Drivers

Jim Nichol

Rolling your own speakers can be rewarding. Unfortunately, aside from the usual advertising hype and puffery, very little information is available on speaker drivers. This article will attempt to remedy the situation by surveying the frequency response characteristics of some of the tweeters and midrange drivers available on the market.

Seven Turkeys in Tweeter's Clothing

This first batch of tweeters is composed of basket cases available at local Lafayette, Olson, Radio Shack, and You-Do-It Stores.

The upper curve on each tweeter plot is the on-axis response; the lower, dotted curve is the response taken 45 degrees off-axis to give some idea of dispersion. The vertical scale is 1 dB per division.

The Olson S-162 multi-cell horn tweeter is particularly bad, especially considering that it costs \$18. The first one I bought was missing the drive unit—just had a horn. The second unit made a clattering sound whenever driven to even moderate levels. Unscrewing the horn from the driver revealed a coil of wire dangling from the speaker terminals. If you use this speaker (and I hope nobody does) mount it with the speaker terminals up. Immediately in front of the moving coil was a small, bullet-shaped piece of plastic, which is probably what was rattling around. Could it be a phasing plug? That's doubtful, as a phasing plug is usually used in conjunction with a diaphragm, and the designer of this thing doesn't believe in diaphragms.

Tweeters

We can now move to a more detailed look at some good tweeters. In addition to the on-axis and 45 degrees off-axis sine wave response curves, I have included tone burst ringing data. In general, it appears that good speakers have good tone burst characteristics and that bad speakers have innumerable tone burst envelope distortions at different frequencies, so I have not attempted to catalog all the various tone burst distortions involved. I have noted instances of ringing excited by tone bursts.

I have also included data on a particularly vicious test of speaker drivers, which I discovered accidentally, namely triangle waves. Upon completing a tone generator using an 8038 function generator IC, I found that most speakers exhibited gross amounts of distortion, visible on an oscilloscope, when tested with the generator. I tracked this down to a certain amount of "triangle distortion" present in the sine waves generated by the function generator (the 8038 function generator first generates a triangle wave, then reshapes it into a sine wave). Ordinarily I would have disregarded the behavior of speakers in response to triangle waves (let's all agree that reproduc-

tion of triangle waves is not a requirement for music reproduction, unless one listens to synthesizers), but I found that a few speakers do reproduce triangle waves reasonably well or merely smooth them out to sine-like waveforms. This is probably a severe test of cone breakup and perhaps of other factors. Again, it is interesting to note that the results of all three tests—frequency response, tone burst, and triangle wave test—are very good for good speakers and are inevitably poor for undistinguished speakers.

In my references to speaker distributors, note the following addresses: McGee Radio, 1901 McGee St., Kansas City, MO 64108; Transline Sound, 452 Poplar, Lyandotte, MI 48192; Custom Sound, 8460 Marsh Road, Algonac, MI 48001; Speakerlab, P.O. Box 15780, Seattle, WA 98115.

I have found McGee to be functionally illiterate, but they have good prices on speakers, some good, some very strange. Transline and Custom Sound have very reasonable prices and are very helpful, writing long, handwritten letters filled with information and crossover advice. Also, Custom Sound offers a money-back guarantee, which they do honor. If you don't like the speaker you ordered, send it back with no solder on the terminals and no damage to the unit and get a full refund. All three companies are recommended. Speakerlab, on the other hand, has very high prices, and, contrary to their advertising, they are not the good guys in white hats, they are not the saviors of the world, and they don't know all there is to know about speakers.

SEAS speakers are distributed by Sennheiser, 10 West 37th St., New York, NY 10018, also recommended.

The first three tweeters are horns. The Electro-Voice T-35 is available from Custom Sound for \$30 (model H1) and from Speakerlab for \$36 (model HT-3500). This unit had ringing throughout its range. The second horn, a Motorola piezoelectric tweeter, is available from McGee for \$8 (the super horn) and from Speakerlab for \$12 (the PT2500s). The third horn, again piezoelectric, is available from McGee for \$10, as model PET-2X5.

The next two tweeters are Philips units. The AD160/T8 is available from McGee for \$8 and from Speakerlab for \$9. The AD616/T8 is supposedly a high-power version of the 160 and is available from McGee for \$9 and from Custom Sound for \$10.

The MG-T6, a 1-inch dome tweeter, looks pretty good, especially when one considers that it costs only \$5 at McGee. This is the only good, inexpensive tweeter around.

The Peerless K010DT is available at Transline for \$11, at Speakerlab for \$16 (model DT101), and at You-Do-It Electronics for \$14 (model NTT).

The KEF T-27 dome tweeter is available from Transline Sound for \$18.

SEAS has two dome tweeters: the 1½-inch 87H for \$19 and the 1-inch 86H for \$17. The 87H has particularly low triangle distortion, but poor high-frequency dispersion. Curiously, the smaller dome has better low-frequency response.

The Long L15E 2-inch dome is a 4-ohm unit available from McGee for \$10 and from Speakerlab for \$18 (model DT150).

Midrange Drivers

The first midrange is a fugitive from part one. Olson strikes again with the SP-037, a 2-inch dome unit for \$11. The off-axis response appears to follow the on-axis response closely making it all the sadder that the on-axis response is so bad. It's also available from Lafayette (21F29096), two for \$22.

The CTS 1077 (actually a Heppner unit for the duration of the strike at CTS) is an especially peaky driver available from McGee for \$8. This unit has a strange triangle wave response below 2 kHz—not quite outright distortion, but small ripples all along the wave. Severe triangle distortion occurs above 2 kHz.

The next three drivers are horns. The first is the midrange used in folded horn speaker systems. I am not sure if this is the exact unit used in Klipschorns, but it is available at Custom Sound for \$85 and at Speakerlab for \$95. The second is the home-built wooden horn with the SEAS driver described in the 1/74 issue of Audio Amateur. I have tried the horn with other drivers, with similar results. The third is a metal diffraction horn available from Custom Sound for \$47 (model MHA) and from Speakerlab for \$49 (model HM700). Custom Sound also has a model MHB with a different (1823M) driver for \$67.

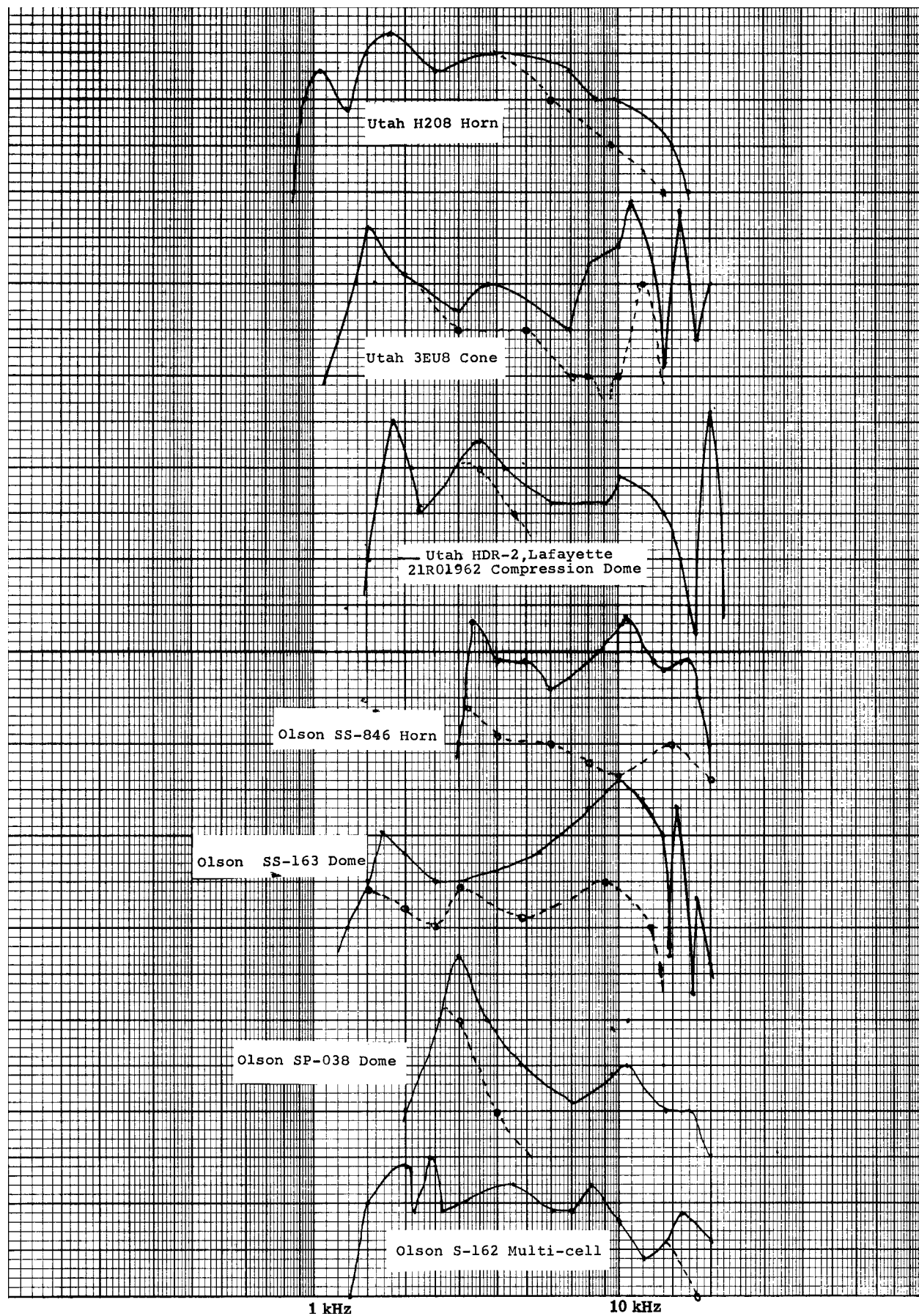
The KEF B110 is a 5-inch unit with a very high compliance plastic cone, available from Transline Sound for \$26 and from Sound Advice, 536 State Road, Emmaus, PA 18049, for \$51 per pair. I ran the curves shown in a small isolation chamber, 6 inches in diameter by 4¼ inches deep. In larger enclosures a peak occurs about 1 kHz, which can be pulled down with a 2-mH, 16-series resonance circuit across the speaker terminals, described in a KEF pamphlet available from Sound Advice.

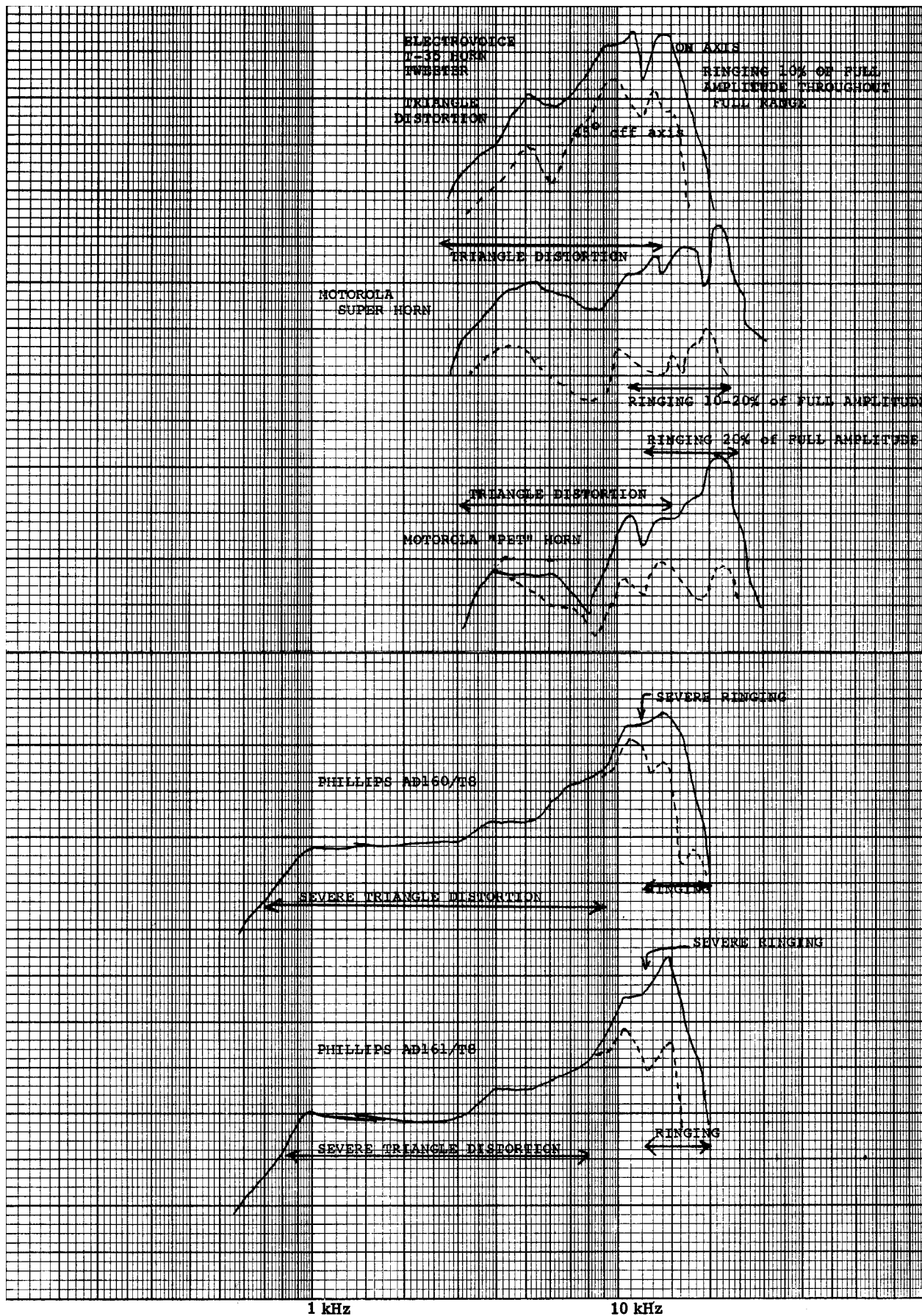
Next are five Philips units. The first is the 5060/M8, which doesn't seem to be available now. It was used in the Rectilinear III. The second is the 5061/M8, which is a high compliance, 5-inch unit stamped "Made in Holland" and available from McGee for \$6. This is the midrange design Rectilinear is currently using in the Model III, though they actually use a similar unit made in Belgium, which appears to be better made and to have better bass response. A better alternative would be the Philips 5060/SQ8, which is available at McGee for \$12. This speaker appears to be a very good value for the money and illustrates how a good speaker performs well on frequency response, tone burst, and triangle wave tests. The last Philips unit is the 210/SQ8, a 2-inch dome available from McGee for \$17 and from Custom Sound for \$19.

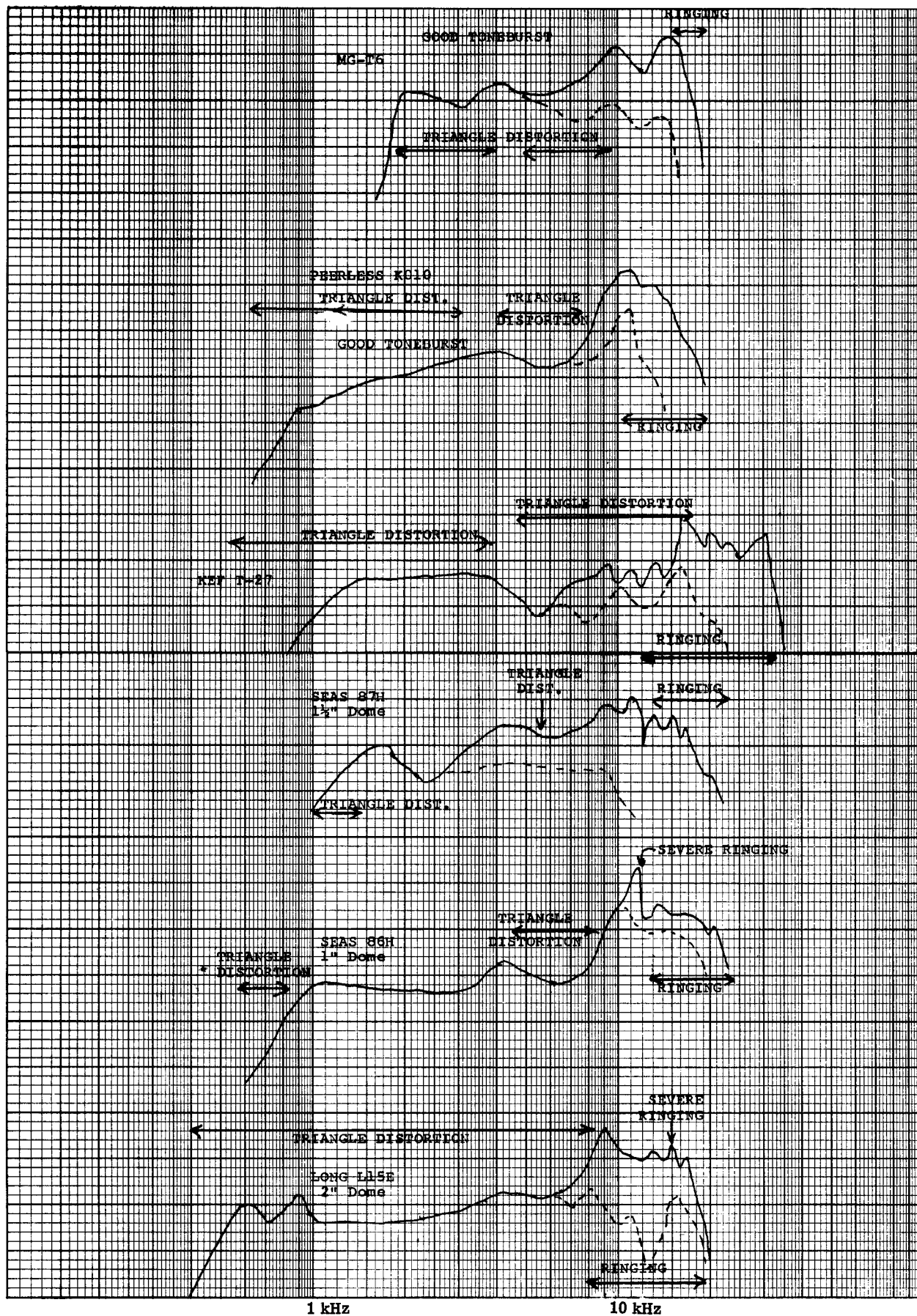
The Peerless K040 is another good unit, available from Custom Sound for \$16 (model 5B), from Transline Sound for \$15, and from Speakerlab for \$18 (model M600B).

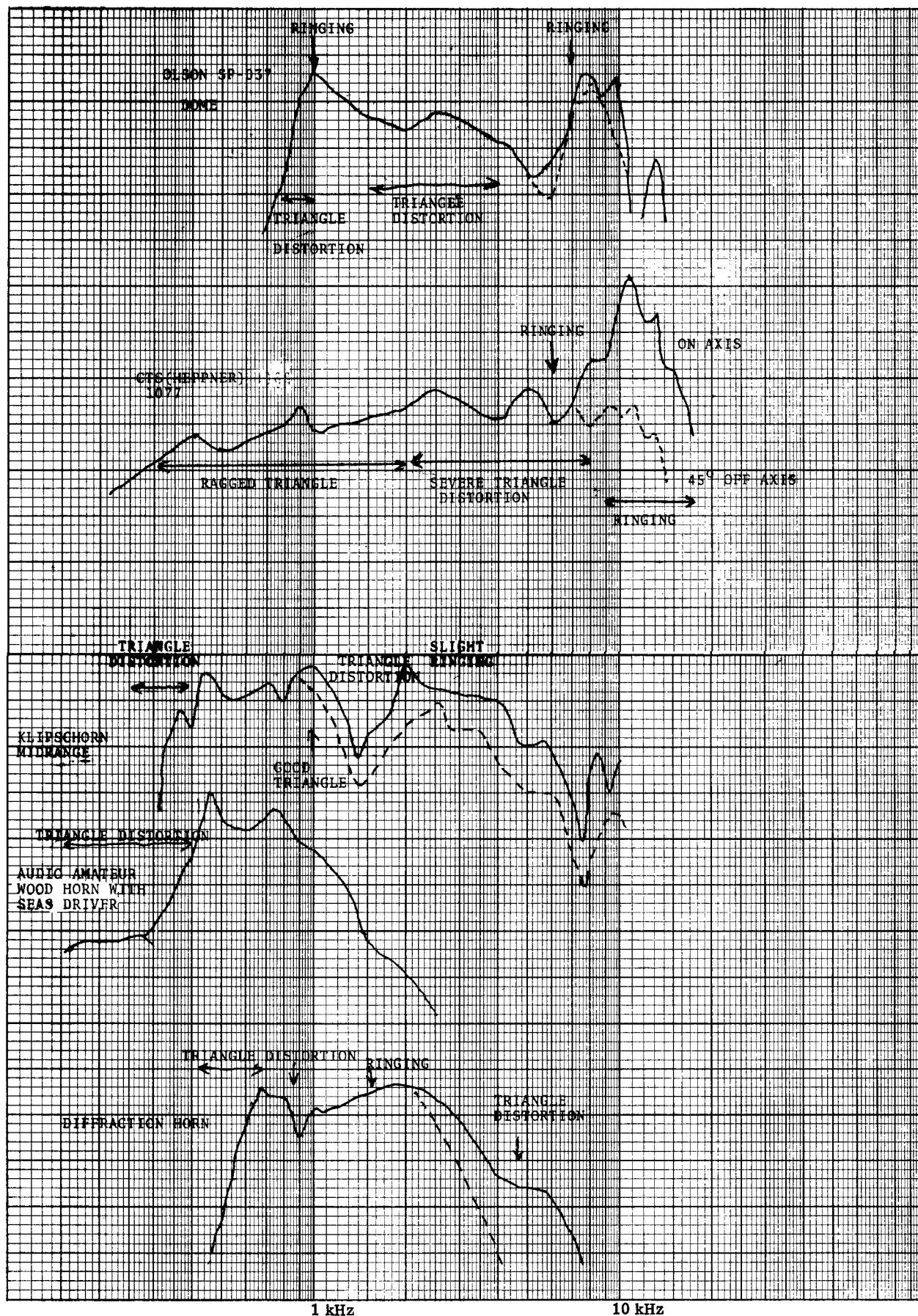
The Long L15ax is a 1½-inch, 4-ohm midrange available at McGee for \$13 and at Speakerlab for \$18 (model DM155). This appears to be the only reasonable dome midrange available, if only they had an 8-ohm unit. It might be useful for adding a midrange to an Advent.

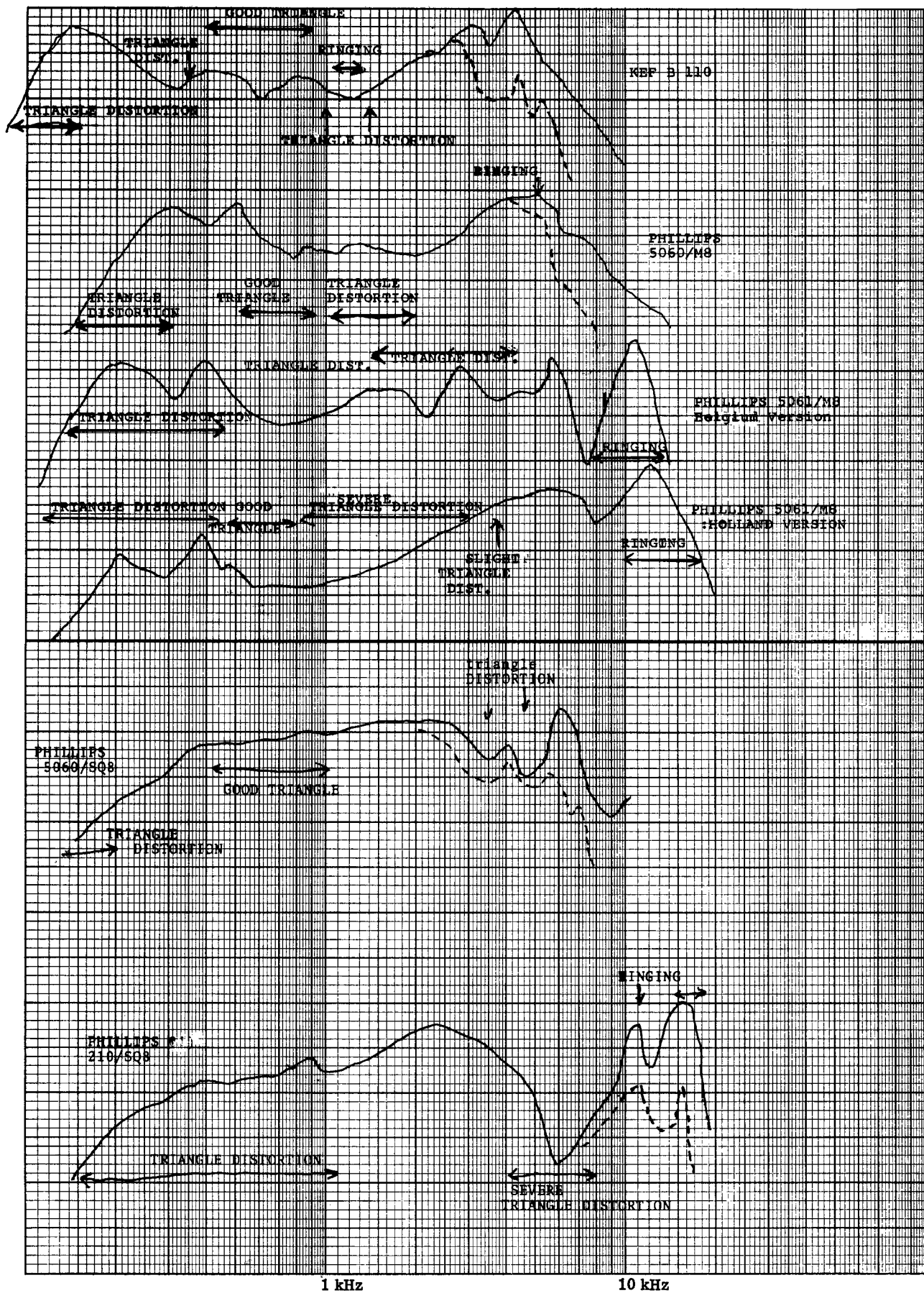
The last unit, the Utah M5JFC, is a 5-inch midrange available locally from You-Do-It Electronics for \$9.25.

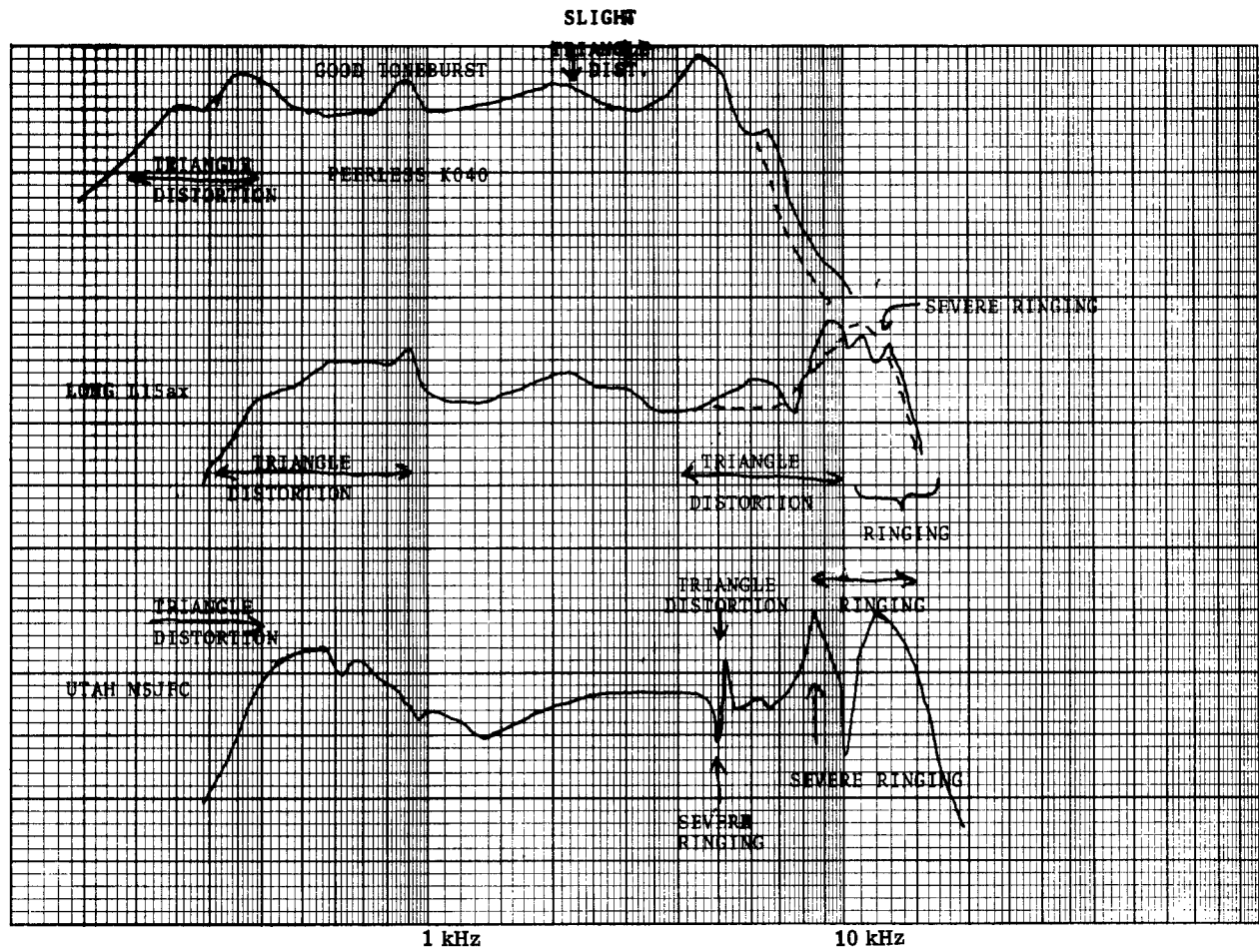












The Boston Audio Society does not endorse or criticize products, dealers, or services. Opinions expressed herein reflect the views of their authors and are for the information of the members.

The Grace 704 Damped Unipivot Tonearm

H. Gallegos

Among arc-describing tonearms, damped unipivots offer the greatest potential for low friction and resonance-free disc reproduction. On the other hand, they are relatively difficult to design and to build, which may explain why their potential has never been fully realized.

Unipivots have always been unreliable and difficult to use because of one or more of the following problems:

- a) Poor detailing,
- b) Poor quality of materials and/or imprecise construction,
- c) Extremely difficult setting up and balancing procedures,
- d) Erratic behavior,
- e) Balance affected by normal use,
- f) Unnecessarily high mass.

My experience with the Decca, the KMAL, the Grace 940, and the Mayware Formula IV tonearms has been, consequently and to say the least, disappointing.

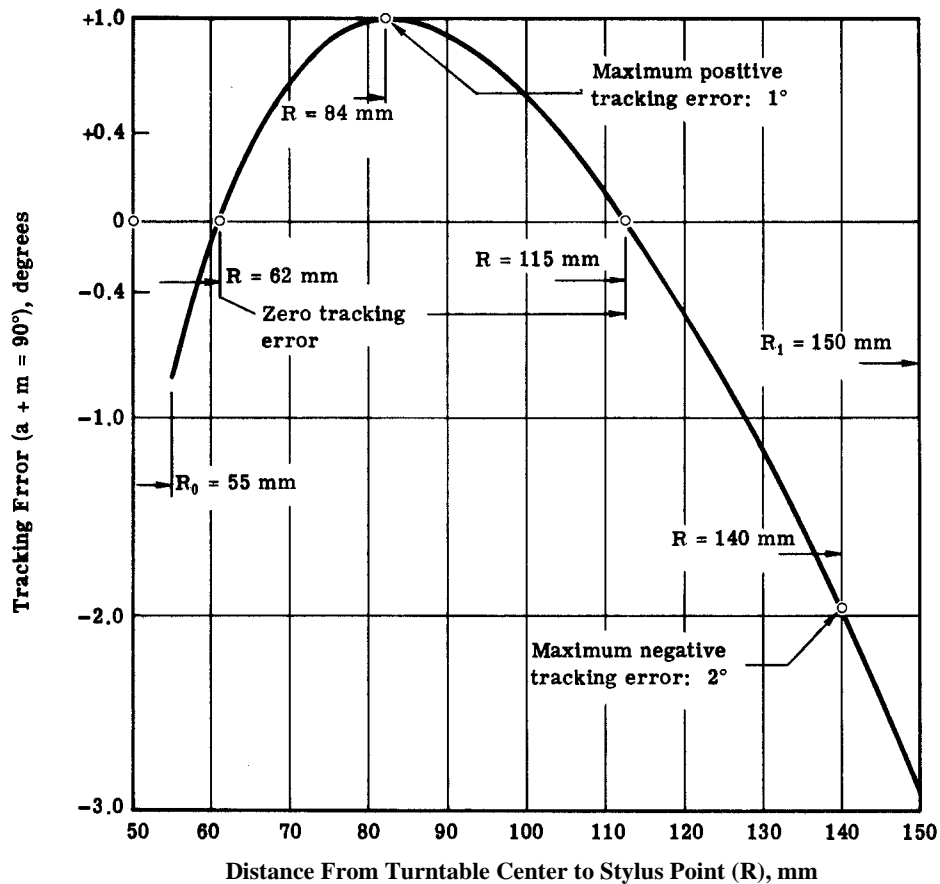
Fortunately for unipivot fans, a relatively new (November 1975 in Japan) damped unipivot is available: the Grace 704 (about \$120 in Japan). It uses essentially the same unipivot system as the Grace 940, but employs instead of the usual round tube a very precise, variable channel cross-section arm frame machined from a solid block of aluminum alloy. (The Grace 714, a similar tonearm that costs about 10% less, uses a wood frame instead of aluminum.) The 704 is made from first-class materials and is excellently detailed and constructed (at least as well built as the SME or the STAX), and it is very easy to install and to balance. Its mass is reasonably low (surpassed only by the Vestigal, the Mayware, and the Grace 707), maintaining at the same time an adequate moment of inertia in all planes, which makes it pleasant to handle. As indicated in Table 1, it has no bias compensation (is it really needed?), tracks admirably with all cartridges I have tested, and uses a special kind of removable headshell (two are supplied with the arm). When the headshell is removed, a circuit is closed in the arm frame, reducing hum and noise through the preamp.

As my instruction manual is in Japanese, I could only use what Grace suggests in a drawing as a procedure for setting the stylus location—an overhang of 15 mm over the turntable spindle. Unfortunately, it's not a very practical technique, so I decided to prepare a tracing diagram of the arm and to plot the tracking error versus distance from the turntable center. This permitted the preparation of a protractor (similar to the one supplied with the SME) to set the stylus overhang perfectly. I include the diagrams; they may serve as an example of what can be done for the correct setting up of other arms.

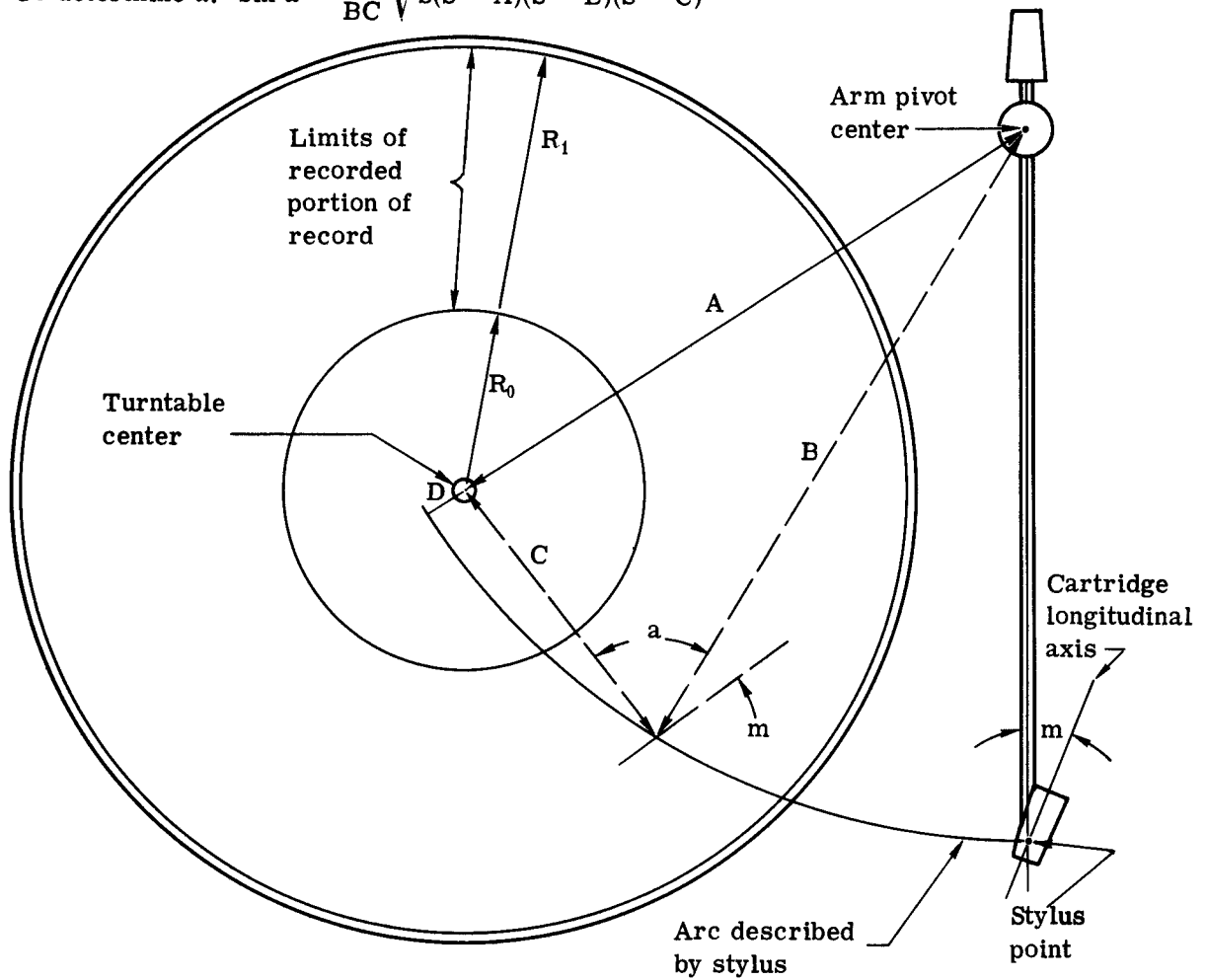
Table 1 — Comparison of Damped Unipivot Tonearms

Tonearm	Available	Headshell	Bias	Lateral balancing	Problems (see text)	Tracking*
Decca	No	Removable non-standard	Yes, variable magnetic	Eccentric main counterweight	a, b, d, e	Acceptable
KMAL	Yes	Fixed	Yes, fixed magnetic	Eccentric main counterweight	b, d, c	Not tested
Grace 940	Yes	Removable standard	No	Lateral independent weight	f	Good
Mayware Formula W	Yes	Fixed	Yes, variable weights	Eccentric main counterweight	a, b, e	Very good
Grace 704	In Japan	Removable non-standard	No	Lateral independent weight	Cable-arm connection and arm frame plastic support are not top notch	Excellent

* Tracking was tested using the Nakamichi Reference, the Denon DL-103S, the Decca Mk. V, and the Sonus Blue Label cartridges, the Shure TTR-110 and D.G. 641001 test records, examining the signal on an oscilloscope. At 315 Hz, the Grace 704 admitted at least 2 dB more recording level without mistracking than any other arm.



To determine a : $\sin a = \frac{2}{BC} \sqrt{S(S-A)(S-B)(S-C)}$



- A — Distance between arm pivot center and turntable center, indicated by manufacturer: 222 mm
- B — Distance between arm pivot center and stylus, indicated by manufacturer: 237 mm
- C — Variable distance between turntable center and stylus; its limits are $R_0 = 55$ mm and $R_1 = 150$ mm
- D — Stylus point overhang, indicated by manufacturer: 15 mm
- a — Variable angle, to be determined as a function of A, B, and C
- m — Fixed angle between cartridge and stylus/arm pivot axis, measured: 21.5°