THE B.A.S. SPEAKER

THE BOSTON AUDIO SOCIETY
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Membership Renewal

Last month we included a membership application/renewal form as the last page of the Speaker. If you have not returned your renewal form along with your check for $12, this will be your last issue. We hope you will all be continuing your membership.

We are including a renewal form again this month. Note that this form is slightly changed from that which accompanied last month's Speaker—use this month's if you can conveniently do so, and pass the duplicate to a hi-fi-oriented friend. Remember that the more of us there are, the more we can do for each other.

In This Issue

This month's Speaker is largely devoted to equipment. The publication is a three-part comparison article: A first-look report on the new Koss electrostatic loudspeakers; a report on several excellent tuners, including the Sequerra Model 1; and a faceoff between the Tandberg 9100X and Sony 377.

The Koss report makes clear why there will probably be fewer "in-the-home" tests of this speaker than of others. And also why first impressions might not be as applicable in the long run as with other speakers, i.e., the speaker itself may change if rumor is to be believed. Al Foster of Boston did the leg-and-ear work on this report.

The tuner report will be of more than usual interest as it took place under realistic listening conditions and includes notes on tuner capabilities relative to field strength (as measured using the Sequerra's panoramic display). This sort of report relates more to the conditions under which we must operate than does raw test "numerology." Syracuse BAS member Larry Hardin did the job.

The Tandberg-Sony comparison was conducted by MIT's Mark Davis, and grew out of the topic of phase shift covered in this month's meeting report. In trying to find noticeable or measurable differences between the crossfield biased Tandberg and the more typical Sony, Davis wound up doing a full-fledged comparison with results that will please some and surprise others.

Questionnaire. The subject of test reports leads into this year's BAS questionnaire. Test reports historically have been one of the more popular of the Speaker's features according to prior questionnaires. Now it is again your turn to help influence the Speaker's editorial content by telling us what you want to read and see in its pages. We can't stress enough the need for you to return the questionnaire included in this month's Speaker; it is the major source of feed-back to us, and so far as we know, it is one of the very few attempts made by any publication to steer a course charted by its readers—but you have to take the big step and invest a First-Class stamp and the time needed to fill out the form. Please do so, and as rapidly as is convenient for you.

Help Wanted

The BAS Speaker is looking for people who are willing to help with the production of the newsletter. More than anything else we need people to write articles on musical topics, followed closely by articles on other topics of interest to the audiophile. You don't need to be an accomplished writer (though that is certainly nice); we have editing skills available to put articles into a readable style. In fact, all articles will of necessity be edited (we hope this results in improvement, but on occasion we, like all others, make errors—note the errata that appear regularly). Not all contributions need be in the form of articles—news notes, book reviews, musical performance comparisons, technical notes, etc., all are welcome.

If you can participate a bit more regularly, we'd like to expand our publications committee with more people interested in writing meeting summaries and in being coordinating editors. If it is possible to find another two or three people to write meeting summaries, it would mean that no one would have to do this more than three times a year.

As for coordinating editor, this job is a bit more complex and time-consuming and a bit less well defined. The coordinating editor is responsible for putting together the month's newsletter. He must see that all contributions are received on time and he then edits the material, writes any introductory sections that are needed, and prepares the whole thing for delivery to Bob Borden, our production manager. He does not have to edit the "publications." We now have three regular coordinating editors, but we need at least one more (so each has the job no more than three times a year) and preferably several more to take care of emergencies. From an operational standpoint, coordinating editors must be local members.

Finally, we always need spare hands at meetings to distribute labels and newsletters and to package those newsletters that are being mailed. In fact, if you can even take a box of packaged newsletters to the post office the next day, that would be a great help.

— Jim Brinton

Car Pool to GTE

Rick Richardson will be coordinating the car pool to meetings at the GTE Research Laboratories in Waltham. Those needing rides from Boston University's George Sherman Union should call Richardson after 10:00 a.m. on the Saturday before the meeting at 492-4448 (leave a message if he is busy). He will then arrange for enough cars to stop by the Union to pick up passengers. If you are willing to be on call as a driver, Richardson would like to hear from you as well.

Errata and Other Changes

A Transformerless Balanced-Line Preamp for the Phantom 814 Microphone. On page 2, last paragraph, fifth line a typographical error changed "desired" to "undesired." It is the desired audio signal that is carried as the voltage difference between the two conductors.

Using the BAS Oscillator. A typesetting error occurred in the section of this article entitled "Test Dolby Tracking." The next to the last sentence in the first paragraph of this section should read, "Change the oscillator frequency to 2 kHz ..." rather than the erroneous 1 kHz appearing in the published text. Also in the first line of the second paragraph of this section, "1-kHz" should read "2-kHz."

In the section on "Measuring Frequency Response," an editorial change was made from "-25" to "-20 to -25 dB" for measuring the frequency response of cassettes. The author wishes it known that in his opinion objectionable amounts of tape saturation will occur at -20 dB and that -25 dB is a more appropriate level for this measurement.

Equipment for Sale

- Dyna Stereo 400 with meters, PAT-5, FM-5, PAT-4. All assembled by equipment reviewer, then factory checked. New, original cartons. Sell for kit price plus shipping. Telephone (212) 539-8060 evenings, or write to 27-41 Jackson Ave., Long Island City, New York 11101. Jon Graham.
- Rectilinear Mini-III's, one pair, $130; Micro-Acoustics add-on tweeters, one pair, $79; Shure A86A cable transformer (low-Z mike to high-Z input), $13. Call (617) 729-5700, days only. Ira Leonard.
- Southwest Technical Products (SWTP) graphic equalizer, constructed from kit, in mint condition, $75. Call 899-8090 days or 369-1949 evenings. Gary Rancourt.

Letter

Allison:One

To a new company such as Allison Acoustics, a product review is far more important than it would be to a well-established concern. Therefore I am more than casually appreciative of the time and effort expended by BAS members in reviewing the Allison:One, and I am of course pleased by and grateful for the many kind comments of the reviewers (July 1975, Volume 3, No. 10).

For the record, I would like to offer a few minor corrections of fact and an explanation or two.

1. On the front page I was credited with designing the "Air Coupler" in the early 1950's. This is not correct; the basic idea was conceived by Edmund Flewelling. My contribution was in developing working models and extending the idea to multitube designs with smoother response. You are correct in pointing out that Edgar Villchur's acoustic-suspension system made this and several other bass-reinforcement techniques obsolete.

2. Frank Callahan, our plant manager, did not supervise manufacturing at AR. He was quality control manager there.

3. In the body of the review there were conflicting comments on efficiency as compared with AR-3a and LST systems. Most likely the confusion was caused by the difference in impedance, which is about 2½ to 1. As a result, an AR-3a or LST takes 4 dB more power from the amplifier than an Allison speaker at the same setting of the volume control. Because the Allison:One is only 2 dB more efficient than the others, a 2-dB level advantage remains with the AR systems when a direct switch is made without a compensating change in amplifier voltage gain. Under these circumstances the actual efficiency difference seems to be reversed.

4. The same effect may explain why two of the reviewers thought that the AR systems produced more extreme bass output than the Allison:One. Fletcher-Munson equal-loudness contours show that our ears have great sensitivity to small changes in level at very low frequencies, and a 2-dB advantage at 35 Hz would be quite audible.—Roy Allison, President, Allison Acoustics Inc.

Allison:One Loudspeaker Review Update

In the July BAS test report on the Allison:One speaker, several references were made to room placement problems with the Allisons located near a fireplace. Now we can say confidently that we were fooled, and that the fireplace did not do "damage to the midrange and upper bass." We built a heavily braced, thick particle board panel and sealed the fireplace with it, then ran frequency response curves trying to, confirm objectively what we thought we heard. The frequency response was unchanged with the fireplace sealed. After between 8 and 10 manhours of work, we can only say that the room has some peculiarity that we do not yet understand. So any member who felt from his reading of the review that Allisons and fireplaces are incompatible should be reassured.

— Jim Brinton

A Transformerless Balanced-Line Preamp for the Phantom 814 Microphone Revisited

Peter Mitchell's idea for a transformerless balanced-line preamp capable of phantom powering the 814 microphone, as published in the August BAS Speaker, is basically a good one, but it ignores one or two practical points that materially affect performance. Mitchell's circuit is reproduced below in Fig. 1.

One problem with the circuit is that, because the source resistor in the FET has been shorted with a wire between pins S and G the FET is operating at a bias voltage of $V_{gs} = 0$. This can lead to non-zero gate current on positive signal swings, causing moderate amounts of distortion. The solution is simply to bypass the source resistor with a suitable capacitor instead of shorting it out. This will allow enough negative bias to exist to prevent gate current (see Fig. 2).

The other problem is that the effects of source impedance on common-mode rejection have been ignored. Source impedance has little effect on the common-mode rejection of a real transformer, because a transformer has effectively infinite common-mode impedance. Here the common-mode input impedance of the transformerless input is finite, and its effect must be compensated for.

Mitchell quotes a formula for common-mode gain which, in the notation of Fig. 1, is

$$G_{cm} = \frac{R4 \cdot R3 - R2 \cdot R5}{R3(R2 + R4)}$$

A more realistic figure of merit is the ratio of the differential gain to the common-mode gain, or the common-mode rejection ratio (CMRR). Deriving an expression for the CMRR from the above formula, we get

$$CMRR = \frac{R4(R3 + R5)}{R3 \cdot R4 - R2 \cdot R5}$$

We want the CMRR to be as high as possible and in fact if the values in Fig. 1 are substituted in the above expression, we get a perfect CMRR of infinity.

Unfortunately, the above two equations are accurate only if the source impedance is zero ohms. In this circuit the source impedance happens to be R1, or 4.7 Kohms. We can derive the actual expression for the CMRR from the circuit values using basic network theory:

$$CMRR = \frac{R4 \cdot R6(R3 + R5)}{R6[R3 \cdot R4 - R5(R1 + R2)] - R1 \cdot R5(R2 + R4)}$$

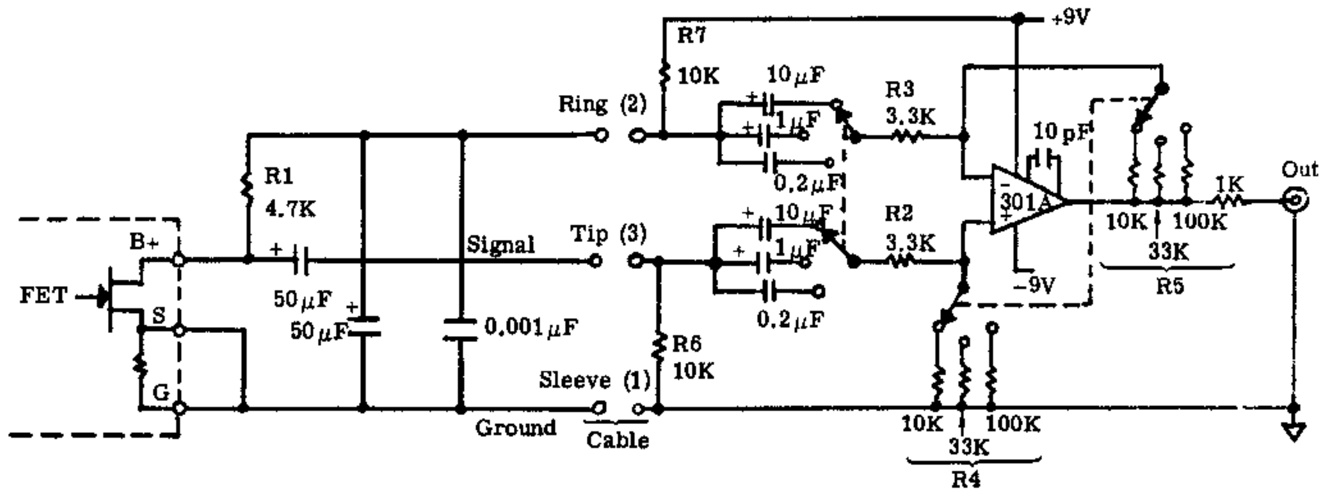


Fig. 1. Original schematic

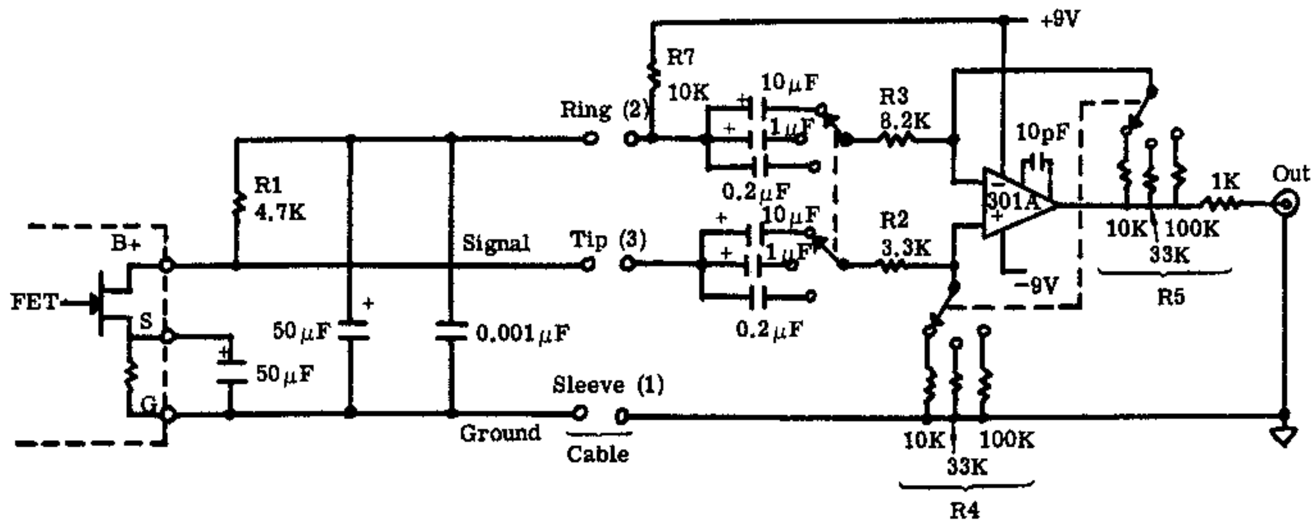


Fig. 2. Revised circuit

R7 is used to supply bias to the FET but it has no effect on the CMRR since both its ends are tied to ac ground.

Substituting the values shown in Fig. 1 in the above expression, we get the following values of CMRR for the three gain settings:

Gain, dB	R4, R5	CMRR, dB
4	10K	2
16	33K	4
26	100K	6

The CMRR is considerably worse than the prediction of the simple expression, and the gain is somewhat less as well.

While the situation could be improved by adding one or more active devices at the microphone to lower its output impedance, a simpler solution for one who has already constructed the above circuit is to adjust the component values to bring it into true balance. The first thing we can do is note that since R7 does not affect the CMRR, we don't need R6 to balance it. If we discard R6, the expression for the CMRR becomes

$$\mathrm{CMRR} = \frac{R4(R3 + R5)}{R3 \cdot R4 - R5(R1 + R2)}$$

We can now achieve infinite CMRR by adhering to the following two conditions:

R4 = R5,
R3 = R1 + R2.

Since R4 and R5 are already equal for all three values of gain, we need change only R3:

R3 = R1 + R2 = 4.7K + 3.3K = 8.0K.

The final circuit is shown in Fig. 2. Note that since R6 was removed, the 50-µF capacitor that was connected from the B+ terminal of the microphone to the plug tip is no longer needed, and can be used instead as the bypass capacitor mentioned earlier. If 5% tolerance resistors are used, the standard value nearest to 8.0K is 8.2K. The performance to be expected with 5% resistors is as follows:

Gain, dB	R4, R5	CMRR, dB (approx.)
2	10K	33
12	33K	40
22	100K	48

The finite source impedance is still holding the gain down below expected values, and in fact changing R3 to 8.2K has reduced it slightly. Should more gain be needed, one has only to increase both R4 and R5 by the desired amount.

— Mark Davis

[Editor's Note: Obviously, the higher the CMRR, the greater the rejection of unwanted noise like hum or low-frequency electromagnetic interference. CMRR-crazed perfectionists can use a trimmer resistor instead of the 8.2K resistor (R3), and—as they apply a common signal to both audio lines—view the preamp's output on an oscilloscope (or use a sensitive ac-VTVM) to trim for a null. It might be possible to get from 10 to 30 dB more CMRR this way.

But even this might not be enough if you are confronted with radio-frequency interference from TV, FM, or air-ground communications links. These services operate (as we know) in the neighborhood of 100 MHz, and circuits can look "different" from that point of view. If this sort of RFI seems to be your problem, Mark Davis suggests as possible cures that you might connect a 30-pF capacitor between the + and - inputs of the op-amp, and/or bypass the +9 V and -9 V supplies to ground using 0.01-µF discs. In the former case, get the capacitor as close to the op-amp inputs as possible, and in both cases keep leads as short as possible.—JBB]

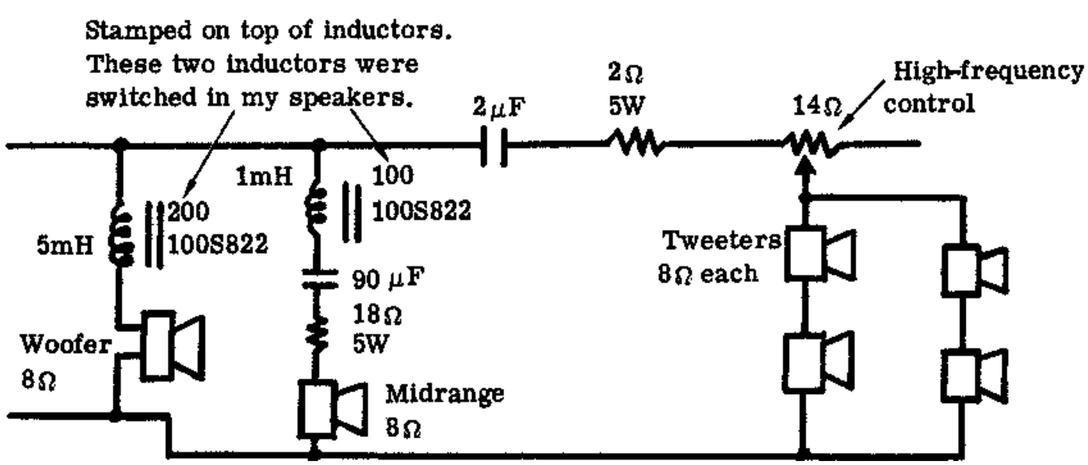
[Editor's Note: At press time we had not received any comment from Peter Mitchell, but we understand he is in agreement with the indicated changes.—JBB]

Checking for Sloppy Loudspeaker Quality Control

Last winter when Victor Campos unleashed the Micro-Acoustics QDC-1 cartridge upon the Boston audio world, I immediately went out and purchased one. But in contrast to everyone else's experiences with the cartridge, I could not hear any difference between the Micro-Acoustics cartridge and my Audio Technica AT-11. I concluded something was very wrong somewhere; and besides my system never had sounded as good as others I have heard.

After much investigation I found several problems in my Rectilinear III speakers. First, the open-framed cone tweeters did not have sub-enclosures covering their backs to protect them from woofer pressure waves. The midrange drivers in my speakers were covered properly, but none of the tweeters were. This was solved by purchasing little freezer containers and placing them over the back of each tweeter with plenty of RTV silicone adhesive. At the same time, I chose to replace the transistor radio drivers Rectilinear uses for tweeters in this unit with higher quality drivers. (I have always wondered why Rectilinear chose to place one of the four tweeters at floor level; presumably to provide mice and other wee beasties with extended highs ... very considerate.)

The second and more serious problem lay with the crossover networks, which had two inductors (a 0.1-millihenry and a 5.0-mH) interchanged in the circuit. The schematic shows the circuit diagram of the crossover networks as they should be with the inductors labeled accordingly. Both my speakers were miswired in the same way, so I assume several units in the production run may have been wired similarly, i.e., wrong.



The third problem appeared when I tried to trace out the wiring and compared the two speaker systems. Red and black wires were used, but the color coding differed between the two units, so I cannot include wire colors in the diagram. If no color code convention is used in manufacturing the speakers, I wonder if speaker phasing problems might not occur. My drivers were in phase, but I suspect it was due to good luck and not careful production work or quality control.

I tested phasing between the woofer and midrange drivers by applying a sine wave to the speaker at the crossover frequency (250 Hz in this case), then moved a sound level meter (see following note) back and forth along a line between the woofer and midrange drivers. [This can be tricky, unless one takes pains to avoid sound waves interfering as they reflect from walls, floor, and cabinet edges, or are emitted from other drivers. A supply of baffling material—say glass fiber—to put in places where sound absorption is needed is a useful safety factor.—JBB] A rise in sound level midway between the two drivers indicates they are in phase; a slight dip in response indicates they are out of phase and the wires to the woofer should be reversed.

For the midrange-tweeter test, a 3000-Hz tone (with Rectilinear III's) should be applied to the speaker and the SLM moved between the midrange and one of the nearby tweeters. If an out-of-phase condition results, it might be that either all four tweeters are out of phase with the midrange, or just this one—or two, or three—is out of phase. At this point it might be better to trace the tweeter wiring to determine proper phase.

I have found the Rectilinear HI's woofer and enclosure work well down to 40 Hz (after the backs of the tweeters are sealed off properly). But from my experience quality control leaves much to be desired and the "stock" tweeters could be greatly improved. I can see quality control is one reason to avoid house brand speakers, but obviously even nationally distributed brands have their own QC problems.

— Jim Nichol

A Nichol Sound Level Meter

In order to fill my need for a wide-range sound level meter (SLM) with known response characteristics for speaker testing, I constructed the microphone amplifier shown in Fig. 1. I used a Thermo-Electron 814 microphone capsule purchased from Electronic Enterprises (3305 Pestana Way, Livermore, California 94550) for $42. Electronic Enterprises will run a frequency-response curve on the capsule for a $10 setup charge (Fig. 2).

The amplifier drives an audio VTVM and the readings are read off the VTVM's dB scale. The scale readings give only <u>relative</u> sound pressure level (SPL) readings, but this is all I require for speaker measurements and room equalization. If desired, this SLM could be calibrated against another SLM using a tone source and matched by varying the feedback resistance. [For limited use as an absolute SPL meter, the 814 could be calibrated against one of the several SPL standards available locally to BAS members. It is necessary that the 814 be properly encased, though. These standards won't work with "bare" mikes.—JBB]

With the 200x gain of the amplifier, I use the lowest couple of ranges on my audio VTVM, 0.03 and 0.01 volt full scale.

I also found it useful to mount the microphone on the end of a multisection telescoping radio antenna. This allows the microphone to be positioned easily in front of the speaker under test, elsewhere about the room for averaged measurements, or where the listener's head is normally located. The antenna "boom" allows meter readings to be taken without affecting the sound field around the microphone, avoiding having to crouch down and crawl up behind the SLM to take readings, making people think you're weird.

— Jim Nichol

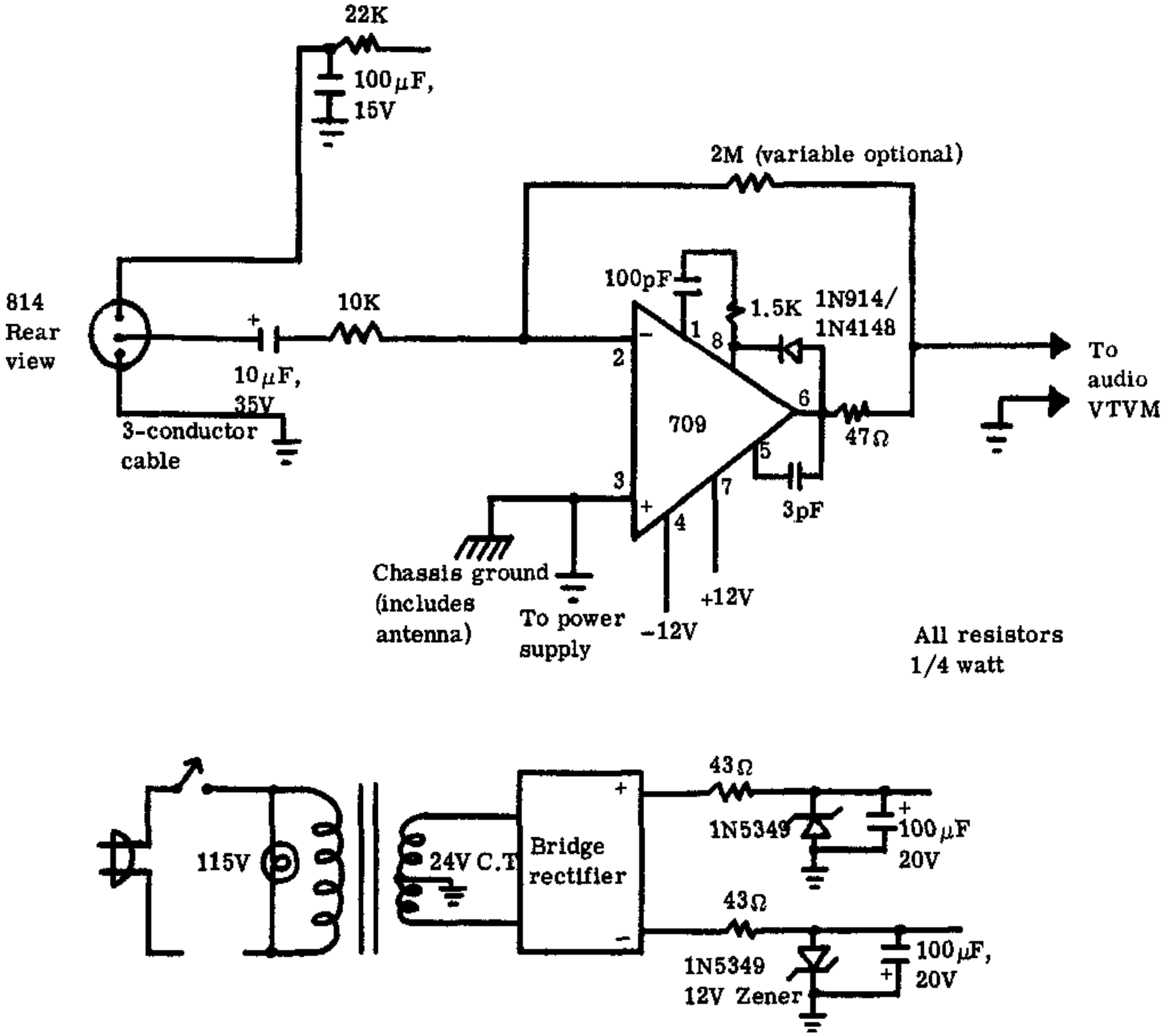


Fig. 1. SLM schematic

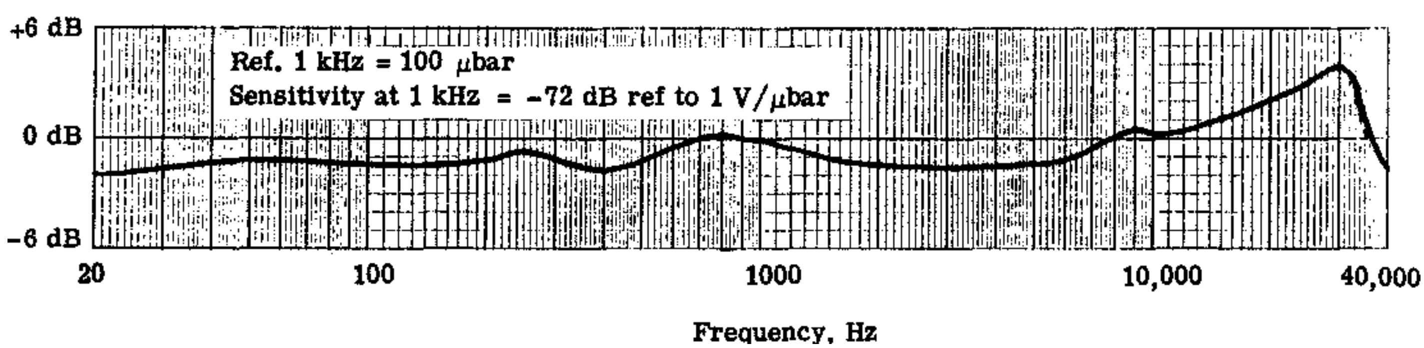
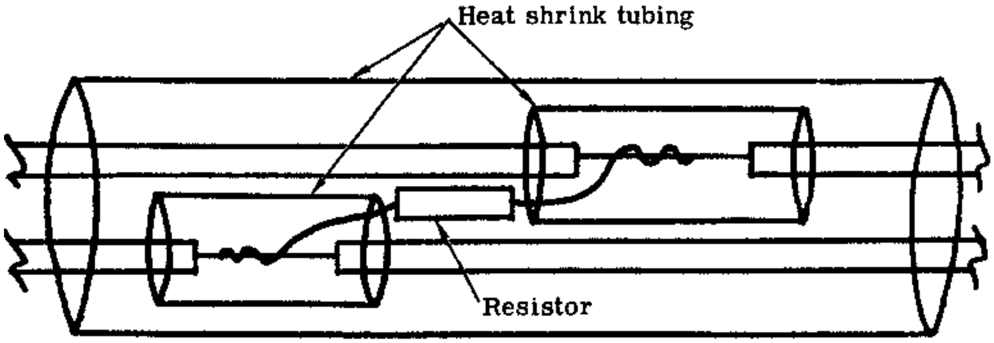


Fig. 2. Sample frequency curve for 814 microphone

A Cheap Earphone

It is common knowledge that the earphones packed with portable transistor radios are little more than afterthoughts, but there is one available from Lafayette Radio that is an exception. It has much more bass, to the point of sounding tubby, than other earphones. Unfortunately, it has an impedance of 6700 ohms (assuming one can trust Lafayette's specs), which will upset some radios that like to drive 8-ohm loads. This may be cured by soldering a low resistance, say 10 to 100 ohms, across the earphone leads as shown.



The earphone is Lafayette no. 40F78010 and is priced at $2.95. Local stores do not usually have this item in stock but they can order it from New York, or you can do so through the catalog.

— Jim Nichol

Everything's Up-to-Date in Buffalo

Next time you OD on prune juice en route to Chicago ... or have some other equally compelling reason to visit Buffalo, New York, ... don't despair, audiophile. In this least likely of cities (aside from Sharpe headphones, there is no hi-fi manufacturing within 150 miles) there are, half a mile apart, two of the most interesting high end hi-fi shops I've seen. Both cater to the same trade, and both began as one man selling at cost plus 10% out of his basement. In almost every other respect, the two dealerships are as different as they could be, because of the markedly different personalities of the proprietors.

The bigger and older (3½ years) store is humbly called Transcendental Audio. The proprietor is Bob Minnick, formerly of Northeastern, Harvard, and Audio Lab. Rather than an audio engineer, or even as amateur expert, Bob comes across as an enthusiast and a sybarite par excellence. He told me, for example, that he had had an ADS 60-watt-per-channel car stereo in his Mercedes 450SL. However, since he replaced the Mercedes with a Ferrari Daytona, he has had no music in his car. When his turbocharged Porsche Carrera arrives to replace the Ferrari, he may get another ADS.

Trans Audio's showrooms reflect this flamboyance. The emphasis is on the visual and psychological effect as much as on the aural. The top end showroom, in addition to unusually good acoustics, has a working fireplace and a large, prominent Union Jack, for example. The Transcriptors turntable with Vestigal tonearm, the B&O turntables and receivers, and the Cambridge Audio amplifiers and speakers stand out on one wall, and one end is consumed by the Dayton-Wright XG-8 Mark 3 full-range electrostatics. Other featured speakers, given visual emphasis over boxes like the IMF's and Celestions, are the B&W and the Gale GS401A. Bob was very enthusiastic about the latter, but not I, in a very brief listening. The Gale cabinet is all metal, and therefore my eyes helped persuade my ears that the sound was quite harsh and peaky, at least compared to the D-W electrostatics, which cost five times as much. This, of course, is the room with the Sequerra.

Bob Minnick professes a fascination with British audio equipment and the thinking that produces it. It began when he started selling Quad in his basement years ago. "Take the problem of static electricity on records," he said. "American engineers could be expected to design an atomic reactor to ionize the air around the turntable. The English [Transcriptors] simply eliminated the platter."

In a second showroom Bob exhibited the giant Tannoys he bought from the salvage of the Queen Mary. I did not hear them, but Bob said they are pleasant to listen to but not very accurate.

The top end accounts for about a third of Trans Audio's sales. The remainder is from the sort of equipment most mortals buy, and the trend at Trans Audio seems to be in that direction. But Bob made an interesting point by saying, "Right now we still don't know how much people will pay for top end stuff. I want to be the Nieman-Marcus of audio. At Nieman-Marcus you can pay $4 for something ... or you can pay $2000 for something else that will accomplish the same purpose. But either one will be the best you can get at the price," Bob concluded.

A half mile away from Transcendental Audio is the Stereo Emporium. No fireplaces or Union Jacks in this cramped dealership; no multi-media sybarites at the helm.

Jerry Bennett of the Stereo Emporium has an MBA and spent two years in a management training program at a large manufacturing organization before deciding he wanted to do something on his own. Like Minnick, he started selling in his basement at cost plus 10%.

"I naively assumed that I could expand that business in a small store, but I found that even cost plus 20% wouldn't cover all the overhead, so now we charge list," he said.

Jerry's partner at the Stereo Emporium is Gary Nowak, BS in chemistry and former music teacher. The third and final employee of the Stereo Emporium is Mike Ortolando, who is the technical expert and has the title of manager.

"The presence of two top-end dealers makes Buffalo appear to be a more sophisticated market than it really is," Gary said, but he went on to state that he feels the top end is the business of the future, even more than the present, at the Stereo Emporium.

"We started out on low-priced equipment," said Mike, "with Japanese receivers I didn't like at all. When the bottom fell out of that part of the market, we were saved by the top. We had customers who wanted to buy top end, and they wanted to buy from us. In February 1974 we got the franchise for Audio Research away from Trans Audio, and we've picked up several more since then."

The Stereo Emporium's tiny showroom is dominated at one end by the Koss electrostatics and six panels of Magneplanars. On the adjacent wall are the Dahlquists, Magnapans, and several bookshelf speakers, especially RTR. The electronics wall includes, beside Audio Research, such names as Levinson, the Quintessence Group, Ampzilla, and SAE. The featured turntables are Linn Sondek and B&O.

Incredibly, there is not a single open-reel tape deck anywhere in the store! There are few cassettes in evidence. Jerry Bennett explained that he has been unable to get a franchise for either Crown or Revox. For marketing reasons he does not want to carry Sony or Teac, and he does not particularly admire Tandberg or Ferrograph products. Trans Audio beat him to the franchise for Neal, an English cassette deck he particularly admires.

A large part of the Stereo Emporium's product line-up problem stems from their 600 square feet of floor space—about a third that of Trans Audio. When their lease runs out next July, Jerry, Gary, and Mike expect to move to larger quarters, but until then, they're stuck. They're planning to open a store in Chicago and are at the moment more concerned about the new facility there than the old one in Buffalo.

So there you have it: in the least likely city east of Peoria, two very different organizations catering to the luxury audio market, one emphasizing luxury and the other emphasizing audio. If you have to be in Buffalo, birthplace of Millard Fillmore, don't miss either. — David F. Temple

Some Sound Advice for *Sound Advice*

The use of double-blind A-B testing by the magazine Sound Advice is a welcome step forward for regular reviews of new audio equipment. However, this in itself is not sufficient for meaningful results.

In my own comparison testing of components, I have found many instances where slight differences in sound could be heard, quite reproducibly, in double-blind tests. And yet the judgment of which component provides something closer to the straight-wire sound can be made to flip-flop back and forth between A. and B by making some small adjustments to tone-control knobs or graphic equalizer sliders. (This topic has been given repeated treatment in the BAS Speaker, but only once, so far as I know, in commercial magazines: Larry Klein's column on page 15 of the July 1975 Stereo Review.)

In many cases these differences are as great from sample to sample as they are from model to model, being rather slight in the first place. In other cases they are due to the loading effects of the impedances and capacitances of the other components in the system.

Except for occasional large differences in the susceptibility to loading effects (including overloading), most differences between the generally well-known amplifiers are trivial in my opinion. This is because such differences are almost random, and they are so readily overwhelmed by other factors.

Therefore, regarding "sound advice" on amplifiers, I wish someone would donate graphic equalizers to their reviewers. Then we could get on to more meaty subjects such as microphones, recording media, and loudspeakers. — Dan Shanefield

More on *Sound Advice*

The results published by Sound Advice re the Phase Linear 400 et al. were interesting. However, I still feel that the 400 confused complex music and (anyway) my wife didn't like to listen to the system with the Phase driving AR-3a's. On KLH-9's the Phase was noticeably zippy. I have not tried the 400 on the Magneplanars I recently bought since I had by then traded the Phase for a piano . . . I suspect that compatibility is the real issue here in the sense that:

1) No system is perfect and different people have differing tolerances for various inaccuracies vis-a-vis the live performance; hence the disagreement as to what equipment is less-inaccurate.

2) Different amps are compatible with different loudspeakers, etc. For example, Ampzilla sounds fine with the KLH-9's but a mite zippy with the Magneplanars, while the Futterman sounds lovely with the Maggies but maybe a little soft with AR-3a's. (However, at $350, the 60-watt-per-channel Futterman could be considered a "best buy" and is also nice on KLH-9's etc.) — Tom Mashey

Record Catalog Available

A record catalog from the Canadian Broadcasting Corporation (CBC) lists recordings made by Canadian artists performing music from various periods, including works by Canadian composers. The records cost $5 and Toronto BAS member Christopher Gupta feels they are good pressings. He adds that "there are some really unusual and rare musical performances available." The booklet entitled "The Canadian Collection Record Catalogue," is available free from: CBC Publications, Box 500, Station A, Toronto, Ontario, Canada M5W 1E6.

Serendipity with a Technics Turntable

I recently bought a Technics 110A turntable and started dreaming up ways to build up the height of the (apparently) low silhouette base. You see, my Decca International tonearm requires 4 ½ inches of depth below the mounting board to accommodate its long magnetic-isolation tube—even when the space-saving right-angle connector is used.

But lo and behold—the Decca fits as is: There's no buildup required. I punched a hole in the turntable base's sheet-metal bottom just below the mounting hole. The magnetic tube drops to a point right above the lower hole. I merely had to tilt the turntable, plug the cable in from the bottom (outside the case), and place the turntable back on its regular legs.

Results? The Shure V-15 Type III sounds great in the arm and the whole system handles like a dream.

I am still working on some other details, but anyone with questions should feel free to call me at (617) 729-5700 or to write me c/o P.O. Box 7. — Ira Leonard

Attention Rabco SL-8E Owners

BAS member Dean Slindee wants to contact anyone interested in experimenting with modifications to the Rabco arm. Slindee is currently offering an oiled redwood cartridge shell modification for the arm through Audio's classified ads and he'd like to continue to improve upon it. He hopes to replace everything within the Rabco pivots with an all-redwood unipivot arm design. He expects that this design approach would result in the lowest mass (lightest weight) arm of any Rabco modification or even of most other arms.

If you're interested, write him at P.O. Box 55, Lansing, Iowa 52151.

Discwasher Fluid

You probably use a Discwasher II on your records; right? If so, you know how expensive "dII" fluid is—$12.00 for 16 ounces "at discount."

My wife sniffed the dII fluid, which is not marked as poisonous, and suggested white distilled vinegar, which is 29¢ a quart. I diluted the vinegar 20 to 30:1 with distilled water (i.e., 20 to 30 parts water to one part distilled white vinegar).

I tried this mixture on a really grubby record that dII fluid could not correct even with multiple cleanings. I put the mixture on with a one-inch sable brush and lifted it off with the Discwasher brush. Results—fantastic! Clean—clean—quiet sound from the disk. The vinegar should be safe on records, but I'll know better after more extensive use. If you try this, let me know how it works. — Tom Mashey

Speaker Magnetism Strikes Again

In June we noted the danger of placing tapes in the magnetic field of a loudspeaker. As Bill Shelton pointed out on "Shop Talk," a speaker's magnetism can also mess up the picture in a color TV if you put the speakers too close to the set (e.g., on either side for TV/FM simulcasts). Heathkit color TV sets are shielded and so are relatively immune to magnetic interference, but other brands may exhibit either mild or grotesque smearing when speakers are located within two or three feet of the picture tube. — Peter Mitchell

In the Literature

Major contributions this month come from Michael Riggs and Dana Craig. We solicit input from all members, especially from those reading the British and Canadian publications and from those with access to the dealer-only trade journals. Send either to P.O. Box 7 or, for faster service near the publication deadline, to H. Zwicker, MIT Lincoln Labs, P.O. Box 73, Lexington, MA 02173. We have had several requests for longer abstracts. Expanded listings will appear, but only in those cases where we will not infringe upon the right of a publication to sell the information it prints.

The Absolute Sound, Vol. 2, No. 6

Features a tuner survey including, among others, the Sequerra I, the Yamaha CT-7000, the Citations, the Pioneer TX-9100, and the Marantz 10B. For their audio quality tests, TAP "broadcast" their own source material with a Sound Technology FM generator and A-B'd the tuner outputs with the source output. Some may find results (9100 just OK, Onkyo 4055 excellent, 10B a thing of the past) surprising. Complete reviews of the Phase Linear 4000, the new ADC cartridges, and the B&O 4002. Twenty-two shorter reviews covering a wide range of products (Classic tape, Micro-Acoustics QDC-le with another manufacturer's nasty reply, AR-LST, etc.). Controversy column contains several letters about the Vestigal arm and Transcriptors' puzzling reply; errors appear on both sides of the issue, so read with several grains of salt. New section discusses TAB cartridge measurement techniques (graphs for several cartridges included) and shares some ideas on how to get the most out of one's system. Victor Brociner contributes an article on speaker measurements—many good thoughts, but the article never gets to comparisons with listening tests. Finally letters (including one from BAS member Al Foster on tubes versus transistors), record reviews, and technical tips. A damn good issue. (But as for the front cover, the implication is that God created the phono pickup, while it should have been man; and the tracking error on that arm would be horrendous unless the disc is played counterclockwise. As for the back cover—well, the BAS is also trying to put more photographs in its publication, but they will never be like that.)

Acoustical Society of America, Journal of the, July 1975

This one is mostly on automobile and aircraft noise, so here's a short list of the less obscure articles: ... Manikin for Acoustic Research (p. 214), (Tone Burst) Loudness Enhancement on time delay effects, (p. 229), and ... Microphone Utilizing an Electret Transducer (p. 273).

Audio—Sept. 1975

Reviews of Ampzilla, Supex SD-900E, Sony TC-755, and Kenwood KP-5022 turntable. This month Audio has chosen to Heyserize the Avid 102. Article on bi-amping discusses optimizing crossover frequencies for best power distribution and lowest distortion and why bi-amped systems sound cleaner even when the total amount of power available doesn't exceed that found in a similar conventional setup. Article on FTC preconditioning rule interesting mainly for table showing average and peak powers for reproduction of various records, FM broadcasts, and noise. The author uses an elegantly simple technique for measuring the long-term average and peak power simultaneously, and he makes a very good case for a 10 to 15% average-versus-peak power amplifier requirement. Article on cassette machines is a throwaway intended for beginners.

Boston Phoenix, August 12, 1975

Who is Klaus Tennstedt?: Check out your library for a copy the next time you are there (front page on section two). This is important mainly because it's there; someone other than the BAS is trying to push this dark-horse candidate for conductor of the year. The article, based on a short interview, states that he will be back to Boston in the summer of 1976 and fall of 1976 and 1977, but not this fall. Please, if anyone discovers when his concerts with the Cleveland or the Philadelphia orchestras will be broadcast, let us know quick!

<u>dB, August 1975</u>

- In the CD-4 Groove: Some excellent photos of CD-4 grooves using a scanning electron microscope. (p. 12)

<u>Electronic Design News, Aug. 5, 1975</u>

- Ease Hard-Limiter Design With Op Amps: Circuit using op-amps as limiters for audio signals. (p. 76)

<u>Electronic Products, Aug. 1975</u>

- *See* More Sound: Article on low frequency spectrum analyzers, specifically HP's 3580A, by an HP engineer. (p. 35)

<u>Electronics, Aug. 21, 1975</u>

- Reticon Readies 1K Analog Delay Device for $10: SAD 1024 contains two independent arrays of 512 elements, each in 16-pin DIP; dynamic range >75 dB; signal bandwidth >100 kHz; sampling frequency 1 to 2 MHz; <1% second-harmonic distortion; N-channel MOS bucket-brigade analog delay device for audio uses. (p. 25)
- Linear Pot and Op Amp Provide Tapered Audio Volume Control: Design uses inexpensive linear pot and op-amp to approximate action of more expensive audio-taper pot. (p. 83)
- Phillips Hopes New BBD will Recapture Its Market Lead: New high-performance bucket brigade from Phillips of The Netherlands. TDA 1022: 512 stages, delay 51.2 to 0.512 milliseconds, dynamic range 2.5 volts rms, typical attenuation 3.5 dB, clock frequencies 5 to 500 kHz, $4 in volume. (p. 55)

<u>IEEE Spectrum, Aug. 1975</u>

- Television on a Silver Platter: Excellent article on competing videodisc systems. (p. 34)

<u>IEEE Transactions on Consumer Electronics, Aug. 1975</u>

Two articles on CD-4 decoders, one from RCA on their LSI chip (without preamp but complete with ANRS) and two from Hitachi, again with ANRS, but with preamplifier and a horrendous number of external components. (pp. 185 and 195, respectively)

- Projection Television: From GE: a review of the "theoretical" requirements for a bright picture. (p. 206)

<u>Popular Electronics, Sept. 1975</u>

- Stereo Scene: A trip through the Capitol pressing plant. (p. 22)
- What Does Your Stereo Receiver Dollar Buy?: A very useful review of the price versus performance schedule of receivers. $400 list is the best-buy range, which agrees with this author's investigation for several friends just getting into audio. (p. 33)
- Build a High Performance CD-4 Demodulator: A kit for $50 from Southwest Technical Products. Very complex, using one LSI IC per channel, almost convinces me that four channel may not be dead if only someone other than RCA would encode some good music. (p. 39)

- •Two audio reviews, as usual repeated from Stereo Review (what a cheap way to fill space), plus the Heath GR-400 color TV kit. (p. 71)

Radio Electronics, Sept. 1975

- •All about Oscilloscopes (Cont.): Brief discussion of measuring audio amplifiers and trouble-shooting equipment (logic circuits, power supplies, and TV). Emphasis on the vertical amplifier (rise time) and on the disruption of the measured stage by a scope probe. Useful, but not vital to the audiophile. (p. 40)
- •Signal-to-Noise—What Does it Mean?: Warnings on misinterpreting the S/N specs among types of equipment (e.g., cassette versus open reel) and between manufacturers. Too brief to complete the job (one article should be devoted to each type of device rather than lumping tape, phono, and turntable specs all into a single article), but this information is useful as a partial compilation of definitions used by various labs and manufacturers (e.g., ARLL, NAB, and DIN-a and -b weightings for rumble). (p. 50)
- •Test reports on the BIC-960 (a rather shallow, incomplete report, stressing features rather than performance and totally ignoring the arm; good acoustic isolation found, however) and the Empire 4000D/111 (also short on data, although this cartridge tracked well at 1 gram and seemed quite flat below 20 kHz for a CD-4 type of cartridge).

StereOpus, Vol. 1, No. 1, Spring 1975

A new magazine in The Absolute Sound and Stereophile mold, though not as ambitious, tart, or well written as its predecessors. Reviews of Micro-Acoustics QDC-le, Grace F8F, Decca 4RC, Micro-Acoustics FRM-1, Infinity Monitors I and II, Phase Linear 400, and Citation 12; short reports on Crown DC-300A and Ohm F. A couple of record reviews included. Editor plans home brew column with emphasis on speaker construction. (StereOpus—Quarterly. $9/year; first class mailing $2 extra. All subscriptions begin with first issue of current volume. StereOpus, P.O. Box 269, Fort Walton. Beach, Florida 32548.)

Wireless World, July 1975

- •Dolby Noise Reducer, Part III: Alignment and use of the WW kit, which apparently completes the series.
- •Noise—Confusion in More Ways Than One, Part W: This installment is of no use to the audiophile.
- •75 Years of Magnetic Recording, Part V.
- •Active Notch Filters: Mostly theory, with little comment on the shortcomings of high-Q filters especially the tendency to ring when used in audio circuits. Too late for use with the Tanglewood birdie. (For another view of a notch filter, see QST for September 1975, where a $10 kit designed for selectable notching at 750 Hz is described; uses cascaded low-Q 741 stages to eliminate ringing.)
- •News of the Month: Comments on VAT and its application to hi-fi goods (still confused) and a note supportive of circularly polarized transmitter antennas for TV, for the same reasons that it is of use in stereo-FM (it reduces ghosts caused by multipath).

August BAS Meeting

Business Meeting

Jim Brinton opened the August meeting at GTE Labs by reminding the 80 BAS members and guests that registration was still open for the final tuner clinic, to be held in September. He also strongly urged more members to become involved in the BAS publication program. There are a number of areas in which contributions can be made. Besides the regular features of In The Literature, Meeting Report, equipment reviews, and technical articles, writeups on industry news, book reviews, and articles on musical topics are needed. Assistance with typing, drawing of figures and schematics, packaging, and mailing are also welcome. This is also an excellent way to meet other members and exchange experiences. You will soon find that the BAS Speaker is only the "tip of the iceberg" of interesting and valuable information circulating within the BAS.

The great buy on cassette tape (C90 chrome, $2.00) offered by Al Foster was quickly consumed at the break. Al will also be taking orders on the new Sheffield disk, LAB-2, with Thelma Houston (not the recently announced dbx disk). This album, their first with a vocalist, will go for $6.75.

Peter Mitchell sadly announced that the year-old BAS oscillator will be yet another month older before it is ready for distribution. Lafayette Radio has been abandoned as a source for the long-awaited switches; they will be purchased from another source at slightly higher prices. Orders may be placed with Peter for the 814 microphone kit as described in last month's newsletter. However, you may want to wait and consider the circuit devised by Rene Jaeger for packaging the 814C with a preamp in the microphone capsule. Distortion measurements were made by Peter on the phantom powered 814, described in the July Speaker. Distortion was very low at 95 dB SPL, 1% at 106 dB SPL, and 3% at 117 dB SPL. It is possible that the 814C, which will be reported upon soon, may yield lower figures.

A new project on the evaluation of automobile monaural and stereo FM radio was announced. Volunteers are needed to test-drive radios on an established route and fill in an evaluation report. This will mean a little work but should result in quality judgments based on more meaningful criteria than have been used in recent test reports. Contact Tom Horrall for further details.

Meeting Feature

A program concerned with the audibility (or inaudibility) of phase shift, including theoretical considerations and demonstrations, was organized by three BAS members: Dennis Colin, Mark Davis, and Rene Jaeger. Dennis is a musician and electronics consultant to Aries, a manufacturer of music synthesizers. Mark, a doctoral candidate studying psychoacoustic phenomena at MIT, is involved in experiments verifying a new model of the hearing process. As chief engineer at dbx, audio and high fidelity are both a vocation and an avocation for Rene.

Lecture. Mark began by describing a new model of the ear that has been proposed by Professor Campbell L. Searle, formerly of MIT and now at Queens University in Kingston, Ontario. The model attempts to account for all of the known psychoacoustic and physiological aspects of the human hearing process in such a way that an electrical analog of the ear may be constructed that will simulate these effects. It is believed that the ear analyzes sounds in 1/3-octave bands spread uniformly through the audio spectrum. This behavior is supported by measurements on cats' ears (which are similar to human ears), which showed individual nerve cells respond over 1/3-octave bands with band-edge response falling off at 96 dB/octave.

The ear model begins with a broadband microphone (representing the eardrum and bones connecting to the cochlea of the inner ear) feeding a bank of 30 1/3-octave filters (the individual frequency-sensitive nerve cells). This is followed by a parallel set of 30 peak detectors whose outputs are proportional to the peak values of the signals from each of the 1/3-octave filters. The detectors have a time constant of 5 milliseconds, which means that for signals beyond a few hundred hertz, the detector can no longer follow instantaneous level fluctuations and responds only to the envelope of the signal. This is more graphically explained with an example from Fig. 1.

At 30 Hz the period of oscillation is 33 msec and the output of the peak detector closely reproduces the intensity fluctuation from the 30-Hz filter. At 630 Hz, however, the oscillations are too rapid to follow and the resulting peak detector output (beyond the initial transient) is a constant level. Peak detector outputs then pass through logarithmic loudness level sensors and a "scratch pad" memory that stores or remembers past acoustic levels for about 30 msec. These processed signals are finally transmitted to the cognitive part of the brain.

It is believed that the functions represented by the last three boxes are performed by the multiplicity of auditory nerves that terminate all along the cochlea in the inner ear and are stimulated by the sound vibrations. There are some 30,000 of these cells in each ear, and simultaneous inputs from all of them would be well beyond the processing capability of the brain. The model suggests that some 1,000 cells are assigned to each 1/3-octave band. Within the band, sensing of dynamic range of sound intensity is distributed over ten sets of 100 cells, a new set responding at each successively higher 20-dB interval. With some overlapping between intervals, this accounts for the 140-dB dynamic range of the ear. The cells making up each set perform the memory function.

It is this memory that allows the ear to discriminate between directly incident sound (remembered) and echos or reverberations (compared to original) that arrive in less than 30 msec. Intelligibility usually suffers when echos or reverbs last longer than 30 msec, as the ear has "forgotten" the original sound. (The eye also has a 30-msec memory, making movies, TV, and artificial lighting appear flickerless.) This distribution of signal processing functions among the nerve cells of the ear reduces the number of auditory inputs to the brain to a manageable level.

The usefulness of the model may be judged by how well it duplicates known hearing phenomena and on its ability to predict new human auditory characteristics that can be experimentally verified. Specifically, the model should reveal why certain types of phase shift are audible and others are not. As a first example, Mark pointed out that phase shifts in the frequency components of continuous signals, such as a square wave, are almost never audible. Examination of the model shows why this is so. Above a few hundred hertz all phase information is lost, since the constant level output of the peak detector is unaffected by phase.

At low frequencies, however, the ear should be able to discriminate phase, as the peak-detector output retains some phase information. Mark verified this, relating an experiment in which separate 200-Hz sine waves, fed into the left and right ears of a subject, appeared to originate to the left or right of center as the left or right led in phase.

According to Mark, this particular phase-detection ability appears to be a binaural property, requiring the inputs from like frequency channels in each ear to be compared at some central point in the brain. Binaural time differences as small as 10 microseconds can be detected. It is this capability that is used in the accurate spatial location of sound sources. By focusing on aural inputs that have a specific binaural time difference, the brain can concentrate on and isolate sounds, such as a single conversation at a cocktail party. It is evidently not possible to discriminate phase differences between two different frequency bands, either monaurally or binaurally, although the model seems to indicate that this should be possible. Mark is investigating this.

One rather specialized case, in which the ear can recognize phase shift in a continuous signal, was related by Mark. If the signal is a train of pulses and the phase of only one of the higher harmonics, say the 15th, is reversed (shifted 180°), the result is quite audible (as

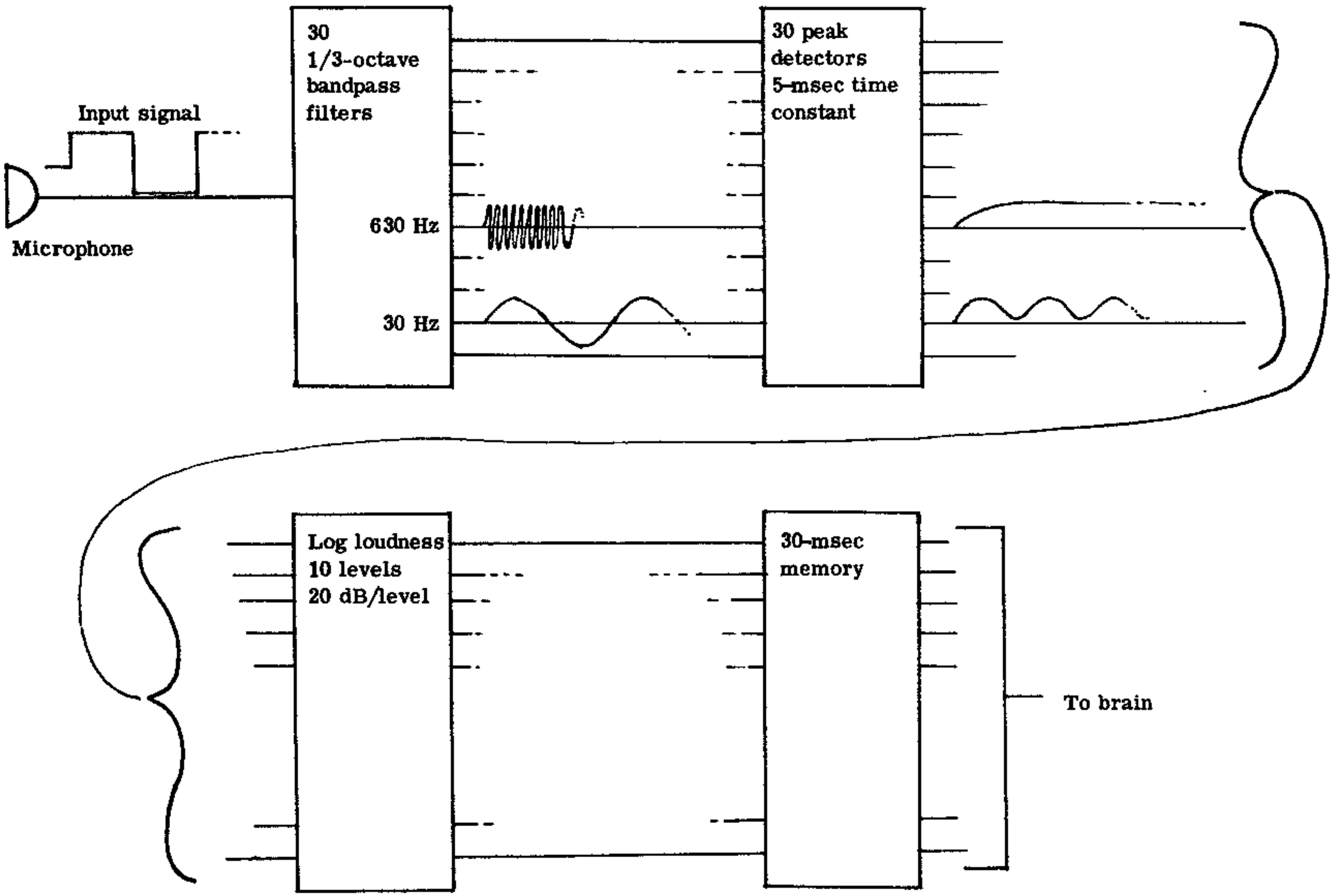


Fig. 1. System model of the ear

demonstrated on "Shop Talk" about a year ago). Simulating this same condition on the model reveals that, at the higher harmonics of the pulse train, adjacent harmonics may be close enough in frequency that two are contained in the passband of one of the 1/3-octave filters. When this happens, the peak value of the signal entering the peak detector is affected by the relative phase of the two harmonics. The changing output level of the peak detector with phase allows the brain to recognize the difference between the two signals.

Phase shifts are much more readily apparent in transient signals with low repetition rates. According to the model, the reason for this can be seen by, again, looking at the output of the peak detector. For a transient signal, the peak detector output would also be a transient, its value following the energy content of that particular frequency band. When phase shift is introduced, the energy in the frequency band over which the phase shift occurs tends to be delayed (phase lag) with respect to the rest of the spectrum, delaying the output of the peak detector (s) in that band. The delay clues the brain that a change has taken place.

In short, Mark felt that a quick test for the audibility of phase shift in any acoustic signal could be made by setting up the electrical equivalent of the model diagrammed in Fig. 1 and viewing the output of the peak detectors on a 30-channel oscilloscope display. If you can see a difference in the signals on the oscilloscope with a given phase change in the input signal, you could probably also hear the difference.

Demonstration. Dennis and Rene had set up an array of equipment for demonstrating the audibility of phase shift with various types of signals. Included were an Aries synthesizer, a phase-shifter box, pink-noise generator, Citation-Eleven preamp, Revox A77 tape machine, Phase-Linear 700 amp, and a pair of Magnepan speakers. Also present was a modified AR-3 as a reference transducer, and a storage CRT display for observing the signals. The phase shifter consisted of an all-pass network with a flat frequency response but a phase shift of 360° at 600 Hz occurring within a 10-Hz band. Rene pointed out that the effect of a phase shift is to delay the energy in the frequency components of the signal around the phase shift frequency, making them emerge later than the rest of the spectrum.

The first signal passed through the Magnepans was an impulse of low repetition rate. This was compared with a spark source built by Dennis which actually generates an acoustic doublet or a short, intense, high-pressure wave followed immediately by a short low-pressure wave. This demonstration did not involve phase shifting but allowed some judgment of the character of the listening room and the transient response of the speaker. The resonances and finite transient response of the stretched diaphragm of the Magnepan were clearly evident in the "thup" sound of the impulse as compared with the sharp snap of the spark source.

Dennis generated a repetitive, exponentially decaying sine wave with the synthesizer. In A-B comparisons of this signal before and after phase shifting, differences were most audible when the fundamental frequency was set at 600 Hz, the center of the phase shift band. This situation could be representative of a speaker in which the fundamental was reproduced by one driver and the harmonics by another with phase shift introduced by the crossover or the spatial relationship of the transducers. The signal, a damped sine wave, is akin to the transient produced by a percussive piano or drum note.

Dennis had made a tape recording of piano and drum solos in a relatively anechoic recording studio. It was much more difficult to recognize differences between the straight and the phase shifted version of these musical signals and no listener consensus was reached.

Two signals in which phase shift should not be audible are the square wave and band-limited pink noise. In the initial runthrough, however, differences were detected. These were quickly attributed to differences in levels in the two compared signals and, when corrected, the expected results were obtained.

One of the problems that has interested Rene is reducing the accumulation of phase shift that occurs in multiple tape copying. While this is due to phase shift both in the electronics and in the recording process, the latter makes the most significant contribution. The phase shift mechanism in recording, described by Rene, has to do with where sound is recorded on the tape by the head. Magnetic flux from low-frequency components tends to penetrate further into the tape and is actually recorded after the tape leaves the trailing edge of the gap. High frequencies, with their smaller fringing fields, are confined more closely to the region of the gap and are transferred to the tape at the trailing edge. On playback, sensing for all frequencies tends to occur near the center of the pickup head. Low frequencies, being physically ahead on the tape, are picked up first, yielding an effective high-frequency phase lag.

Rene demonstrated this by recording and playing back a square wave at 500 Hz. During the recording the monitor output was observed indicating no apparent degradation of the signal by the electronics. On playback, overshoot and ringing could be seen on the leading edge of the square wave. This is the expected result of a high-frequency phase lag.

Rene stated that making a copy of the tape while running it in reverse on the same machine should cancel all of the phase shift introduced by both the electronics and the recording process. Running the tape backwards causes the time sequence of events on the tape to be reversed, yielding high frequencies that lead in phase. This is cancelled by the normal phase lag introduced in the re-recording process, restoring the signal to its original condition. Mark put this technique on a more rigorous basis by using system analysis with block diagrams to show the conditions that are required for the process to be valid. Two primary conditions, flat frequency response and system linearity, can be difficult to meet in tape recording unless extreme care is taken (see the next section).

As a test of this principle, Rene had prepared a demonstration comparing program material copied forward ten times with the same material copied five times forward and five times backward. After experimenting with a number of machines, he finally chose a 15-ips half-track Revox, which he tweaked to obtain a frequency response ±1/4 dB from 25 Hz to 22 kHz. In preliminary tests with a 500-Hz square wave, he found the phase shift introduced in three forward copies was substantially eliminated in only two reverse copies. Rene could offer no explanation for this anomalous result but Mark conjectured that nonlinearities due to asperity or modulation noise could be a cause for imperfect phase cancellation.

With this recognized flaw, comparison of the ten-forward and five-forward/five-backward copies of program material, while interesting, was indeterminate. The only general conclusion reached was that the noise level at high frequencies (5 to 10 kHz) was greater in the five-forward/five-backward copy. Rene indicated he will be continuing his investigations of phase shift in tape recording and intends to report on how the problems encountered in this demonstration are resolved.

— John Schlafer

More on Phase "Error" Cancellation in Tape Recorders

If the phase cancellation technique outlined above is to be used with magnetic tape recorders to determine the possibility of audible phase error, then two full-track machines must be used, one to record and the other to play, if theory is to be strictly complied with.

To understand this, first consider what we are trying to do. We want to arrange an A-B test of music, speech, noise, or other material where the only difference between the A and B samples is the presence or absence of phase shift. We want all other parameters to be identical, including noise, frequency response, level, and distortion. Then, if there is any audible difference between the A and B samples, it must be due to phase shift.

We must assume that the level, frequency response, noise, and distortion of a signal recorded and played back is independent of whether that signal is fed to the tape machine in a forward or backward direction, but that the <u>phase shift</u> in the tape and the electronics will be added to a <u>forward</u> signal and <u>subtracted</u> from a <u>backward</u> signal. If we run the signal forward through the tape machine twice to make the A sample, we will have an amount of phase shift equal to twice the amount introduced by a single pass through the machine. If we make the B sample by running the original sample through the machine once forward and once backward, the phase shift introduced in the forward pass will be cancelled by the phase shift in the backward pass, leaving us without phase shift. Since both the A and B samples have run through the recording process twice, they have the same amount of noise, distortion, etc. This process of re-recording can be extended an arbitrary number of times, as long as the B sample is allowed to pass through the machine an equal number of times both forward and backward.

Consider the implications of this process by taking the simplest, two-pass case. We start with the original signal and record it once in the forward direction. The tape is then rewound and played in the forward direction while being recorded on another track or another tape. This second tape is then rewound and is the A sample upon being played a second time. Let's express this whole process in shorthand as $R_f P_f R_f P_f$, where R is record, P is play, and f is forward. Note that the signal must pass through the record electronics to get to the record head; hence they can be considered one unit with a net phase and frequency response; similarly for the play head and its associated electronics.

To make the B sample, we again record the original signal in the forward direction. Then the tape is reversed and played backward while being re-recorded. Note that the signal passes through the play head, the play electronics, the record electronics, and the record head all in the reverse direction. The resulting second-generation tape is then reversed so that when it is played, the signal is again in the forward direction, forming the B sample. In shorthand notation, $R_f P_b R_b P_f$, with b of course meaning backward.

If the phase in the B sample is to be exactly canceled out, the R_f and the R_b must be made using exactly the same record electronics and head. The same electronics and head must be used for the two R_f's in the A sample so the frequency responses will exactly match. Similar comments apply to the play processes. (Indeed, the same reel of tape should be used to make all dubs.)

Can this process be carried out on a single multitrack machine? Suppose the first R_f for the A sample is recorded on track one. When the tape is rewound for the first P_f it will obviously be played by the track one play head. Since the record head precedes the play head, the output from the track one play channel must be recorded on some other track. But this violates the condition made above that all recording be done with the same record head and electronics. * So a single multitrack machine cannot be used.

Suppose then we use a pair of multitrack machines, one to do all the recording and the other to do all the playing. Say we make our first R_f recording on track one of the recording machine. We can then rewind the tape, play it on track one of the play machine, make a second recording on track one of the recording machine and play it back again on track one of the play machine, making the A sample. So far, so good.

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*I am using the term "record head" to refer to a specific track, so that a stereo machine, for example, would have a left record head and a right record head, even though those two "heads" are in the same physical package. The important point is that they and their associated electronics would be similar, but not identical, as the theory requires.

To make the B sample, we must again use track one of the recording machine, since we used it to make the A sample. The tape is then reversed and played on the second machine. However, because the tape has been reversed, it will no longer be played by the track one playback head, as was done to make the A sample. In a two-track stereo machine, it will now be played by track two; in a four-track machine, by track four, etc. This again violates the conditions for the test.

Only by using a pair of full-track machines, one for recording and one for playback, can we ensure that the signal is recorded by the same record head all four times, and played back by the same play head, regardless of whether the tape is played forward or backward. Of course, the full-track format makes it impossible to A-B at will during playback. We must either play each sample through in its entirety, while trying to remember what the other one sounded like, or we must choose our A-B points ahead of time and splice the two final tapes.

In preparing and presenting his demonstration of the above process at the August BAS meeting, Rene Jaeger compared the results of ten forward passes against five forward and five backward passes with a variety of material, but made the mistake of using two two-track machines. Comparison of the A and B samples yielded a mismatch of several dB in the high-frequency response. Rene stated that the machines were flat to within a phenomenal 0.25 dB. Even so, it does not seem unreasonable to assume that at one or more points in the frequency spectrum, one track was up by 0.25 dB while the other was down by 0.25 dB. After ten passes, one sample would be up by 2.5 dB while the other would be down 2.5 dB, making a difference of 5 dB, which would certainly be audible. Ergo, the importance of using the same record and play system throughout the experiment. With differences in frequency response of that order, no conclusion can be reasonably drawn about the possible audibility of nonlinear phase response in tape machines on the basis of the August demonstration.

— Mark Davis

B.A.S. Test Reports

Alvin Foster, Larry Hardin, and Mark Davis

A Preliminary Look at the Koss Electrostatic Speakers

The BAS's guest lecturer at its February 1975 meeting was Howard Souther, Senior Vice President of the Koss Corporation and the person most responsible for their new full-range electrostatic speaker.

Mr. Souther gave an excellent description of the speaker's design while simultaneously tantalizing our Audio appetites. Physically, the speakers are just as he described them. They weigh an incredible 100 pounds each, and appear much larger than their measurements (4 feet high by about 2½ feet wide by 11 inches deep) would indicate. They are very handsome and cost a mere $1500 per pair.

This is a preliminary report because I was able to listen to the speakers only in a dealer's showroom, using their electronic equipment. However, I was using familiar musical material and my listening session lasted about an hour. This report also may be skewed because rumor has it that Koss plans to distribute the speakers about the country, collect feedback on their apparent sound quality from audiophiles, and then redesign the speakers to be free of such faults as are uncovered.

The Koss speakers will be difficult for most of us to review in the home because of their size and weight; I doubt that the speaker will fit in most station wagons. If you happen to receive a defective pair and you are forced to return them to the factory, throw away your wallet: The shipping cost would be astronomical.

With these reservations in mind concerning auditioning limitations, my overall report on the speakers' sound quality is that they have potential—lots of it. The speaker is handsome and presents a good bipolar sound stage while avoiding 10-foot-wide piano images. Here is a rapid rundown of my impressions:

1) The speakers are beamy. This was confirmed after passing interstation FM hiss through the speakers and walking back and forth in front of them. The high-frequency sound quality varied tremendously with listening position, more so than with the Dahlquist or KLH-9.

2) The speakers are equally directional vertically. Their sound character changes dramatically when you sit down or stand up in front of them. It is as if you are auditioning two different sets of speakers. This seems to be most noticeable in the midrange, 500 to 1500 Hz. When you are sitting directly in front of the speakers, the midrange seems to dominate. However, when you are standing, the midrange may seem a bit thin.

3) The speakers lack very low bass below 60 cycles. The opening organ pedal of "Also Sprach Zarathustra" was lacking in depth. However, the Koss worked deeper into the bass than the KLH-9, another bipolar design.

4) The speakers have an exaggerated lower mid-bass hump around 150 cycles. This frequency response area is heavily affected by room placement. Perhaps if I had had the opportunity to move the 100-pound speakers around the room, this defect could have been eliminated. My guess is that the mid-bass hump is real. It did add excitement to the opening heart beat of Pink Floyd's "Speak to Me," but for most music it is unnatural.

5) The speakers have a peak around 5000 Hz. This causes them to exaggerate the effect of cartridge mistracking and tape hiss. As this defect also emphasizes record pops and ticks, it leaves you with the impression that all of your records and tapes are grainy and belong in the trash heap.

In summary I feel the design approach of the Koss is quite innovative and is destined to be copied. And I am sure that after collecting feedback from audiophiles around the country, Koss will eventually introduce a Model II that probably will be the speaker to be reckoned with.

— Alvin Foster

Tuner Comparisons

I have often wondered how FM tuner specs translate into audible performance in my area (Syracuse, New York). Spurred by the tuner reviews in the December 1974 and April 1975 issues of the BAS Speaker, I decided to slouch toward some of my own interim conclusions. With the cooperation of my friendly local Tech Hi-Fi dealer, I was able to compare the Sequerra One, the Kenwood 700T, the Kenwood KT-8007, the Citation XV, and my own "golden oldie," a McIntosh MR67 (recently aligned).

The terrain in Syracuse is hilly, and I am fortunate enough to live fairly high. In my tests, I used an eleven-element Winegard antenna, a Radio Shack two-set coupler, Infinity Servo-Statik Speakers, and Stax SRX headphones. Tuner output levels were carefully matched for A-B comparisons, and on one occasion, I brought in a critical friend for a blind comparison. The reference stations were WONO, a classical music station with a strong, multipath-free stereo signal at my location and generally careful station practices (low average modulation, 0.6% THD at midband, board-to-antenna, etc.) (owned by Charles River Broadcasting who also own WCRB); WBFB in Rochester, which I receive with an average signal strength of between 30 and 50 microvolts; WAER, a 12,000-watt station whose transmitter is only five blocks away; and some weak 2- to 20-µV signals from Albany, Ottawa, Hamilton (Ontario), and other stations between 100 and 200 miles distant. There was a wide variety of program material on records, first-generation tapes, duplicated tapes (BSO, etc.), and two live studio broadcasts of bluegrass music.

On local broadcasts, the tuners were virtually indistinguishable. The McIntosh sounded slightly different from the others at the high end (test reports have shown it to be down 3 dB at 15 kHz) with slightly less separation overall. Beyond that, I could detect no differences at all in noise, frequency response, or distortion. The only SCA transmission in Syracuse is carried by a mono Muzak station of execrable quality, so I could not test for SCA rejection. All of the tuners seemed to have satisfactory multipath rejection. The multipath indicator on the MR67 is just as insensitive as the one on the later MR78. The Kenwood and Citation devices are better, but a scope is best.

Now why do reviewers like Feldman and Holt hear differences between, say, the Sequerra and other tuners that I don't hear? Are their ears finer? Probably. But I doubt that most audio buffs would experience things differently from me in their own serious listening. I incline to think that the Emperor wears a G-string.

But some differences did emerge beyond the local service area, differences in stereo noise, quieting slope, selectivity, and capture ratio. Here the Citation was a clear loser. It could not get WBFB, Rochester, in stereo. The other tuners always got WBFB in stereo. Generally, the Citation just couldn't hack fringe work, so it was disregarded in what follows.

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All of the tuners successfully received CKWS (4 AV), separating it from WCMF (20 µV) 200 kHz away. All except the McIntosh could receive WRVO (5 µV) at 88.3 free of background program emanating from WAER (12,000 watts, five blocks away) at 89.9, with the antenna pointed toward WAER. Three other distant stations (more than 100 miles) are on frequencies close to strong local stations. The McIntosh could not receive them intelligibly, but the Sequerra and both Kenwoods were successful in obtaining a clear signal.

In general, any task that the Sequerra could perform, the Kenwoods could perform equally well. The McIntosh matched the others except under extreme conditions. Similar specifications did yield similar results. The least important specification seems to be harmonic distortion if that is already acceptably low (about 0.5% in stereo) and the tuner satisfactorily excludes 19-kHz and SCA products.

The supertuners ought, I think, to be viewed with some skepticism in the light of the strong performance of medium priced equipment like the Pioneer TX 9100 (given a good sample) or the Kenwood KT-8007. The Kenwood 700T has a sexy front panel (by comparison, the KT-8007 looks a bit like a dowager), a fancy tuning system (which requires the patience of Job to use), and a $750 price tag (the KT-8007 costs $425). Otherwise the specs are the same as the 8007, along with most of the circuitry.

The aristocratic tuners are nice to play with and think about. I like the 700T's individual performance graphs and nothing in audio is more fun than the Sequerra. But I really doubt that they give us better meat-and-potatoes than bourgeois Pioneers and Kenwoods. Spend the difference on a good antenna system. (But if Mr. Barrett is correct, buy the Sequerra, sell the Winegard, and purchase good rabbit ears!) — Larry Hardin

Comparison Test: Tandberg 9100X and Sony 377

A Tandberg 9100X tape deck was tested to see if its crossfield bias arrangement made an audible difference in its recording ability. Use of the crossfield head is supposed to result in less phase shift at mid frequencies, which might or might not be audible. It has no effect on playback, so that a tape made with a crossfield head should in theory sound better than an ordinary one, regardless of the playback machine. For comparison purposes, a Sony 377 was tested, being chosen because it's a good machine with a conventional bias arrangement, and because it happened to be handy.

In order to avoid having the test influenced by frequency response, each machine had its record and play responses carefully adjusted. Overall, the Tandberg was flat within ±1.5 dB from 35 Hz to 25 kHz, the Sony from 26 Hz to 27 kHz with the same tolerance (see Fig. 1). The Sony was quieter by a few dB, with a 62 dB weighted S/N versus 57 dB for the 9100X, but the Tandberg exhibited less high-frequency saturation for a given flux level on the tape. The two effects effectively canceled each other. Both machines were quiet. Both seemed to have similar harmonic distortion levels, although this was gauged only approximately by listening to a 1-kHz sine wave being recorded and reproduced. Both machines exhibited audible (barely) distortion of the sine wave 3 to 5 dB below Ampex standard level, which was not in evidence 7 to 9 dB down. Flutter was inaudible on both machines.

Unfortunately, to measure the phase shift of a tape machine, one needs some fairly high powered gear, such as a precision delay line with no phase shift of its own, or a computer with a fast Fourier transform program. Lacking such amenities, I attempted to roughly gauge the phase error introduced by observing square wave responses across the frequency range. Both machines introduced small amounts of tilt, overshoot, and rounding to the square waves at various frequencies, but there was really no frequency where the waveform coming out of one machine could be called noticeably better than the other, with the possible exception of the deep bass, where a sharp cutoff filter in the Tandberg at 33 Hz caused the waveform to look considerably less like a square wave than the output of the Sony, but the effect was not audible and in any case has nothing to do with the crossfield head.

Having failed to establish any clear difference by measuring the two machines, I fell back on listening to them. Source material included the Sheffield III disc and the very demanding AR demonstration disc, ENY-1. Care was taken to keep the levels on the tape identical from one machine to the other, which was a little tricky since the Tandberg has frequency-weighted peak-reading meters that cannot read the level on the tape while recording, while the Sony has flat, average-reading meters that can.

How did they sound? After spending several hours listening to each machine via speakers and electrostatic headphones, I was able to conclude that each machine produced a copy that was extremely close to the source in quality, but I was frankly unable to say for certain that the 9100X sounded better than the 377. They were just too close. Perhaps with the right combination of speakers, room, source material, and tape it might in theory be possible to discern a difference, but with the equipment used here, the machines were well nigh indistinguishable.

It might be noted that both the Tandberg 9100X and the Sony 377 seem to be well-engineered machines. The $900 Tandberg offers excellent performance coupled to a smooth three-motor transport. The Sony has to get along with only a single motor, but at $400, it can be considered something of a bargain.

I would like to thank the people at Atlantis Sound in Harvard Square for the use of the Tandberg 9100X.

— Mark Davis

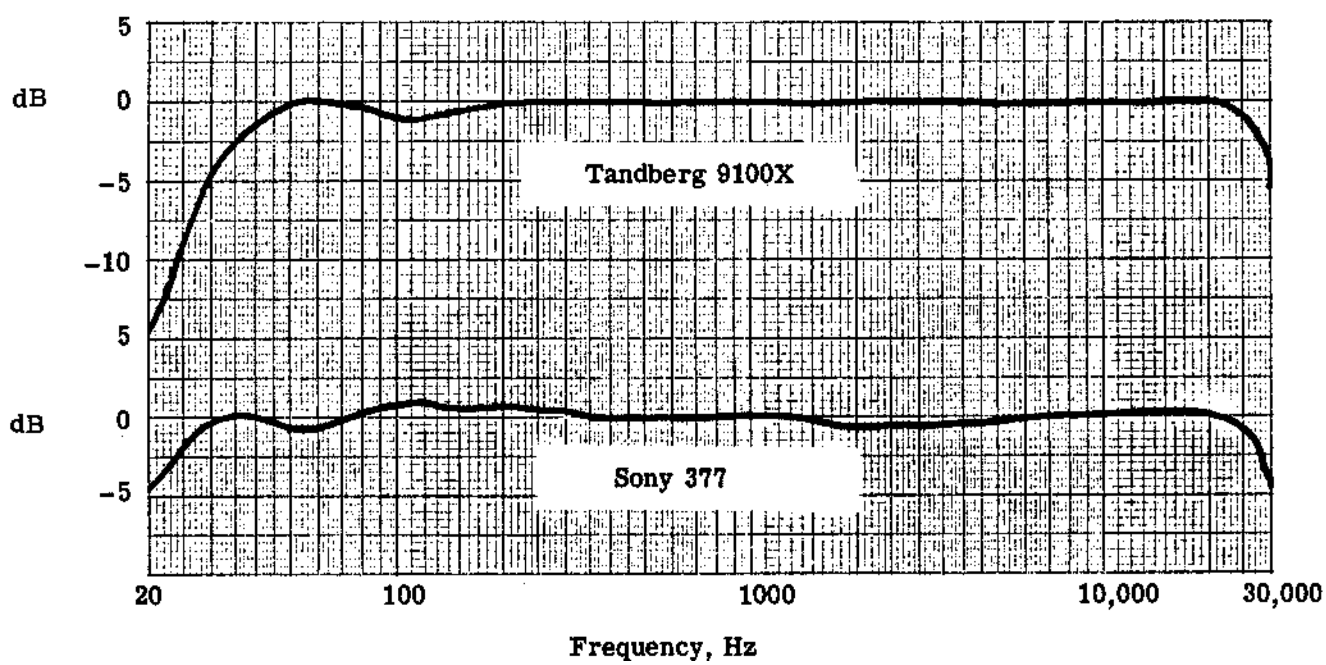

Fig. 1. Record-playback response curves at 7.5 ips

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Membership Preference Questionnaire—September 1975

To help the executive committee plan programs and select publications for the coming year, please fill out and return this questionnaire. Below is a list of each of the programs and publications for the past fourteen months, the period since the last questionnaire was circulated. Please grade each item with the appropriate letter:

E Excellent, G Good, F Fair, P Poor

There is space at the end of the questionnaire for extended suggestions and comments, and all are welcome. Space is also included for you to list your name if you are interested in writing material for the Speaker or if you are willing to volunteer for any other BAS activities.

Publications

Test Report: Cassette vs. Cassette vs. Open Reel: Advent 201, Lafayette RKD-50, Sony 152-SD, and Revox A-77, by Davis

_____ In Defense of the Piano, by Cote

IC Op Amps—The Audiophile's Friend, by Mitchell

_____ Test Report: Scotch Classic and TDK Audua—How the New Open Reel Tapes Compare, by Foster

A Vacuum System for Cleaning Records, by Borden

Strategies for AB Listening Tests of Audio Components—Imaginary Witches for Real-Life Glitches?, by Shanefield

_____ User's Report: the dbx 122, by Brinton

_____ SCA Interference, Cause and Cure, by Southwick

User's Report: The Burwen DNF-1200, by Brinton and Cohen

_____ The Role of Damping in Tonearm/Cartridge Performance, by Phoenix

White or Pink? Adding a Little Noise to Your Life—How to Build a Noise Generator, by Jaeger

_____ A Quasi-Complementary Discussion of Microphones (Including the Thermo-Electron 814), by Mitchell

Son of a Witch-Glitch Switch, by Shanefield

A Case for the 814, by Southwick

Improving the Thermo-Electron 814 Microphone, by Mitchell

Test Report: Tuners: Pioneer TX-9100 Versus McIntosh MR-78, by Foster

White and Pink Noise Revisited, by Jaeger and Southwick

Improving the Performance of the AR Tonearm, by Phoenix

Feedback on Phono Noise—Micro-Acoustics Versus Shure, by Zwicker

Making a Compact Headphone Amplifier — (or Two), by Mitchell

Audio Myths, by Shanefield

Phase Distortion and Transient Response, by Colin

Comments on Records, Cleaning, and the Discwasher, by Zwicker

Test Report: The Allison:One Speaker, by Brinton et al.

A Transformerless Balanced-Line Preamp for the Phantom 814 Microphone, by Mitchell

Using the BAS Oscillator, by Mitchell

Meeting Programs

Tom Horrall, BB&N: Description and demonstration of BB&N concert hall acoustics simulator

John Draper and Dave McIntosh, Epicure Products: Discussion and demonstration of Epicure products,

Dick Goldwater and David Ranada: Comparisons of recorded performances of music

_____ Roy Allison and Dick Burwen: Demonstration of Allison:One speaker and discussion and demonstration of Burwen dynamic noise filter

Panel discussion on dealer/customer interface
Arnold Schwartz, Micro-Acoustics: Discussion of disc cutting and discussion and demonstration of QDC-1 cartridge
A-B equipment comparisons: Damped vs. undamped arms, QDC-1 vs. XLM, Marantz 7C vs. SAE, tape vs. disc
Howard Souther, Koss Corporation: Discussion of headphone design
Rene Jaeger, dbx: Why the audiophile cannot get through the maze
Bob Tucker and Ed Laurent, Dynaco: Discussion of Dynaco's product line, particularly the new preamp and power amp and mini-clinic on amplifier distortion
Ron Dunlap, Dunlap-Clarke Electronics: Discussion and demonstration of the relationship between loudspeaker impedance characteristics and power-amp design-requirements
Fred Barrett, Sequerra Company: Discussion and demonstration of the Sequerra broadcast monitor
Sam Walinsky, Hybrid Systems: Demonstration of delay line
David Ranada and David Satz: Comparison of musical performances
Denis Colin, Rene Jaeger, and Mark Davis: Demonstration and discussion of audibility of phase shift through various hi-fi components

Newsletter Features

BAS Meeting summaries
Audio Engineering Society meeting summaries
Used equipment for sale or wanted
Notices of discounts or bargains
Industry news
Criticisms of industry practice
Membership directory
Technical notes
In the Literature
Table of Contents (annual listing)
Letters from members
Equipment survey

Would you like more or less of the following:

Use reports (qualitative material)
Test reports (stressing more quantitative material)
Designs for do-it-yourself projects
Articles to educate you in audio electronics
Articles to educate you about musical topics
Letters from members
Industry news

If you are willing to do any of the following, please sign your name and indicate your interests:

I am willing to pick up passengers at BU for meetings at GTE Labs.
I am interested in writing material for the newsletter.
Type of material:
I would like to work on the newsletter (meeting summaries, coordinating editor, distribution, mailing, copy editing, etc.)
I would be interested in volunteering for some other BAS activity, such as

What things is the BAS doing that you like most?

What things should the BAS be doing that it is not?

What things is the BAS doing that should be done differently?

Suggestions for future meeting programs:

Suggestions for future BAS Speaker publications:

Please make any additional comments or criticisms you feel appropriate.