THE B.A.S. SPEAKER

THE BOSTON AUDIO SOCIETY
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IN THIS ISSUE.. .

After reading this month's meeting summary and short articles on tonearm damping, you may reach the conclusion that STP is the "Audiophile's Edge." Indeed, tonearm damping appears to offer a high return on the investment of a little time and material, and if you don't like it, no damage done—so long as you don't spill the STP.

In addition to tonearm damping, we have a short note by Tom Holman on the latest audiophilic anxiety, transient intermodulation (TIM) distortion. Read this before you decide to trade in your old equipment. We have still another simple, inexpensive method to detect and minimize FM multipath distortion. And California member Nate Garfinkle has some interesting comments on cartridges and records.

Dan Shanefield's article on A-B comparisons promised for this issue of the Speaker will definitely appear next month. In the meantime, the summary of the January meeting points out some of the pitfalls and difficulties one can encounter in such endeavors.

The Compleat Microphone. In this month's article, Peter Mitchell presents us with a complete primer on microphone characteristics and specifications. Peter explains the relationships among sound pressure level, mike output level, system voltage level, and impedances. This is followed by a review of the Thermo Electron 814 mike (an outstanding value) and some suggested miking techniques.

GROUP PURCHASE OF THERMO-ELECTRON MIKE CAPSULES

These capsules, much talked about at recent meetings, and described this month in Peter Mitchell's article, may be ordered by BAS members in two versions: the 814 (which has a 120-dB-SPL overload capability and requires a 1- to 20-volt power source) and the 814C (which overloads at 145-dB SPL, but requires an 8- to 30-volt supply—note that in such cases higher supply voltages generally are better than lower ones). The price per capsule should be less than $30, but the exact amount will depend on the number ordered and other factors. If you plan to order, do so quickly, being sure to specify the version you want, and also being sure to include a check or money order for the estimated total cost. On delivery, the BAS will either rebate any overpayment or request the incremental difference. Checks should be made payable to the Boston Audio Society and mailed to P.O. Box 7.

VOLUNTEER WANTED

Some BAS members are visually handicapped, making the reading of the Speaker either a hardship or an impossibility fur them. One or more volunteers are therefore wanted to read the Speaker onto a cassette in "talking-book" style. It appears that the users of such cassettes might be able to pay a small fee for reading, so if anyone has the inclination to perform such a service, would he or she please notify the Executive Committee in care of Box 7. We will pass on your offer, and thank you.

UP AGAINST THE IVY WALLS

Times are changing at WBUR, and for the worse. For the past several years, Boston University has been subsidizing the so-called "listener-sponsored" programming of WBUR-FM. As of about July 1, 1975, the $100,000 or so yearly transfusion will become a thing of the past and WBUR will be thrown on its own resources. Thus, "listener sponsors" now will have to begin making the sort of contributions that they always should have.

Only about four percent of WBUR-FM's listeners support it with contributions. If only a fraction of the free-riding 96 percent would chip in, all of the station's fine music and innovative jazz programming could continue. As is, contingency plans already are being made for "evening only" programming, as well as for less drastic cutbacks in air time if finances permit. But the mood is one of retrenchment.

This is a frank fund-raising pitch. BAS members in the local area owe it to themselves to become listener sponsors for $12 to $20 a year. WBUR is about to get a new, high-quality transmitter, with Dolby-B capability, and thus could be broadcasting some of the cleanest sound in town by the fatal thirtieth of June. Programming on the way includes master tape broadcasts by BAS members Peter Mitchell and Dave Letterman similar in intent to "Adventures in Sound." But it will all stop when the money runs out. If you haven't subscribed, or if you know someone who listens and has not, either subscribe or exert some peer-group pressure, but get out the bucks for WBUR.—Jim Brinton

EQUIPMENT FOR SALE

- Advent Dolby-B noise-reduction unit, Model 100. Works fine; has mike inputs. $150. E. Bernier; call evenings at (617)631-8971, or days at (617)321-7300. Leave message.

- Cassette decks, each used less than ten hours: Tandberg TCD-300, $300; Teac 450, $300. KLH loudspeakers: one pair of original Model Sixes at $75 each; one pair of Model Fives at $125 each. Electro-Voice Interface-A loudspeakers; one pair, $250. Burwen noise filter, DNF-1201, $200. Call Will Henry, days, at (617)661-9500, ext. 328.

SAN FRANCISCO IHF SHOW

As forecast in the December Speaker, the forthcoming West Coast IHF Hi-Fi Exposition will be combined with a series of rock concerts in the 6000-seat San Francisco Civic Auditorium, from April 1-6. If anybody attends and manages to hear anything interesting, let us know.

SOMETHING GOOD AT RADIO SHACK

During the last three years many of us have discovered how handy Radio Shack's Sound Level Meter is for comparing live to reproduced music levels, for educating our ears to the difference between subjectively and objectively measured levels, and for various measurements. For the last year and a half the price of the meter has been $50, and BSR has recently marketed an identical unit for $60. But Radio Shack has cut the price to $40 until March 2, so grab one before the sale ends. (For further information, it was reviewed in Audio in July 1971 and in Stereo Review in August 1972.)

Tonearm Resonance

In his excellent note last month on the use of the damped Decca tonearm, Al Foster suggested that when the cartridge/tonearm resonance occurs at 13 Hz, this is "a very dangerous place to have oscillations." In my opinion this is incorrect. If we have to have a subsonic resonance at all (and we can't avoid it as long as our tonearms are not properly damped), the best place for it is at a frequency where minimum stimulus of stylus motion will occur. Various studies have shown (August 1973 Audio, for example) that the range from 10 to 20 Hz best fits this requirement, and 13 Hz is a fine place for the resonance.—Peter Mitchell

Phono Preamp Noise

There was a bit of confusion in last month's meeting report, concerning how the impedance of a phono cartridge affects the signal-to-noise of the phono preamp. By coincidence, the formula for computing the noise was printed in the same issue, in Rene Jeager's noise generator article. The theoretical minimum noise voltage in any preamp is determined by the impedance, Z, of the signal source connected to it. Over a 15-kHz bandwidth, the noise voltage is $V_n = 0.016\sqrt{Z}$ microvolts. Now in a typical phono cartridge, the impedance varies with frequency, consisting of a 600-ohm dc resistance plus the impedance of the coils. The latter is $Z = 2\pi fL$, where L is the coil inductance in henries (typically 700 millihenries, or 0.7 henry). This makes the computation of the noise complex, but to illustrate the principle involved, consider the contribution at 10,000 Hz. At 10 kHz the impedance of the cartridge is about 6.28 x 10,000 x 0.7 = 44,000 ohms. This is 11 times greater than the purely resistive 3900-ohm impedance claimed for the Micro-Acoustics QDC-1E, with which the preamp noise would be a factor of Ain = 3.3 lower (i.e., 10 dB). The real noise reduction would be less than this nominal value (for example, real circuits have other noise besides that due to the source impedance), but the improvement may still be significant.—Peter Mitchell

IS TIM DISTORTION AUDIBLE?

The February issue of Audio includes a feature article on transient intermodulation (TIM) distortion (which has previously been discussed in the AES Journal by M. Otala). TIM occurs principally in loud high-frequency passages. It is said to result from the modern practice of designing amplifiers to have extremely high open-loop gain and large amounts of negative feedback, which then requires stabilizing "compensation" to prevent ultrasonic oscillation. The usual lag-compensation technique reduces the open loop bandwidth and creates the possibility that intermediate stages in an amplifier can overload on fast transients. It has been suggested that this may be responsible for audible differences among amplifiers.

An interesting experiment to perform on the audibility of TIM is to build two amplifiers that are identical except for compensation technique. In a recent test, two amplifiers were so arranged. One had a slew rate capability equal to the most inexpensive kinds of amplifier designs, while the other was more than ten times faster. Instrumentation did show the difference, but in an A-B comparison, my ear was not able to hear any difference on several kinds of available source material. This test is certainly not concluded to be definitive, but it may be instructive to reproduce Otala's psychoacoustic experiments for confirmation.—Tom Holman

[Ed. Note: Op-amp IC's such as the popular 741 are said to be particularly susceptible to TIM distortion, so Rene Jeager has searched for it in some op-amp circuits employing 741's. He did not find it. Is TIM an overrated phenomenon, or is it just hard to identify? Time will tell.]

Simple Multipath Distortion Detection Rediscovered

Recently, after adding some rear channels for ambiance effect, I became aware of some obvious distortion in the rear center-channel speaker when listening to an FM broadcast. This speaker was connected so as to produce the difference signal between the main left and right channels (left minus right). When I tuned to another station, the distortion dropped below perception. Killing the rear channels altogether, I listened to both stations in normal stereo and could detect a much less noticeable but higher distortion on the first station than on the second.

What had, in fact, occurred was a rediscovery of a multipath distortion detection scheme that works for any stereo FM system. In his April 1973 article "Adding a Null Switch to Your Stereo," Peter Mitchell pointed out that multipath detection was a major feature of the null switch's repertoire. For those who don't have the article, the null switch is simply installed on any amplifier that has a common speaker ground. Join the ground side wires from your left and right speakers and connect them to one post of a single-pole, single-throw switch. Connect the other post of the switch to the amplifier's speaker ground terminals. When the switch is open, the signal to both speakers, which are then in series, is only the difference between the left and right channels. On an FM-stereo broadcast, this is the multiplex decoder output without the main signal.

Peter suggests adjusting the antenna for maximum left-minus-right amplitude and minimum distortion. I found that serious misalignment on a weak station sometimes gives an increased albeit distorted (difference) signal, so concentrate on lowest distortion.

Since there is no multipath indicator or oscilloscope output on my receiver, rediscovering the null switch method saved it from a major operation at the last minute and vastly improved the sound on at least one important station.

I was, however, able to try Harry Zwicker's multipath vertical output listening method (BAS <u>Speaker</u>, January 1975) on a friend's Pioneer TX-9100 tuner and found a high-volume rumble obscuring any variation with antenna rotation on WGBH-FM using my main antenna. With a split wire loop antenna, the system worked well. Very strong signals may have saturated the automatic gain control, lowering the multipath sensitivity. But the null switch worked equally well with either antenna.—Joel Cohen

MORE ON TONEARM DAMPING

Soon after modifying his SME tonearm to include the simple "dashpot" damping scheme discussed in last month's <u>Speaker</u>, Bob Graham wrote to SME Ltd., asking for the company's opinion on such things. He quickly received a reply from Managing Director A. Robertson-Aikman, the pertinent portion of which is included below. On reading it, one is pleased with the courtesy of the letter, but wishes that SME could have made the reasons for its position more specific.

"In designing a cartridge, its compliance has to be limited according to the effective mass of the stylus and its cantilever. The weight of the cartridge (tare) is usually dictated by its electrical output. There must be a proper relationship between these and other things. If, for example, the cartridge has a very high compliance and a high tare, the stylus-compliance/arm-cartridge resonance maybe unacceptably low, even with a low-mass arm, giving rise to the type of problem you have observed. [i.e., poor performance on warped records—Ed.]. Damping a pickup arm to meet this situation is critical and degrades its performance in other [unspecified—Ed.] areas. For this reason, we do not offer it."

By way of non-damning criticism, we must point out that in his January 1973 review of the SME 3009, Series II Improved arm, John Wright of Hi-Fi Sound (Great Britain) noted:

"We have sometimes stated that a small amount of damping can be an advantage in increasing [arm/cartridge system] stability and reducing the Q of low-frequency (arm/cartridge) resonances. The disadvantage [of damping] has been that it can be a messy, inconsistent operation to undertake during manufacture and presents problems of sealing the damping [fluid] in the bearings during transit. Also, it can make the arm feel stiff, which many might mistake for friction. (Damping, of course, should not be confused. . . with friction where the maximum force is required to get the arm into motion, and once in motion, it runs fairly freely. With damping, negligible starting impedance is presented; it merely resists movement against sudden shock waves or low-frequency resonance.) However, with the SME Improved, the greatly reduced effective mass does seem to largely negate the need for additional damping with even the best of current cartridges, and the writer has yet to investigate what improvement if any, might be brought about by lightly damping the bearings."

In addition, in a review of the 3009, Series II Improved arm initialed by B.J.W. in the British magazine Hi-Fi for Pleasure, May 1973, the reviewer notes:

"SME have never damped their arms, save to special order, and nothing in our experience. . . suggested any need for it, whereas both the ADC and Ortofon (cartridges) demanded addition of damping to the previous model [the 'Unimproved' Series II]. We confess to wondering what effect might be produced by the careful damping of the new model."

At the BAS, we think we have found out. Damping does improve performance of the SME arm when carefully—even if crudely—applied. See this month's meeting summary, the notes by Bob Graham and Al Foster in last month's Speaker, and the remarks on damping in this issue.—
Jim Brinton

Good Grief. . . STILL MORE ON TONEARM DAMPING

Even after the notes in last month's Speaker and the demonstration of the effectiveness of viscous damping at the January meeting of the BAS, it seems that still more must be said on the subject of tonearm damping. Not only is the topic controversial (see the opinion of SME's managing director elsewhere in this issue), but also, some of those present at the demonstration appeared uncertain of the results and advantages of the technique.

Those of us who have experimented with viscous damping our tonearms feel that it may be the least expensive modification an audiophile can make to an existing system that can significantly improve performance. This is not to say that any high-mass, high-friction arm turns into a wonder with the addition of paddles and STP—nothing we have said should be construed to favor poor arm design—but if you already have an arm of reasonable quality, like the AR arm, its performance and that of your cartridge, can be upgraded. What follows is a series of analogies (mostly) showing perhaps a little more clearly than before why such improvements occur, and incidentally, noting some improvements we weren't fully aware of when last month's Speaker was put to bed.

First, some basic premises:

• No record is physically flat.

• A stylus-cantilever assembly should respond only to record groove modulations—any other source of cantilever movement causes distortion.

• Damping is not a cure for poor arm design—low mass and low pivot friction still are desirable.

Second, some of those at the demonstration last month appear to have been left with the idea that damping's sole benefit lies in improved reproduction of warped records. In fact:

• Damping controls (and can almost totally eliminate) unwanted large excursions of the stylus cantilever in response to warps, or to arm oscillations caused by shock or vibration.

• Damping makes some very badly warped records playable for what may be the first time.

• Damping seems (subjectively) to reduce playback distortion (perhaps formerly caused by uncertain stylus/groove-wall contact).

• Damping seems to dramatically solidify the stereo image.

• Low-frequency reproduction is (subjectively) greatly enhanced through damping, probably because of the controlling effect damping has on subsonic arm/cartridge resonances, or perhaps because of elimination of occasional preamp overload due to these subsonics. The Q of such resonances is dramatically reduced with appropriate damping.

• Damping tends to simplify the problem of arm/cartridge mating. Problems of incompatibility due to the subsonic resonances just mentioned are greatly eased.

That's a summary of the benefits uncovered to date by about half a dozen BAS members who have experimented with tonearm damping over the past months. So far, no disadvantages have been uncovered; although it is possible to overdamp or to underdamp (the results are obvious and correctable), or to spill the damping fluid, simple care and sweet reason are more than adequate to assure near optimum performance.

But <u>why</u> and <u>how</u> does simple tonearm damping accomplish the feats listed for it? Here come the analogies.

In many ways, a phono cartridge operating in an undamped tonearm is similar to an automobile operating without shock absorbers. The feeling is by turns jarring and floating as the car and occupants overreact to dips and bumps in the road. Reduced greatly in size, but greatly <u>increased</u> in relative intensity, much the same thing happens with a phono cartridge in an undamped arm.

Autos are generally less than five feet high these days, or about 60 inches. We all know how it feels to ride over a one-inch bump or dip, even with shock absorbers. And most cars float or sag their way even through gradual dips or humps of the same height. Now consider that the average phono cartridge is about three-quarters of an inch high and compare this to the average record warp. Today's records simply are not flat, even though they may not appear obviously warped; if your worst record has a peak-to-peak warp of only one-eighth of an inch, you are a lucky audiophile.

To a phono cartridge, the car's easily felt one-inch hump scales down to a "warp" less than one-sixty-fourth of an inch high (0.015625 inch). Thus tracking the typical disc must be almost the equivalent of a motion picture car chase through San Francisco's hills.

The auto/phonograph analogy holds up well. The best riding and handling cars have what engineers call "low unsprung weight" (substitute low effective tip mass), attempt to keep their wheels in firm contact with the road (read: highly conformal tracing of the groove by the stylus), and so far as possible all decouple the effect of the mass and movement of the body and frame (tonearm, cartridge body) from that of the wheels (stylus assembly) through the use of shock absorbers (viscous damping—literally).

After all this analogizing, it should be clear that cartridges in damped arms will have an easier time tracing warped records. But what about the other claims made for damping—the solidified stereo image, reduced playback distortion, etc.

These effects follow naturally. Because damping tends to eliminate continuing oscillations at the natural frequency of the arm/cartridge combination, the stylus assembly is going to be subjected to far less unwarranted movement, and this is where the dividends are paid.

A magnetic phono cartridge is like most other electromagnetic transducers in that it has a region of most linear operation. Stylus pressure is—or should be—specified with more than groove-wall contact in mind. It is that range of tracking forces (IT) within which the cartridge moves about its optimum position; thus T F is a dynamic parameter, not just a static one because the movements of the tonearm sideways and up and down can lower and raise TF by relatively large amounts, or reduce TF on one groove wall while increasing it on the other. With a bad enough warp, for example, the tonearm will continue to rise after the stylus has crested the warp and begun to move downward, thus creating an instant of very high TF followed by one of very low TF. If the extremes are great enough, as with a badly warped disc, or with a massive or high-friction tonearm, the cartridge will bottom out from too much pressure or be dragged upward out of the groove-

Some variation of this scenario is played out every time a disc is played. Normally the stylus isn't thrown out of the groove, but there can be repeated momentary losses of groove-wall contact because of temporarily low T F. Or distortion may result as the cantilever is pressed into the body of the cartridge and in turn moves the "moving magnet," "moving coil," or "variable-reluctance" element outside of the transducer's linear operating range. By keeping the system from flailing about, viscous damping helps prevent such distortion.

To help you better grasp the effect of wide cantilever excursions on cartridge linearity, think of a loudspeaker forced to operate with an amplifier having a dc offset. Since some current is always flowing through the speaker, it will be somewhat in front of or behind its normal "at-rest" position. Thus when a high-level signal comes along, the voice coil will either move outward, away from the well-controlled portion of the motor's magnetic field and distort its output, or be pulled too far back into the motor, perhaps bottoming, and at least distorting again. Dynamic stylus pressure variations can be viewed as a sort of continually varying offset.

The first of the group to damp his arm noted that where the stereo image had formerly been a bit diffuse, with instruments hard to place in space, now with damping—directionality was positive, and spatial perspective was much more exact. As more of us added damping, the results were duplicated and records sounded better than ever.

Just why this effect occurred isn't fully understood yet, but there's a lot of educated guessing going on. One member feels that because of more intimate groove-wall contact, he is getting more--or less distorted—high-frequency information associated with instrumental attacks—data the brain uses to "position" sound sources. Another postulates that dynamic variations in tracking force are causing subtle variations in channel balance and overall output. "It's as if somebody were continually varying—by a subliminally small amount—the balance and volume-control pots of my preamp," he says, adding that "the effect, now that I think about it, is a little like the difference between a fluttery turntable's reproduction and that of one with minimal flutter- The feeling of uncertainty is gone with damping."

That said, it should have been possible to find at the output of a cartridge amplitude variations that were reduced with the addition of damping. This has been seen on oscilloscopes by several members, as have "dampable" variations in output between channels of a cartridge playing back a mono record.

If groove-wall contact is indeed improved as much as is suspected, it should also be possible to see a reduction in playback distortion with the addition of damping; this is an experiment we haven't gotten to as yet, but we plan to attempt it.

That's a fair summary of the add-on damping state of the art in the local section of the BAS. Perhaps the out of staters, who now make up the bulk of the membership, will do some experimenting and pass in their results to us.—Jim Brinton

ERRATUM: PINK-NOISE GENERATOR

Two capacitors and a switch were misidentified or unidentified in the schematic for last month's pink-white noise generator. The components at issue lie in the "pinking filter" section of the schematic (in the lower right corner of the page): the 3.3-microfarad and 1.0-microfarad capacitors both should be nanofarad devices. Also, the switch at the right-hand side of the drawing is (obviously) for selecting either pink- or white-noise output. In the position shown, the system yields pink noise. Finally, the 33K resistor that comes into play when the switch is in its opposite position may be replaced by a variable resistance (pot, trimmer) for level matching between the pink- and white-noise signals.

Next month, for those who would rather not fool with such a relatively sensitive circuit, Rene Jaeger and Alan Southwick have in store a black box to make pink or white noise from FM-tuner interstation hiss.—Jim Brinton

LETTERS

The Trials of Nate Garfinkle (and Us)

Despite the fact that he's had his work published in magazines like Hi-Fi News and Record Review, Nate Garfinkle still has many of the same doubts and problems as other BAS members. Herewith, a few.

"After reading (The Absolute Sound's) opinion of the ADC-XLM, and discussing it with (Harry Pearson), I had to try one. (Pearson) had told me that my Goldring 800SE was a poor imitation of the ADC induced-magnetic principle; further, he felt that while the ADC was not the flattest or least colored cartridge, it had the greatest 'depth of field' of any, (and) a superior ability to 'focus-in' on individual sounds ... which created a sense of realism missing in most other cartridges.

"(My) XLM was mounted in an SME 3009/II Improved arm with non-detachable shell (which AS says is a must). The reproduction was very accurate ... and very exciting. However, as I played more records, I became aware of a ... harshness ... not present with the Goldring. Many records became unplayable, surface noise was higher in all cases, and the wiry sound of strings became very annoying. Needless to say, I have reverted to my 'poor imitation' of the ADC. This was my first disillusionment with AS."

(We have tried both cartridges here, and would more nearly favor the XLM over the Goldring—to the extent that one can make such a recommendation. In high-fidelity, where technology and taste overlap with the esthetic exercise of listening to music, cut-and-dried recommendations of one transducer, especially, over another are almost impossible to defend. All else equal, your own taste must guide you here. But to continue, Nate also has had, as we all have, his share of software problems.—Jim Brinton)

"Over the years, I've read in The Stereophile, and more recently in AS, about the superiority of all imported pressings over those made domestically . . . With all this conditioning, I have done much listening to those imported labels available to me, and agree that . . . the import usually is quieter, flatter, and more natural. Here I agree with both Stereophile and AS—that is, in all cases but that of Decca versus London.

"The 'Tin-Eared-American' editorial in a recent Stereophile was very convincing on this topic. Then came the AS's record reviews ... and HP really laid into Decca 6BB/121-2 versus London CSP-8 (Beethoven, Symphony No. 9, Solti). He (stated that) the London issue had no bass below 50 Hz, a poor high end, noted differences in the 'musical framework' between the two issues, and compared their surfaces, with the Decca version winning.

"Having a number of other flawed Londons, on which I had gotten precious little satisfaction locally, I decided to send the batch off to the United Kingdom. After being ignored for months, I was told to send them instead to London in the United States. I was properly fried.

"At about the same time, I wrote John Crabbe of Hi-Fi News and Record Review, who, in response, asked me to write a `letter from America' stating my views on Decca versus London and EMI versus Angel records. Crabbe felt that this would be another point of view to add to the arguments (still) taking place in the UK on 'natural recordings.' The letter came out as 'The Great American Record Robbery' in August 1974, and even before I had gotten my issue, I received a letter from the Director of Decca Records, A.C. Haddy. Scared the hell out of me—were they going to sue? No, just curious. I sent him the various pieces that had appeared here and notes on some discs that I felt merited a lookover.

"Haddy soon afterward telegraphed a request that I send back my Londons for Deccas to be exchanged from his British stock. He made the exchanges as promised and included London copies of the same works in addition to the Decca-labelled pressings, asking that I compare them and comment. He noted that so far as he knew, the pressings were identical, and to back it up, he suggested a look at the plate numbers—they matched.

"Well, I have compared them all, and they are identical. I am backing down on my prejudices where Decca/London discs are concerned.

"However, this shows how conditioning and prejudice can affect our judgment; I tried to write Pearson of The Absolute Sound on this, but so far he has declined to answer. So, I am in full agreement with your suggestion that we use magazines like AS as a starting point, and then depend on our own ears for final results."—Nate Garfinkle (California)

READER COMMENT ON FM TUNERS

Thomas C. Mashey writes to (properly) disagree with some of the conclusions that might have been drawn about the Citation 14 tuner as a result of our "quick and dirty" comparison reported in the December Speaker:

"I have heard two Citation 14 tuners—through Sennheisers, equalized AR-3A's, etc., and neither one hummed. Also, the one Pioneer TX-9100 that I heard had a dry, `transistor' sound. Both Citation 14 tuners were, to my ears, far more musical. However, I do agree that the Pioneer TX-9100 has more pulling power; I just don't like its sound.

"These observations came about accidentally since I was comparing loudspeakers to determine whether I would replace my AR-3A's with the Dahlquist DQ-10 or the Heil Tower. I didn't."—Thomas C. Mashey

The hum in the Citation tuner we tested shouldn't be considered a feature of the line; we now are pretty sure that the unit was defective. Still, for the Citation's asking price, one could reasonably expect a higher level of quality control (QC). As for the TX-9100's sound, I suspect the unit auditioned by Tom Mashey might have suffered from the same sort of QC ills as the three units tested here and reported on in the October 1973 Speaker. Since then, many more of these tuners have passed through members' systems, without much improvement in Pioneer's QC batting average. Both H-K and Pioneer build fine tuners, but I suspect that it's necessary to be circumspect when purchasing one.--Jim Brinton

IN THE LITERATURE

Journal of the Acoustical Society of America, June 1972

- Directivity of the Bowed Stringed Instruments and Its Effect on Orchestral Sound in Concert Halls, Jürgen Meyer. From abstract: "At higher frequencies, there are regions of preferred radiation, which change their directions and angle width with frequency. The results suggest that different seating arrangements for the strings would be optimum for different concert halls and different styles of musical compositions."

Audio Amateur, Issue 2, 1974 (Current)

- Contains a brief introduction to one man's preamp-control console, a New York Audio Society report on quadriphonics, a description of one approach to regulated power supply design, the usual interesting columns on transistors and op-amps, and an editorial that mentions the BAS. A letter from Dolby Labs re the modified Advent unit is also of interest.

Audio Scene/Canada, Nov. 1974

- Listening to Phono Cartridges: Notes on three methods for A-B comparison of phono cartridge performance. None is without some critical flaw, however. Interesting mostly as an index of how easily even highly trained individuals can err.
- Testing Speakers in Ordinary Rooms, Part II: Part I appeared in October issue, and the pair of articles offers some excellent advice for audiophiles who are about to try the near impossible, i.e., trying to make objective judgments about speaker performance in the home.

Audio Scene/Canada, Jan. 1975

- Hi-Fi System Test Record: A Technical Review. A review of the new B&K 2011 pink noise test record plus an excellent discussion of the pitfalls of trying to equalize your room using pink noise and a sound level meter.

Electronics, Jan. 23, 1975

- Three-in-One Audio Tester is priced at $78.95. An audio sine/square wave generator (1.5%HD), sweep generator, and analog frequency meter, produced by Production Devices of San Diego.

Photomethods, Oct. 1974

- Audio in Television. Basics of professional recording equipment.

Popular Electronics, Feb. 1975

- Stereo Scene: Tape Head Alignment, by Ralph Hodges. Several how-to-do-it hints on checking the physical condition of your tape heads.
- How Phase-Locked Loops Work. Brief and simple, yet useful, description of the 560 family of PLL's, with a schematic for an SCA demodulator.
- A Vu Meter With No Moving Parts. Comparator circuits used to drive the new ten-bar incandescent display (Readouts, Inc., Del Mar, Calif.) in 3-dB voltage increments, with a tenth bar used to set the 0 VU reference level on peaks.
- Review of the Garrard Zero 100SB Turntable, which states "the low frequency resonance was at 5 Hz (indicating a moderately high mass), and an amplitude of 10 dB."

Radio Electronics, Feb. 1975

- Noiseless Discs at Last, by Len Feldman. Discussion of the dbx phonograph record noise reduction system.
- Taming the Bass Reflex, by David Weems. How to tune the port of homemade systems.

RECORDING HIGHS AND LOWS

Well so far the largest contribution to this column has been Dave Ranada's letter in the January issue. I found a number of his points informative, and his criticism of the brevity and technical inaccuracy of such a column is sound. However, knowledgeably written record reviews abound (everywhere but here). The idea behind this column was to get likes and dislikes exchanged among members. I hate to believe Bach really wrote for sour horns, but yielding to superior knowledge and judgment on that point, I still believe he would have liked it better on modern well-tempered instruments. Giving up even that retrenchment doesn't affect my own preference for clear, sweet sound.

Whether other members do or don't utilize these brief listings is the key question. Until the dribble of contributions drys up or until July (whichever comes first), we'll continue this column. Your lack of response will serve as well as additional criticism.

Sheffield III. To ease everyone's mind, my poor surfaced disc was cheerfully replaced by the factory, and Dan Shanefield owns a perfectly sound copy with the same pressing numbers as my bad one. Evidently a unique occurrence. (P.S. I found Volume II more fun.)—Joel Cohen

• Stravinsky/Petrouchka/London Symphony/Mackerras/Vanguard VSQ30021/SQ encoded but sounds great in stereo too. Unusually horny horns, room-shaking kettledrums, good violins, etc.—Dan Shanefield

[Anyone out there had experiences with SQ discs holding up on non-quad gear or not ?—Ed.]

• Richard Strauss/Don Juan (on side 2)/Berlin Philharmonic/Bohm/DGG 138866 SLPM/ Good horns and violins.—Dan Shanefield

• Another vote for Rachmaninoff/Piano Concerto No. 2/Richter, piano/Warsaw Symphony/ Wislocki/ A friend has a copy with no surface defects.—Dan Shanefield

• Holst/Vaughn Williams/Grainger/Music for Symphonic Band/Eastman Wind Ensemble/ Fennell/Mercury Golden Imports SRI 75011/ One of the best sounding rechanneled stereo records you will ever hear.—Warren Schroeger (Hear, hear! —Jim Brinton, Bob Graham)

- Respighi/The Birds/Brazilian Impressions/London Symphony/Dorati/Mercury Golden Imports SRI 75023/ A refreshing alternative to Debussy/Ravel and friends. Superb pressing. Intelligent liner notes by James Lyons.—Warren Schroeger

- DuMage/Livre d'Orgue/D'Andrieu/Premier Livre d'Orgue/Frank Taylor, organist/ Available only by mail order from Elysee Editions, 88 Lowell Rd., Wellesley 02181/ Excellent sonic perspective using a two-mike technique. Excellent sonic definition and low distortion.—Rene Jaeger.

- Mozart/Six Sonatas for Flute and Harpsichord/Rampal & Veyron-Lacroix/Odyssey Y32970/ Delightful, especially first movement of Sonata No. 5. Well performed (better than Nonesuch). Recording good except harpsichord a trifle thin; decent pressing.—Jim Richardson

- Hovaness/St. Vartan/National Philharmonic/Hovaness/Unicorn RHS317/ Interesting work using some near-Eastern elements. Well recorded; good pressing except for shushing in transitional grooves.—Jim Richardson

- California BAS member Nate Garfinkle (see letter in this issue), after a recent trip to Montreal, notes that London of Canada Cassettes (of which he bought a good number) are superb—far better than London/Ampex. Made in Canada on British equipment, installed by British engineers, according to A. C. Haddy, Director of Decca Records. Haddy adds that Decca has no control over Ampex issues stateside.

JANUARY BAS MEETING

BUSINESS AND OPEN DISCUSSION

About 100 people met at BU on January 19. Peter Mitchell began to take orders for the Thermo Electron electret microphone capsules. Since this is just the capsule, purchasers must make their own mounting and cabling arrangements. Al Southwick described a cable connector, the Switchcraft S3FM, that can easily be modified to make a shell for mounting the mike capsule with a standard Amphenol connector on one end. Fitting batteries inside, however, may be a tight squeeze. These connectors are available locally through A. W. Mayer in Newton for about $4. Larry Kaufman and Mark Forman both said they could arrange a group purchase of mike cables. Mark showed a sample cable that was particularly interesting because of its light weight.

A tour of the DGG recording facilities at Symphony Hall was also announced by Peter. The tour was arranged for Monday evening, January 20, by the Boston Section of the AES. Tom Mowry, producer for many of the. Boston Symphony Orchestra recordings on DGG, was the host.

For those who like to listen to music while testing their system (and are not owners of the latest Shure cartridge), Al Foster has offered to obtain the Shure trackability test record "An Audio Obstacle Course—Era III" at $4.25. This record contains a number of musical tracks recorded at successively higher levels and an explanation of how to listen for mistracking. Al will also arrange for purchases of the Shure TTR-103 test record, which consists of trackability tests in the low-, mid-, and high-frequency bands that can be interpreted by ear. Recorded velocities range up to 40 cm/sec in mid-band and 30 cm/sec in the high and low bands. The purchase price is $9.50.

A new program series, produced by Dave Letterman and Peter Mitchell, will debut on WBUR in March on Sunday evenings from 7 to 9 p.m. Dave and Peter will be playing tapes of performances of such local musical groups as the Boston Philharmonia. Some of these broadcasts should rival the BSO broadcasts in sound quality.

MEETING FEATURE: A-B EQUIPMENT COMPARISONS

The program consisted of a review of the techniques used and precautions to be taken in A-B-ing equipment, and a demonstration of the pitfalls and pratfalls one can encounter in dealing with real live equipment. The presentation by BAS members Jim Brinton, Al Foster, Bob Graham, and Peter Mitchell compared the SAE Mark IB (transistor) and Marantz 7C (tube) pre-amps, ADC-XLM and Micro-Acoustics QDC-1E cartridges, records versus tapes, and damped versus undamped SME tonearms. These components were played through Rectilinear Mini-3 speakers with added Micro-Acoustics tweeter arrays, driven by a Phase Linear 700.

Jim started off by presenting a set of guidelines for A-B equipment comparison that seemed simple enough: Keep the variables to a minimum—change only one thing at a time. Make sure the sound levels are the same when switching between A and B, by checking output at 1 to 3 kHz or with pink noise using a VTVM, sound pressure level meter, or your ears. Use blind tests when possible; it's easy to make that new piece of equipment you just dropped a wad on come out a winner if you know whether it's A or B. Comparing two cartridges by playing the same record simultaneously may put the trailing cartridge at a disadvantage, since insufficient time has been allowed for the vinyl to recover from the first play (this usually takes hours). It may be that this comparison is better done using a separate pressing for each cartridge (from the same stamper, if possible) and interchanging them often. Other precautionary measures include eliminating acoustic feedback, optimizing the cartridge load impedance, and ensuring that the A-B switch does not introduce excess capacitance or inductance, which might alter the frequency response, a particular problem for cartridges.

As it turned out, these precautions are easier said than followed, and many of them were violated during the actual demonstrations, mostly for the sake of expediency. As Jim cautioned, "Do what we say, not what we do."

Damped Versus Undamped Arms

The first demonstration was a comparison of the tracking ability of damped and undamped tonearms. Bob Graham supplied an AR turntable equipped with an SME 3009-U and a 3009-II Improved, the newer arm having been damped using a modification of the technique described in last month's Speaker. Bob demonstrated the performance differences between the two arms by playing a badly warped record with each arm using an ADC-XLM cartridge. In the undamped arm, the warp-stimulated arm/cartridge resonance lifted the stylus completely out of the groove, making the combination inoperable. In the damped arm, the same cartridge played the same record with only a slightly audible warp-induced modulation of the music. This was, admittedly, an extreme demonstration (the disc had been warped over a hot stove), but it proved the point that damping can dramatically reduce the Q of arm/cartridge resonances.

Bob then described the latest version of his simple damping system. It consists of a stiff wire attached to the tonearm with a plastic cable clamp about two inches ahead of the pivot on the cartridge side of the tonearm. The wire runs downward into a trough of STP sitting on the mounting board. At the end of the wire, completely submerged in STP, are two 3/8-inch-diameter discs (or paddles) soldered horizontally and vertically perpendicular to the wire. Thus greater damping force is offered to the vertical motions of the arm than to the horizontal. The mounting was shifted to this location from the rear of the counterweight, as described last month, because the counterweight decoupling of the new SME arm also tended to decouple the damping and reduce its effectiveness. It was the consensus among members who have had experience with damped tonearms (see Al Foster's comments in the January Speaker) that benefits realized from further reductions in tonearm mass and pivot friction beyond that achieved in today's good quality arms were miniscule compared with the improvements gained by damping. Bob admitted that he probably would not have bought the new SME had he tried damping on his old SME-XLM combination.

QDC-1E Versus ADC-XLM

The XLM was mounted in the damped new SME arm and the QDC-1E in the old SME. This was felt to be justified in that because of its high compliance the XLM was much more affected by arm/cartridge resonances than was the QDC-1 and would therefore be at a disadvantage in an undamped arm. Both cartridges played the same record through the Marantz 7C preamp. Since the cartridge sensitivities were different, it was necessary to ride gain when switching between them to achieve a semblance of equal sound levels. With these rather wide deviations from ideal A-B conditions, and given the poor acoustic quality of the room, it was possible to conclude only that the XLM had a brighter high end response. No judgment could be made as to which cartridge was the more accurate reproducer, though the QDC was producing more hum (perhaps due to a grounding problem).

Marantz 7C Versus SAE Mark IB

This comparison was to illustrate what audible differences could be detected between the archetypal tube preamp and a very good transistor design. The signal source was the ADC-XLM in the damped SME arm. After extensive switching, there was little agreement as to what differences could be heard or what they were. A few even heard differences when they were unaware the Marantz 7C was being switched to itself in a blind A-A test.

Tape Versus Discs

It is often said that discs can be as high a quality source of sound as tape but usually are not. This is a result of compromises that are made when cutting and pressing the disc, and imperfections in the arm/cartridge playback system. About the former, the audiophile can do little but buy the best recordings available. Given these, are cartridges accurate enough that the disc may be compared favorably with the original tape?

In an attempt to answer this question, a number of tapes were A-B compared with their disc counterparts. Selections from the Dolbyized cassette and disc versions of the same performance were chosen from Musical Heritage Society and Advent-Nonesuch releases. Other pieces were taken from tapes made of Victor Compos' master-tape broadcasts, "Adventures in Sound," on WGBH, and compared with their commercial releases on disc. The Advent 201 and Revox A77 were the tape machines, compared to either the QDC-1E or ADC-XLM in the damped SME arm on the discs.

Under the poor listening conditions of the meeting room and with the recorded material chosen, the sonic advantages often claimed for tape over disc were not always evident. Some tapes appeared to have a fuller, less limited bass spectrum than their disc counterparts. Yet, it was impossible to conclude which medium recreated most naturally the sound of musical instruments.

While the A-B comparisons during the evening were, for the most part, inconclusive, the experience highlighted some additional points which should be included in the list of things to be considered when setting up A-B comparisons. Optimum results require a good listening environment (neither too bright nor too dead), time to become familiar with the equipment, and a system (especially speakers) capable of delineating musical nuances to illuminate the very subtle differences in components and sources.---John Schlafer

BAS MEMBERSHIP LIST SUPPLEMENT

This list is for the use of BAS members only

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Belot, Henry	298-0794
Bires, Glen	(815) 338-4218
Browne, Barrie	(416) 521-9835
Cabrera, Victor	(off.) (716) 883-4436
Carpenter, E. Scott	1-562-5819
Cohen, Joel	(off.) 890-1727
Diller, Wendell	Minnesota
Disend, Joel	1-429-5339
Edgar, Robert	(808) 261-6209
Fasi, Joseph	(313) 836-2972
Gage, Richard	1-658-3527
Goldberg, Harry	(314) 726-6540
Goodhart, Bertram	(305) 235-6596
Goodman, Samuel	(416) 783-9426
Gorra, Chip	782-5379
Gupta, Chris	(416) 362-4167
Gut, Gary	New York
Hammermeister, Rudolf	(412) 683-4234
Henderson, J. T.	(415) 285-9480
Hull, Arthur	828-9066
Hurley, Charles	(603) 883-5353

Kaiser, John J.	Illinois
Kaufman, George	272-5814
Kelly, Peter	(416) 655-4932
Leventhal, Les	(204) 837-6670
Long, William	334-5526
Lush, Morley	1-369-4426
McCarville, Terry	(off.) (716) 883-4447
Mindel, Laurence	(608) 271-4734
Nicholakakos, Peter	(212) 256-4998
Paradis, Jack	1-957-4433
Parker, John	Texas
Preis, Douglas	492-1228
Pressman, David	729-5173
Richardson, Charles	(301) 757-3733
Saklad, Mark	
Sanders, Malcolm	(713) 833-3644
Schroeger, Warren	(off.) 523-6611
Shedd, Thomas	(312) 251-3592
Swain, Michael	731-9648
Trotsky, Ron	924-0058
Vreeland, Whitney	Oregon
Wargo, Lorand	1-396-1374
Zahora, Robert	1-369-2832

A Publication of the BAS

A Quasi-Complementary Discussion of Microphones

(Including the Thermo-Electron 814)

Peter W. Mitchell

PART ONE: SOME NOTES ON MICROPHONE SPECIFICATIONS

Frequency Response

It used to be thought that the "frequency-response" curve of a loudspeaker (as measured in an anechoic chamber using a calibrated mike located on axis 1 meter away) would provide a useful guide to the speaker's sound. Judging from their test reports, some hi-fi magazines (especially in Britain) still operate on this assumption. But during the past decade it has been widely accepted that the on-axis frequency response does not describe the sound we hear (unless you sit only a foot or two from the speaker—try it and see). At a normal 10-foot listening distance, the sound we hear consists of the direct on-axis sound plus a great deal of off-axis sound reflected off the walls. So a better index of the sound is a graph of the speaker's total acoustic power output (at all angles) versus frequency. In an ideal loudspeaker, both the on-axis and total-power curves will be smooth.

This is a familiar story, but it has a point. Microphones are essentially loudspeakers acting in reverse. (In fact, if you ever need a mike in a hurry, just plug a loudspeaker into your mike input and talk into the speaker; you'll find it makes a decent mike!) And the important questions about meaningful loudspeaker specifications apply also to microphones. We are faced with a serious deficiency when we try to evaluate mike specifications: usually the only response curve that is published for mikes is the on-axis frequency response. This is meaningful if you are going to record with the mike only a foot or two from the sound source and pointed directly at it. But usually you don't want to record this way, because the recording will be very "dead," devoid of the "air" and spaciousness that hall reverberation provides.

For general concert recording with a basic stereo pair of mikes, the optimum location for the mikes is near the boundary that separates the "direct field" from the "reverberant field" of the hall. At this location, the mikes will pick up a mixture of direct and reverberant sound, with the reverberant component arriving at the mike from all directions. In a typical recording locale, this boundary is 10 to 20 feet from the musicians. Since the sound of a recording will depend on the response of the mike to instruments or voices located at various angles relative to the mike axis, as well as on its response to reverb sound from all directions, the most useful specification would be a response curve describing the mike's total response to sound arriving from all directions. But this omni-response curve, or "random-incidence" response, is not provided by manufacturers nor by the few test reports that are published. As with loudspeakers, if a mike's on-axis frequency-response curve and its omni-response curve both were smooth, it probably would be an excellent mike for general recording. Experience indicates that some mikes have a good on-axis response but poor omni-response; they sound good in close-up recording but are lousy for general orchestra pickup. A cardioid mike is especially likely to have a rough off-axis response despite an attractive on-axis curve.

The result is that you cannot use published specifications as a basis for choosing microphones. The widely published on-axis frequency-response curve is of limited use and the desirable random-incidence curve is not available. The failure of manufacturers and test labs to provide adequate specs means that microphone choice is, by default, at least as subjective a process as loudspeaker choice. The only sure way to choose a mike is to use it in actual recording sessions and compare the results against recordings made with other mikes. Now, this may be practical for big recording studios, to whom the major mike manufacturers willingly lend mikes free for evaluation. But for individual consumers and small studios, this opportunity does not exist. So microphone choice is a frustrating and defeating process. It is not surprising that so many people buy tape recorders and so few people buy microphones, when they are faced with a substantial price tag but are given no basis for rational choice.

The confusion about published frequency response curves is fully as important with "omnidirectional" mikes as it is with cardioids, because at high frequencies "omni" mikes are not omnidirectional. A mike can be omnidirectional only at frequencies for which the diameter of the mike is much smaller than the wavelength of the sound (just as a loudspeaker can have wide dispersion only at frequencies for which the diaphragm size is well under a wavelength). If the mike is larger than about 1/3 of the wavelength, its response will vary with direction. Knowing the speed of sound, you can show that the maximum omnidirectional frequency of a mike is approximately $f = 4500$ Hz/d, where d is the diameter of the business end of the mike, in inches.

Recognizing this, manufacturers make omnidirectional mikes with two kinds of high-frequency response: (1) "free-field" mikes intended for on-axis use in anechoic, outdoor, or close-up situations (the on-axis response is flat and the off-axis response falls off, as does the omni-response), and (2) "random-incidence" mikes whose response is effectively flat in a reverberant sound field (the response rises on axis, is approximately flat at angles of 70 to 90 degrees off axis, and droops at angles behind the mike). Obviously, when choosing and using an omni mike, it is important to know which kind of high-end response it has.

<u>Sensitivity</u>

Mike sensitivity specs are confusing because there are at least five ways of expressing the sensitivity of a mike. A single mike could have a rated output of -47 dB, -51 dB, -67 dB, -71 dB, or -144 dB. The last of these, the EIA rating, may be ignored. But the confusing variety of the other four is due (1) to the use of either the mike's output voltage or its output power as a ratings quantity, and (2) to the use of two different sound levels as standards at which to rate the mike output. Obviously we need to be able to convert these four into one standard rating in order to compare mikes or relate a mike to the preamp's specs. Fortunately it is not difficult.

Consider the sound levels first. Some mike ratings are referred to a sound pressure of 1 microbar. (Since the nominal threshold of hearing is 0.0002 µbar, 1 µbar amounts to a sound pressure level of 74 dB SPL.) Instead of using the simple 1-µbar reference level, some manufacturers prefer to rate their mikes at a 10-µbar reference level (i.e., at 94 dB SPL). One advantage of this is that 94 dB, give or take a few dB, is in fact the highest sustained SPL that a mike ordinarily will encounter in a wide range of common situations, such as a classical orchestra at a distance of 20 feet, a piano at 5 feet, or a speaking voice at 6 inches. Of course there are exceptions that are either higher (a Mahler climax, a rock band at full tilt) or lower (a lute or clavichord), but the rated output at 10 µbars is a good typical value of the maximum sustained output that the mike will deliver in a recording, and so can be compared directly with your recorder's sensitivity for 0-VU recording level.

So the first step in making sense of a mike rating is to look at the reference sound level. If it is 1 µbar, add 20 dB to the rating to convert it to the 10-µbar reference. If no reference is stated, the value of the rating itself may provide a clue; a mike sensitivity of between -40 and -60 dB probably was measured at the 10-µbar sound level, and a rating of between -60 and -80 dB probably was measured at 1 µbar. Thus a mike rating of -73 dB at 1 µbar is the same as -53 dB at 10 µbars.

The next step in straightening out mike ratings is to convert from power to voltage or vice versa. Mike output commonly is expressed either in dBm (dB relative to 1 milliwatt of power), dBV (dB relative to 1 volt, measured "open-circuit" with no preamp connected to the mike), or directly in millivolts (mV, again open-circuit). The relation between power and voltage depends on the mike's impedance. The nomograph in Fig. 1 provides an easy way to do the conversion;

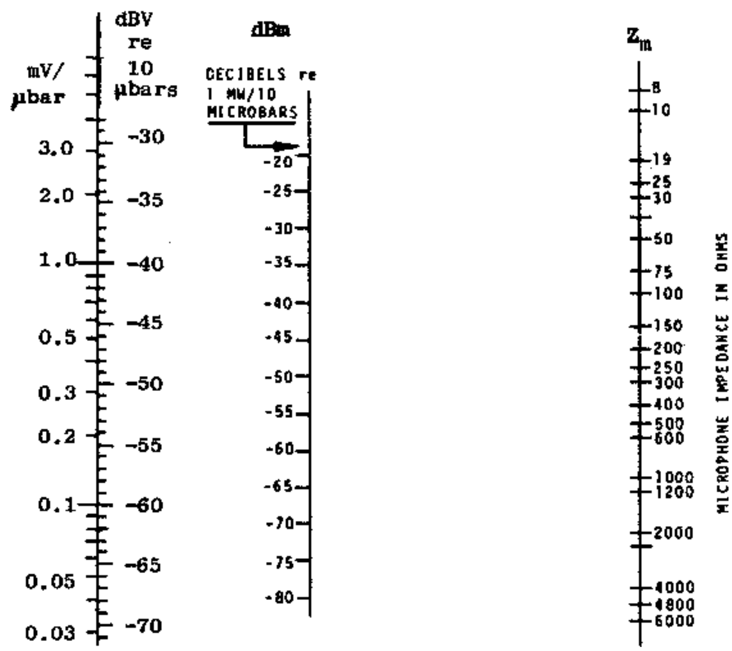

Fig. 1. Microphone sensitivity nomograph

just draw a straight line from the impedance to one output rating and it will intersect the other rating value. The graph also shows the relation between dBV (at 94 dB SPL) and millivolts (at 74 dB SPL). For example, a typical dynamic mike by AKG, Beyer, or Senheiser has an open-circuit output voltage of 0.2 mV/µbar. This corresponds to -54 dBV at 10 µbars; if the mike impedance is 500 ohms, this translates to -57 dBm. For comparison, the AKG C451 condenser mike is rated at -38 dBm for 10 µbars; since its impedance is 200 ohms, the nomograph shows its voltage rating to be -39 dBV, which is 15 dB higher in output than the dynamic mikes.

The value of a sensitivity rating, of course, is to enable you to predict a mike's compatibility with a preamp. The essential preamp specs are noise level, overload level, gain, and input impedance. Consider, for example, the Advent MPR-1 mike preamp, one of whose many virtues is that meaningful specs for it are available. Its noise level is 0.4 µV (-128 dBV). Suppose you use it with a dynamic mike rated at -54 dBV for 10 µbars. The preamp noise is then 74 dB below the mike's rated level. Since the mike spec was rated at a sound level of 94 dB SPL, the preamp noise is equivalent to a sound level of 94 – 74 = 20 dB SPL. The background noise level occurring at a typical concert is around 25 to 30 dB SPL, so the preamp noise will be inaudible in the recording. (But if the preamp's noise-equivalent SPL were over 30 dB, as is true of the preamps in many recorders, its hiss would mask some of the subtle hall reverb that makes a concert recording sound lifelike.)

The preamp overloads at an output of 3 volts rms (+10 dBV rms, or +13 dBV peak, since the peak of a sine wave is 3 dB above its rms value). The preamp's gain is 40 dB, so the input overload level is +13 - 40 = -27 dBV peak. With a mike rated at -54 dBV, the overload is then 27 dB above the rated mike level; since the mike rating is for a sound pressure of 10 µbars (94 dB SPL), preamp overload would occur on peaks of 94 + 27 = 121 dB SPL. This is an ample margin unless you record very close to a very loud source.

One other factor in mike/preamp compatibility should be mentioned; the preamp's input impedance, which "loads" the mike, reducing the voltage delivered to the preamp circuit in proportion to the ratio $Z_p/(Z_p + Z_m)$ where Z_p and Z_m are the preamp and mike impedances. Figure 2 shows how much the signal is reduced in the case of Z_p = 1000 ohms (the impedance of the Advent preamp). The loading effect is typically about 3 dB, so the mike rated at -54 dBV actually delivers -57 dBV into the preamp; corrected values of preamp noise and overload level would then be 23 dB and 124 dB SPL, respectively.

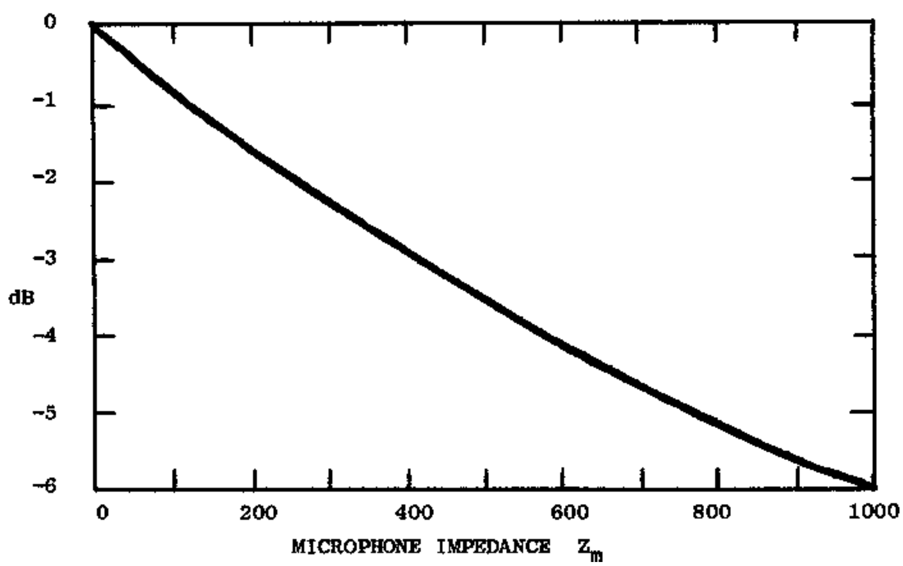

Fig. 2. Preamp loading (Zp = 1000 ohms)

PART TWO: AN EXTRAORDINARY MICROPHONE BARGAIN

In the consumer microphone business the word "electret" has sales magic. Ever since the electret mike was introduced in the mid-1960's, it has been widely claimed (and believed) that the electret principle provides professional condenser microphone performance at dynamic mike prices—better transient response, smoother frequency response with less coloration, elimination of diaphragm resonances and breakup, better clarity, higher highs and lower lows, all at a moderate cost. Unfortunately electrets generally have not lived up to their early promise. The electret microphones that have been widely available with under-$100 price tags have exhibited a multitude of deficiencies: annoying sample-to-sample variability, poorly damped resonant

a

peaks in the 10- to 13-kHz region causing sibilance in voices and edginess in string sound, a high noise level and very poor overload level in the built-in FET impedance converter, and insufficient reliability (subject to loss of output in the presence of humidity, heat, or mechanical impact).

A couple of years ago we heard about an electret microphone module, the Thermo Electron 5333, which was available for $10 and was said to be unusually good. We tried one and found it to have an impressively smooth midrange and extended bass response, but with a severely sibilant 12-kHz resonance and a noisy FET that overloaded easily. This experience was so disappointing that we did not get around to testing Thermo Electron's larger module, the 814, until recently.

When we did, the result was astonishing. The 814 is the smoothest sounding mike I have used. The midrange is absolutely neutral, the bass is full and rich, the top end is airy, detailed, and free of sibilance, the built-in FET is quiet, and the overload level is higher than any sound I have exposed the mike to yet. Furthermore, it appears to be both consistent and reliable. The three samples I have tested had identical sensitivity and sound quality. I have torture-tested one sample by dropping it, letting it get rained on, and carrying it around in a dirty coat pocket without producing any evident change in its performance. The TE 814 appears to be the first electret mike that really does combine the performance of a good condenser mike with the practicality of a dynamic mike.

A Little Background

A dynamic mike actually is a small loudspeaker working in reverse, and contains a diaphragm attached to a voice coil suspended in the field of a magnet assembly. (In fact the drivers in some popular European dynamic headphones actually are mike elements.) These mechanical parts can be made quite rugged at low cost; there are no internal electronics, so no electrical power needs to be supplied to the mike; and the output signal depends only on the physical characteristics of the diaphragm, the winding of the coil, and the magnetic field, all of which can be mass-produced very uniformly. So dynamic mikes can be made good, consistent, and reliable, at low cost. Just as with dynamic loudspeakers, the performance improves rapidly with increasing cost until you get up to about $50 or $100, beyond which the design engineers apparently start running into the physical limitations of the medium so that additional increments of cost produce smaller improvements in frequency range or in the elimination of colorations due to internal resonances in the diaphragm material.

The "condenser" microphone (which really should be called the capacitor mike) is analogous to the electrostatic loudspeaker. It strikes at the heart of the dynamic mike's limitations by employing a very low-mass diaphragm with an electrostatic force field acting uniformly over its entire surface, completely eliminating diaphragm resonances and producing an extremely smooth, uncolored response. But this performance can be obtained only at high cost. A high-voltage supply (typically 40 to 200 volts) is required to produce the electrostatic field. The impedance of the capacitor element is a hundred million ohms or more, so an electronic impedance-converter circuit must be built into the capsule to transfer the signal out at a usable low impedance. This circuit also requires supply power, and in some cases has been a source of unreliability. Together, the power supply and the impedance-converter circuit substantially increase the cost and reduce the convenience of condenser mikes.

The electret mike was made possible by two technological developments: (1) plastic materials, which, when heated, will take and hold a permanent high-voltage charge, thus maintaining a permanent electrostatic force field (these materials are called "electrets" because they are the electron equivalent of magnets); and (2) field-effect transistors suitable for use as impedance converters needing only a battery to power them. Marrying these gave birth to the electret mike capsule, providing capacitor-mike smoothness and range at low production cost and needing only a battery instead of a costly external power supply.

But early electrets did not perform up to expectations, and the reasons are now well understood. Most electrets have been made of Teflon or polycarbonate, and under high humidity the voltage charge would simply leak off them, leaving the mike useless. If the mike was subjected to mechanical shock, the diaphragm could come into contact with the backplate and stick there, permanently shorting out the mike. The electrode-contact method usually used to apply the high-voltage charge in manufacturing usually resulted in large variations in the distribution of the charge over the area of the electret, causing erratic behavior. The low-frequency range and the impedance converter's noise level were found to depend critically on the value of the bias resistor in the capsule. The overload level depends on the FET circuit. And the high-frequency resonance of the capsule depends on the tension and mass of the diaphragm.

With these parameters understood, Thermo Electron claims to have developed mike capsules in which all of these problems are under control. Their electret material is chlorotrifluoroethylene polymer (chosen for its water resistance), gold-coated. The high-voltage charging (390 volts) is done via corona discharge, which is said to produce an unusually high charge density that is uniform to within 2% over the diaphragm area. The diaphragm is supported by a grid that prevents contact with the backplate, and the mike elements are made of very low-density (i.e., low-inertia) materials; consequently they are quite immune to mechanical shock. The bias resistor is in excess of a billion ohms, ensuring an extended bass range together with a noise level in the vicinity of 25 dB SPL. Two versions of the model 814 mike are available, a standard version in which the FET impedance converter maintains low distortion up to about 115 dB SPL (operating from battery voltages between 1 and 20 volts), and a "C" version in which the impedance converter remains clean up to about 140 dB SPL (requiring a battery of 8 to 30 volts). In both cases the high-frequency resonance of the electret is beyond the audible range. As for environmental stability, TE has reported aging studies in which samples of the mike were exposed to 120 °F and 95% relative humidity continuously for extended periods. Typically the electret loses 1 dB during the first week of this test and then remains stable for at least 20 to 30 weeks thereafter (which would correspond to several decades at normal temperature and humidity).

Comments on Its Use

The spec sheet for the TE 814 capsule (in last month's Speaker) states that the mike is an omnidirectional, not a cardioid. Cardioid mikes outsell omnis by about 10-to-1, reflecting a popular prejudice against omnis. Certainly in public-address work and in taping interviews, a cardioid is valuable to reject noise that would mask the words. But in concert recording, I think cardioid mikes are very widely misused. A common mistake in professional recording is to use close-up cardioid mikes to capture lots of detail, making the recording so "dead" that they then have to mix in extra "hall mikes" or add artificial reverb to restore the hall sound that was rejected by the cardioids. This approach often produces poor stereo imagery as well (see the March WBUR Folio for a commentary on mike technique). I concur with what Jim Long of E-V said in his excellent "Microphone Primer" in Audio (Dec. 1972 - Jan. 1973):

> "If you can afford only one type of microphone for recording, buy an omnidirectional. The need, in recording, for a cardioid's reduction of room reverberation and unwanted background noise is often overrated by amateurs and professionals alike. It is frequently a fight to get enough of the room 'sound' that is so much a part of an effective listening experience, live or recorded. Two properly placed omnidirectional microphones can provide impressive musical results with a minimum of risk and frustration!"

The model 814 is a "random-incidence" mike. The capsule diameter is 0.656 inch, so the on-axis frequency response rises above 7000 Hz. The reverberant-field response is close to flat, as is the direct-field response at right angles to the mike.

The exceptional smoothness and neutrality of the 814's midrange response is largely responsible for its very pleasing string sound in orchestral recordings and its natural reproduction of the pipe organ. It also sounds fine on the singing voice. But it may be too neutral for some uses. For example, if your loudspeakers have a particularly soft midrange response, the 814's sound may seem to lack detail. A professional recordist of my acquaintance doesn't like the 814 because in comparison with other mikes it seems to have a bit of an "absence" peak in the midrange where many mikes have a slight presence peak. I placed an 814 and an Advent MDC-1 mike side by side and recorded a talk simultaneously on each. The 814 sounded natural and lifelike while the MDC-1 sounded like a recording; yet the MDC-1 was better because its slightly emphasized "presence" range made the speaker's words clearer and easier to understand. (I find that the MDC-1 is superb as an accent mike for vocal or instrumental solos, with two 814's providing the general stereo pickup of orchestra or chorus. For a mixer I use two Advent MPR-1 preamps plus the IC op-amp mixer described in the September 1974 BAS Speaker.)

The impedance of the 814 is not specified. The resistor across which the output signal is developed is about 5600 ohms, but the mike's output impedance depends on the FET. I measured the impedance is 500 ohms. To make the measurement, the mike was connected to a high-impedance (50K) preamp whose output fed a voltmeter, and the mike was driven by a constant 300-Hz tone at 94 dB SPL. A variable resistance was connected across the mike's output terminals and was varied until the output signal dropped 6 dB from its open-circuit value, at which point the variable resistor is equal to the mike impedance. I have not measured the impedance of the 814C (the high-level version); it may be different since another FET is employed.

The rated sensitivity of the 814 is –68 dBV at 1 µbar, or –48 dBV at 10 µbars, about 6 dB higher than a typical dynamic. Therefore the Advent MPR-1 mike preamp will overload with the 814 on peaks of about 118 dB SPL (see Part I). This is similar to the level at which the 814's own FET overloads, and is ample for the general concert recording that I do (including large orchestras and choruses). Of course if you plan to do very close-up miking, you would need to use the 814C, which has about 2 dB more sensitivity and can be used up to about 140 dB SPL. But to use this dynamic range you would have to either (1) use a 20-dB attenuator to knock the level down so that a normal preamp like the Advent MPR-1 can handle it, or (2) use a preamp with an extremely high input capacity, or (3) bypass the mike preamp altogether (at 140 dB the 814C puts out a full volt, more than enough to drive a recorder's line inputs directly). When considering mike and preamp overload levels, it is wise to keep in mind the inverse-square law of distances. The sound intensity from a source goes up very fast as you come close to it. For example, Fig. 3 indicates the variation of level versus distance for a source producing 100 dB at

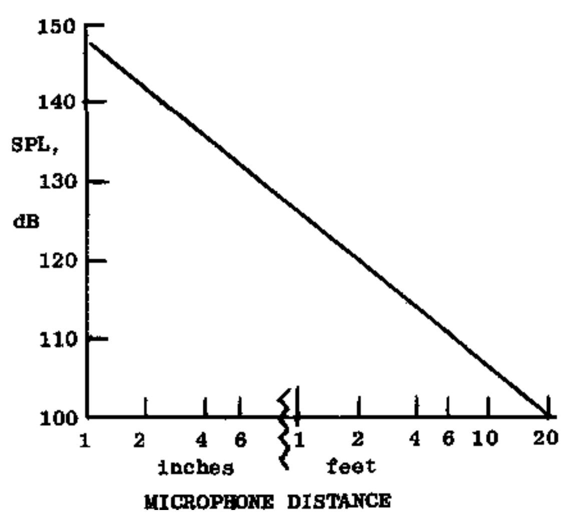

Fig. 3. The inverse square law

20 feet such as a trumpet or drum. The level reaches 120 dB at 2 feet and 132 dB at 6 inches. So keeping your mikes back at a respectful distance from the musicians not only gives an attractive, well-balanced sound with some hall reverb, it also helps you avoid mike and preamp overload.

All you buy in the 814 is the capsule (the electret plus the FET impedance converter). It is up to you how you mount it, wire it, and supply the battery voltage. Al Southwick and Mark Formar have suggested that it can be directly press-fit into a Switchcraft/Cannon cable adapter. Personally I would prefer to shock-mount the capsule in its case (even though it is quite rugged); and it isn't evident how you would fit a battery and an on-off switch into the cable adapter with the mike, though with some ingenuity this obstacle might be overcome. If you want to mount the capsule in a case for hand-held use, I would suggest using ordinary one-inch aluminum tubing, available at a hardware store; cut the tubing to whatever length you like. The capsule could be glued into a sleeve of polyurethane foam for shock isolation, which would fit into the tube.

Battery voltage is up to you. The 814 will work with a 1.5-volt AA penlight battery, but I prefer to use a 9-volt source because I got radio interference when I experimented at 1.5 volts. The type 126 mercury battery is convenient, as it supplies a rock-stable 8.4 volts, which will last practically forever in this application; the battery is cylindrical (3/4 inch diameter by 2 inches long), fits a standard no. 139 battery holder, will fit into the one-inch tube, and is widely available (even at Radio Shack).

It isn't necessary to mount the mike capsule in a case, of course. I use it bare, with the battery taped to the signal cable a couple of inches from the mike. At some concerts the mike cable carrying the capsule is simply taped to a string tied across between balconies. For recording where I can't hang mikes, I tape the cable along a lightweight extension boom (3/4-inch aluminum tubing 8 feet long) which I have fitted to my cheap mike stands; the end of the cable carrying the mike protrudes a couple of inches beyond the end of the boom.

You also have the choice of whether to wire the mike for unbalanced or balanced-line operation. For serious recording the balanced-line configuration is essential in order to permit adequate cable lengths (50 feet or more) for on-location recording without picking up hum and interference. The wiring arrangement for unbalanced operation is straightforward since the terminals are identified on the spec sheet (see the January Speaker). To wire for balanced-line operation using a cable with two signal leads plus shield, connect the "signal hot" (light-colored) lead to the mike's "signal" terminal and the "signal ground" (dark) cable lead to the mike's ground terminal. If using the mike bare, leave the cable shield unconnected at this end; if the capsule is mounted in a case, connect the cable shield to the case. (In this situation the capsule must be electrically insulated from the case or tube, since the entire capsule exterior is already connected to the "signal ground" which must be kept isolated from the cable shield ground.) At the other end of the mike cable, connect your plug in the normal way for balanced-line operation. If you use three-circuit phone plugs to mate with the Advent preamp, the "signal hot" lead goes to the plug tip, the "signal ground" lead to the ring, and the cable shield to the plug sleeve. In the bare-mike system, the cable shield will continue to be effective so long as the end closest to the preamp is thus connected to the preamp ground.

Incidentally, when soldering wires to the mike capsule, use the same precautions you would when soldering a transistor. I overheated one of my capsules in soldering, causing it to lose about 2 dB in sensitivity.

If you use the 814 with a professional-type preamp having balanced-line input transformers (such as the Advent MPR-1), the transformer has an interesting effect. Its impedance is at some specified value in the audio range (1000 ohms in the case of the MPR-1) but at subsonic frequencies the input impedance rolls off, getting down almost to zero at dc. As a result, the preamp loading on the mike increases at subsonic frequencies (Part I), rolling off the effective mike response. This very convenient filter prevents large subsonic signals (due to common building rumble and floor vibration) from getting into the preamp and muddying up the recording—a common problem with some condenser-mike/preamp combinations, necessitating elaborate and often ineffectual shock mounting of the mikes. However, Rene Jeager has pointed out that the very low dc resistance of the input transformer effectively short-circuits the output resistor in the capsule and so may alter the operating point of the FET. I have used the 814 extensively with the Advent MPR-1 preamp and have noted no ill effects from this situation. But it may be worth a closer look, especially with the 814C. If significant, the problem can be cured by installing a capacitor (a 10-µF, 6-WV electrolytic would be suitable) between the capsule's signal terminal and the cable's "signal hot" lead.

The Thermo Electron 814 high-performance electret microphone module is not widely available. The only direct retail source I know of is Electronic Enterprises in California, who sell the module for about $42, and a complete kit including aluminum tube, PC board with battery holder and on-off switch, and cable for unbalanced-line use, for about $100. (But if you want to spend that much money, check out AKG's new electret mike, not a kit, at $115.) BAS members can obtain the 814 module, or the 814C if you prefer, via a BAS group purchase for less than $30.