THE B.A.S. SPEAKER

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

VOLUME 3, NUMBER 6
MARCH 1975

In This Issue

This is very much a hardware issue; nearly everything reflects it, from the meeting report on Koss headphones and electrostatic speakers to the publications. The long-awaited report on the QDC-1 phono cartridge appears this month, and with it an interesting companion piece. During the concentrated listening triggered by the QDC-1 comparisons, some members found themselves reevaluating old favorites; you'll be interested in the results of a runoff between the ADC-XLM and the Shure V-15 Type III.

There's interesting, if brief, mention of tonearm damping again, this time as applied to a popular record changer—some of the problems are different. And on the topic of tonearms, one of our out-of-state members weighs in with a pointed indictment of some of the most misleading advertising in recent years. See the letters column for this and for short notes on a modification to the Citation XI preamp and a multiple speaker installation in Kansas City.

There are short reports on two cheap products, one of which costs a lot more than it should—a so-called reverb amp. The other may be one of the few real bargains about—a $25 FM converter for your AM car radio that works well beyond its price category.

Finally, there's a brief review of an excellent book on audio; one filled with fact instead of pseudo-science and hype, Slot's Audio Quality. As might have been expected, it didn't sell too well, and thus wound up on the remainder tables. Thus, it is available cheap.

This Month's Publications

Dan Shanefield may go further toward the ultimate in A-B testing than most of us. In this second article on the subject, he deals with comparisons of high-fidelity equipment with live performers. The conclusions he draws as to the state of the hi-fi art are reassuring. Also, there are diagrams for some of the A-B switching gear needed for such experiments. Shanefield is collecting a complete chain of components recommended by the Absolute Sound and plans to find out whether there is much difference between it and the equipment already in his system. As he finds out, we'll pass the word along.

This month's second publication is by Al Southwick and in it he makes a case for the 814 microphone. I suspect the pun is intended. But those of you who have or plan to buy 814 or 814C mike capsules through the BAS will want to read this short paper on turning it into a rugged, good-looking, go-anywhere mike, complete with phantom powering.— Jim Brinton

Equipment for Sale

• Burwen DNF-1201 noise reduction unit, $200. Call David Kunz (617)481-1219.

The QDC-1 Phonograph Cartridge

After a long delay, we finally are able to give some answers to the questions we raised several months ago about the Micro-Acoustics QDC-1. Unfortunately for those of you who may have purchased one in the enthusiasm that greeted the QDC, the recommendation is largely negative.

At least half a dozen of us have auditioned and objectively tested more than ten QDC-1's and, to one degree or another, each unit failed to live up to expectations. The strength of the expectations is illustrated by the fact that some of us tried as many as three QDC-1's before finally giving up.

Since this stance is negative, it is appropriate to say something about the sort of tests we conducted. The objective tests were run using CBS, Shure, and other test records, and in each case a cartridge of a different brand was tested just before or after the QDC to assure us that our test setup was functioning correctly. The record-playing systems used varied from a modified AR turntable equipped with twin SME arms, to Technics, Connoisseur, Philips, Braun, and Thorens tables. Other tonearms ranged from the highly sophisticated Decca to the AR. Listening and A-B tests were conducted with a wide variety of program material and electronics—too long a list to give here. Among the speakers used were KLH-9's and AR-LST-1's. Whatever the hardware, however, the results were mixed at best.

Early Results

During the initial burst of enthusiasm for the QDC-1, several of us purchased samples and after listening, put them through some objective tests. Afterward, we found it hard to account for the disparities; objective tests seemed to indicate that the QDC-1 would be a poor-sounding cartridge. Very careful listening bore this out—but not to the extent indicated by the measurements.

For example, one early production unit had a 3.5-dB response hump at 18 kHz, a 2-dB dip at 8 kHz, and was down 2 dB at 50 Hz. Square waves showed a great deal of undamped ringing, and worse, most of this ringing appeared in the opposite channel, indicating poor transient crosstalk.

This was not the worst cartridge measured (others were up 7 and 10 dB at 20 kHz, for example), nor was it the best of the QDC-1's tested, but it was near average for the ten sampled.

Compared with the ADC-VLM and Shure V-15 Type III, the QDC-1 came in third. The VLM had excellent square waves, with little ringing, and very little crosstalk, whether steady-state or transient.

The Shure rang some but appeared well-damped; there was some rise in its response at very low frequencies, and more crosstalk than the VLM exhibited—especially at lower frequencies, though nothing like that seen with the QDC-1.

Tracking tests made with the 300-Hz band on the CBS STR-110 record produced these results: the Shure easily tracked 18 cm/sec laterally and 12 cm/sec vertically; the ADC began to break up at 15 cm/sec vertically and ran well at 12 cm/sec laterally; the QDC-1 broke up at 9 cm/sec lateral modulation and failed to track at 12 cm/sec vertically. Thus the QDC-1 appeared deficient in low-frequency tracking; this was borne out in listening tests.

On the other hand, the QDC-1 was better at high-frequency tracking. The Shure TTR-103 record includes a cosine-modulated 10.8-kHz tone burst that increases in amplitude from 15 to 30 cm/sec. While the QDC-1 took this well, both the Shure and ADC-VLM broke up.

The addition of damping to later production QDC-1's smoothed out frequency response a bit, but changed little else. For the record, here is how a late production unit measured using the CBS STR-100 test record: down 1 dB at 20 kHz, up 2 dB at 16 kHz, up 2.7 dB at 14 kHz, flat at 10 kHz, down by 0.9 to 1.8 dB from 8 kHz to 2 kHz, flat from there to about 300 Hz, and then down by about 0.6 dB between 50 and 30 Hz. At 20 Hz, output was flat again relative to 1000 Hz.

However, the unit was able to track the Shure TTR-100 bass bands only at a stylus pressure of 2.3 grams and made it through the high-level bands of the TTR-102 at 1.75 grams. Again this isn't poor performance, although it can be bettered by the Shure, ADC, and other cartridges. Note that Micro-Acoustics specifies a maximum tracking force of 1.5 grams.

How It Sounded

Extended listening by several parties usually resulted in the QDC-1 losing to some other cartridge like the XLM, VLM, Shure V-15 Type III, or Ortophon.

Some discs, notably those without deep bass or with a dull upper midrange or high-frequency characteristic, sounded better with the QDC-1. But this is the sort of effect better left to an equalizer, it was felt.

Subjective Comments on the Sound

"Strings sound zippy in comparison with Shure, Ortophon SL-15E Mk II, and ADC-XLM- More hum-prone than any of these three cartridges."

"QDC-1 refuses to track pedal notes on (my) organ records, and was extremely sensitive to acoustic feedback and vibration even in a damped tonearm."

"Unit had more 'ambience' information in all recordings than the Shure. There also appeared to be more 'air' around instruments."

"A 'kaaahhh' sound from the cartridge gives it a reticent sound, while some of its high-end brightness tends to make other sounds brittle."

"QDC has a strange combination of lack of detail and over-brightness."

"What seems to be a subtle hump near 300 Hz makes music sound as if it were coming from a barrel."

"Some records sound so 'hard' with the QDC-1, they hurt."

"A previously good-sounding record now may sound poorly mixed with the QDC-1, an effect hard to understand. Sounds are less well blended than with other cartridges (this comment, of course, cuts both ways. . .)."

"QDC-1 enhances cymbals, triangles, and attack sounds generally, and perhaps unnaturally."

"If this is the best cartridge in the world, then 99 percent of (my) records are unlistenable."

The QDC-1 is not a "bad" cartridge and it makes a few records sound better than before, but we see no reason for paying even half the asking price for the level of performance it achieves. There are more than half a dozen cartridges available (Shure, ADC-XLM, ADC-VLM, Ortophon, Decca, and others), some costing a good deal less, and most audiophiles will prefer their performance and sound.

Contributors to this report included, but weren't limited to, Joyce and Jim Brinton, Al Foster, Bob Graham, Rene Jaeger, and Peter Mitchell. — Jim Brinton

ADC-XLM Versus Shure V-15 Type III

In "The Audio-Technica Cartridges" (BAS Speaker, April 1974), I concluded that the ADC-XLM was in general superior to the AT's and, by implication, to the Shure V-15 Type M- When that article was written the XLM was mounted in a damped Connoisseur BD-2 and the Shure was not. Since that time I have had the opportunity to listen to several Shure and XLM cartridges all properly mounted in damped tonearms, and as a result my opinion has changed.

The XLM has more cult appeal than the Shure and attracts that small group of audio perfectionists who are always in pursuit of the best. ADC is always introducing a new improved version, and each requires expensive, low-mass tonearms to function properly. The latter two qualities combine to give the XLM a special attraction to the purist—who often is of the opinion that all good things must be delicate and require sacrifice.

The Shure V-15 Type III by contrast is mundane in its appeal. It is a large seller, and requires no special consideration. The manufacturer even claims that it works very well with some of the higher quality automatic record players.

Neither of these two descriptions is completely accurate. For best performance the XLM and Shure both require low-mass tonearms, which, it appears, must be damped, the XLM even more so because of its extremely high compliance. In fact, with one sample of the XLM, stylus "bobbing" would not completely disappear no matter what tonearm was used or the amount of damping applied.

The testing procedure consisted of blind A-B testing with several groups of audiophiles, two completely different stereo systems, and identical discs. Output from each cartridge also was recorded on individual tracks of a stereo tape recorder and then A-B'ed.

The Shure was preferred by each group. It was described as (here we go with the vague terms) smoother, more open, and having far more detail than the XLM- In the bass the Shure was "fatter" or less well-defined, though it did seem to go lower. The XLM often sounded harsh, particularly on flutes, and did not fare well on strings. The Shure sounded slightly more forward and had more "air" around the instruments, although the frequency response of both cartridges was very linear.

Both were tested to see what effect varying load capacitance would have on their performance; as expected, the Shure's varied the most but the results were barely audible. Without added capacitance, using the Dual 1218 turntable and the Marantz 7C, the frequency response of the Shure was: +3 dB at 20 kHz, +2.3 dB at 18 kHz, +1 dB at 16 kHz, and flat from 14 kHz to 20 Hz. Adding 300 pF to the left channel and 200 pF to the right channel made the Shure flat to 20 kHz. In the Decca arm no extra capacitance was necessary to achieve flat response to 20 kHz (which indicates that Decca leads have high capacitance—a disadvantage for CD-4). The older Shure V-15 Type II tended to alter its frequency response in the more audible region around 5 kHz; the manufacturer claims that the Type III is less sensitive to capacitance effects, and the new cartridge is not affected below 14 kHz.

The Shure tracked a hair better than the XLM; however, this was not revealed until the Shure TTR-102 test record Vas used. Band seven of this record contains 400-Hz and 4000-Hz tones recorded with peak velocities of 24 cm/sec. When mistracking occurs, the results are visible on a scope and are very audible. The Shure did not mistrack on band seven, while the XLM did mistrack intermittently.

For this report I sampled three different XLM's using four different styli. The oldest sample was about eight months old and the most recent was purchased within the last three months. The newest stylus was so compliant that neither any tonearm nor any amount of damping would produce as stable an oscilloscope trace as any of the three sampled Shure cartridges. In my opinion the most recently purchased XLM was too unstable even for high-quality playback systems. This same cartridge in an undamped tonearm produced large near-dc current surges and exhibited gross intermittent mistracking. The results improved tremendously in the damped arm, but the XLM still was not as stable as the Shure. — Al Foster

Damping a Dual 1019

I soldered a sheet-brass paddle to the tip of the counterweight shaft on my Dual 1019. The shaft was stiffened by folding over three layers of the 1/2-inch-wide brass strip. A right-angle bend in the strip formed the horizontal paddle (for vertical damping). Each paddle was 1/2-inch-wide brass strip. The STP tank was formed from thin sheet copper and soldered at the seams. (Sheet copper and brass are available from school-supply outlets—Hammetts in the Boston area—and hobby shops.)

Warning. The first result was a lesson in wave theory. When I hit the stop control, the vertical paddle on the returning arm pushed up a 1/4-inch wave in the tank filled to within 1/8 inch of the top. Thank Granitelli, the goo flows so slowly I was able to grab a nearby scissors and scrape the stuff back up the outside before it hit wood and works. Leave at least 1/4 inch between the fluid surface and the top edge of the tank on automatic turntables.

The second result was that in the automatic mode, during the lengthened arm-drop time, antiskating compensation pushes the arm back toward its rest, missing the outside of the record. This renders the Dual's automatic mode useless.

Most records are mildly eccentric. A visual check showed substantial lateral needle motion with the 1/2-inch-square paddles. I cut the vertical paddle down to approximately 1/4 inch square, but I still see a slight motion. There was far less vertical needle deflection on warps but I cut the horizontal paddle down to 1/2 by 1/4 inch.

The two paddles must be truly horizontal and vertical. If either one is tilted, it will convert deflections in one axis into a force along the other axis.

A detrimental effect was increased vibration sensitivity. I had to mount the rather large tank on the mounting board. While the turntable is isolated on its springs, floor vibrations reach the mounting board and are transmitted via the STP directly to the arm.

I still think it's worth it. Subjectively the sound seems more stable and clear. High-frequency distortion using the Shure V-15 Type III seems noticeably lower. Using the Shure laterally cut high-frequency tracking test (TTR-110) and listening to a L - R null signal indicates that high-frequency tracking has improved. — Joel Cohen

Notes on the Thermo Electron 814

If you are using the mike capsule naked rather than installing it in a grounded metal tube or case for shielding, you may find that it picks up radio-frequency interference (RFI) when used in very strong signal areas. A 0.001-μF capacitor wired across the signal terminals will eliminate or substantially reduce the RFI.

The BAS price for the mike capsule is $27 for the 814 and $32 for the 814C. This includes sales tax and expenses. If you cannot pick up the mike in person, add 50¢ for shipping.

Thermo Electron has published new spec sheets on the 814 and 814C. They indicate that impedance is 1700 ± 700 ohms for both mikes (in contrast to a measured value of 500 ohms for one of my 814's). They also indicate an even steeper high-frequency rise on-axis than formerly, underlining the importance of aiming the mike 60 to 90 degrees away from the sound source (e.g., up toward the ceiling) in order to get a smooth top end in your recordings. — Peter Mitchell

The Pioneer Reverberation Amplifier

The combination of heightened interest in delay-generated ambience and the recent emergence of the Pioneer SR-202W Reverberation Amplifier ($139.95)—with a front-panel control promising variable "time" out to 2.5 seconds—resulted in a very brief test.

Hnspection showed that the Pioneer was a spring-based unit similar to the old Fisher Space Expander, with which it was compared. The only real difference was that the Pioneer had a smoother frequency response. The Fisher has several resonances and a tinny sound.

The reverberant signal is generated by exciting two springs with slightly different transmission times. The resultant signal has an initial delay to 90% of original amplitude of about 50 milliseconds; decay to 10% of original input level takes about 2.5 seconds.

Both units mix the spring-generated reverberant signal with the original, and front-panel controls (including the misleadingly labeled "time" control on the Pioneer) vary the relative amplitude of reverberant and direct signal. Raising the relative amplitude of the reverberant signal does make signals transmitted through the units seem to "last longer," but that business about time should be corrected. Spring-generated reverb isn't unpleasant, but it is only marginally useful. — Joel Cohen, Al Foster

Going FM with "Brand X"

Recently on Shop Talk, one of Boston's better known audiophiles commented that his automobile's FM radio was unable to receive WBUR beyond the Route 128 perimeter (about 15 miles from the transmitter). At the time I was listening in Acton, perhaps that far beyond Route 128, through an FM converter attached to my AM car radio.

The device is not "high fidelity" but to be able to listen to WBUR, WGBH, WCRB, or WHRB at all is surely better than WEEI—"all news, all day." I've had no trouble listening to these stations far to the west of Boston and have always found good stations available most of the way to Tanglewood. If you do not have FM in your car now but do have a built-in AM radio, then I recommend the purchase of such an FM converter.

Converters operate on the principle that rather than building a complete FM radio, it is easier and less expensive to build a device that will receive FM, convert that signal to AM, and then play the converted signal through the already available AM radio. The converter is tuned to the FM station desired and its output is received on the AM radio near 1400 kHz.

A single IC does most of this conversion job, making the packaging of the complete converter very inexpensive. There are several brands of converters on the market, but I believe there are at most two circuit designs behind the packaging. The converter I have is extremely compact, measuring about 1 by 4 by 5 inches, which makes hiding it from view and theft easy. Installation is simple with only one power wire to solder and two antenna cables to push into appropriate sockets on the rear of the converter and the AM radio. There is no brand name on my unit, only the label "Integrated Circuitry" and there is not even a manufacturer's name, only the warranty address—"National Service Department, Cleveland, Ohio." I've seen the unit advertised lately by Zayre for $25 but I believe the same unit appears in the Olson catalog (No. 275, p. 35, Mini, $24.99) and the Lafayette catalog (No. 750, p. 97, Micro-Mini, $29.95). I believe it is also the same converter being offered by WCRB for $26.95 plus postage and tax.

The converter suffers from multipath interference in the city, especially near Boston University and when between the tall buildings near the Prudential Center, but for the $30 I paid, I have received three years of trouble-free FM.

I don't understand it but many times I have found that the converter can beat the pants off factory installed FM radios in American cars (e.g., Blaupunkt) — Harry Zwicker

Use Report

We found the "Brand-X" FM converter to be a decent performer. We compared it with a Blaupunkt AM-FM-SW unit, and found that while there were some significant differences in design, there were only small ones in performance, and sometimes "Brand X" won out.

Sensitivity was good, perhaps a little higher than that of the Blaupunkt. WBUR, WCRB, and WGBH were received easily almost to Worcester using a vertical whip antenna; this sort of performance wasn't possible on the Blaupunkt without switching to a horizontally polarized strip antenna.

Sound quality was good, but was limited by the performance of the AM tuner used with the adapter. On a cheap AM-only radio, however, the lower distortion and broader frequency response of FM signals were still apparent. A-B-ing WCRB via AM and FM showed not only was the FM signal louder, but cleaner and far less distorted as well. Apparently this cheap AM radio was better than expected, for performance had been very poor without the adapter.

Naturally, the highs tend to be somewhat rolled off unless one is using a high quality (i.e., Blaupunkt or similar) auto radio with the converter. But is is fair to point out that there is a tradeoff here between cost (about $25 for the converter) and high-frequency response. You can pay from $10 to $100 more and get the highs with a dedicated FM auto radio, or you can remember that the ambient noise level in most automobiles is so high that you might not be able to hear the last octave anyway.

The converter was no more susceptible to multipath than any other FM auto radio we were familiar with. In downtown areas where multipath caused by large buildings formerly was troublesome, it was a problem with the converter also—but no worse a problem.

In sum, the RF performance of the unit is surprisingly good and its audio performance will be controlled mostly by the radio with which it is used. If about $25 and 30 minutes of installation work seems about right to you, this is a good buy. — Joyce and Jim Brinton

An Excellent Book

At the February meeting Harry Zwicker recommended a book so good that every audiophile—and the editors of all of the magazines—should read and study it. Audio Quality, by G. Slot, was initially published in 1972 in Europe as one of a Philips series on audio. (The other books in the series that I have seen are much less interesting.) The U.S. publisher is Drake, 381 Park Avenue South, New York, New York 10016.

What makes the book so good is that its 150 pages are populated with facts instead of the assorted generalizations, opinions, and speculations that turn up so often in hi-fi writing (even, occasionally, in the BAS Speaker). It is very much easier to guess at the truth of a situation than to search out the relevant facts. It must have been a big job to write a book as full of useful data as this one is, and G. Slot (whoever he or she is) deserves credit for a considerable accomplishment. Despite its solidly factual orientation, the book is not a technical treatise. It is written for the lay audiophile in plain English (British English at that), and heavy equations are confined to a short appendix.

The core of the book, 100 pages in length, is a series of discussions of the basic audio equipment parameters (power requirements, distortion, frequency response, pitch deviation, S/N ratio, stereophony, and ambience). In each chapter the relevant acoustic and psychoacoustic data are presented, often in the form of graphs reproduced from original research papers, with references given for further reading. These include, for instance, data on the acoustic power of instruments and orchestras, the acoustic attenuation of residential construction materials, the ear's harmonic distortion curves, the ear's masking of IM as a function of SPL, the audibility of pitch deviation, typical pitch error in live music, the perception threshold of wow and flutter, the background noise level in studios and homes, the masking of noise by music, the disturbance of the stereo image by amplitude (i.e., balance), phase shifts, and crosstalk, and the function of time delay in reverberant-field reproduction.

Audio Quality is not a how-to-do-it book, nor does it give buying advice. But after you read it two or three times you will have a better understanding of the basic issues in sound reproduction and the basic limitations as well. You may be surprised to find that some of the things you "know" really aren't true, and many popular myths and assumptions are based far more on speculation (or on marketing needs) than on empirical evidence.

But you will have to hunt to find the book. It has been discontinued and is officially going out of print. It turned up at Reading International selling for $2 instead of its list price of $6, and their small supply disappeared quickly. Search for it in other stores that carry "remainders" at discount prices. If you find a dozen copies somewhere, grab them; the BAS will buy them for resale to members. — Peter Mitchell

Letters

Transcriptors Vestigal Arm

S. P. Lipshitz of the Department of Applied Mathematics of the University of Waterloo, Waterloo, Ontario, has written a scalding letter to The Absolute Sound in which he takes issue with the "Manufacturer's Comment" following the review of the Transcriptors Vestigal (sic) arm. A copy was forwarded to us in early January, and since we have been given permission to reprint it, and since we have read nothing about it in The Absolute Sound, we reproduce it here with only minor cuts.

"I am writing in connection with your review of the Transcriptors 'Vestigal' arm on page 47 of Vol. 2, No. 5, and in particular about the comments from David Gammon of Transcriptors Ltd. I also have on hand the Transcriptors advertising brochure describing the Vestigal arm. I do not think I have ever come across such blatantly false and technically inaccurate statements in any advertisement for any high fidelity product. Moreover, these same statements are made by Mr. Gammon in his comment on your review. I would therefore like to analyze in some detail the principle claims made for this arm and the mechanical laws which govern its behaviour. It should be understood that my comments are intended as a criticism, not of the arm's actual performance, but rather of the manufacturer's ridiculous claims for it.

"Let us examine matters point by point:

"(1) Arm inertia: Consider the distinction between inertia and tracking force which Mr. Gammon is at pains to explain to 'ignorant' American audiophiles in his comments, and which is also elaborated upon in the Transcriptors literature. Although Mr. Gammon is correct that the moment of inertia of a mass about a given axis increases as the square of its distance from the axis, the conclusion which he draws (namely that conventionally pivoted arms like the SME arm, because of their higher moment of inertia, are inherently inferior) is false. Inertia per se is irrelevant, and the only significant figure in regard to an arm's moment of inertia (in that it affects the amount of work the record has to do in moving the arm) is its effective mass at the stylus point. This is its moment of inertia divided by the square of the distance of the stylus from the pivot. So, since the SME arm's pivot-to-stylus distance is 9.375 in. as opposed to the Vestigal's 1.375 in., it can have 46.5 times the Vestigal's moment of inertia and still have the same effective mass.

"Now consider a few figures. According to Transcriptors, the Vestigal arm including cartridge has a moment of inertia of 120 gm • cm^2 in the vertical plane. With a stylus-to-pivot distance of 1.375 in. = 3.5 cm this gives an effective mass of 9.8 gm. Since the ADC-XLM cartridge weighs 5.5 gm, and allowing 0.5 gm for mounting screws, we find that the Vestigal arm itself has an effective mass of 3.8 gm. SME claim an effective mass of 5.5 gm for their nondetachable-headshell arm and Hi-Fi News' measurements (May 1973, p. 1013) give a figure of 6.75 gm for this arm, while Hi-Fi Sound (February 1973, p. 90) measured 6 gm. Thus we find that the Vestigal and SME arms have effective masses of approximately 4 gm and 6 gm, respectively. With cartridges, these figures are approximately 10 gm and 12 gm, respectively, the difference being very small in favour of the Vestigal.

"Since the force required to move an arm over warps is determined by its effective mass, including cartridge, there is little theoretical difference in their warp-tracking abilities. This conclusion is borne out by observing the deflection of the stylus (cantilever) as both these arms track a warped record. Their deflections are comparable. In no way can Gammon's assertion that the SME offers 73 times as much resistance over warps as does the Vestigal' be supported.

"(2) Arm/cartridge resonance frequency: Our conclusions in (1) indicate that the vertical resonance frequency of the Vestigal/ADC-XLM combination is only marginally different from that of the SME/ADC-XLM combination (the resonance frequency varies inversely as the square root of the total effective mass), and certainly is not 180 Hz as claimed by Transcriptors. Neither is such a high resonance frequency desirable, let alone a resonance frequency of over 30 kHz as advocated by Transcriptors for a 'perfect arm.' At frequencies below resonance, the stylus does not deflect relative to the cartridge, and so the cartridge produces no output signal. Hence a resonance of 30 kHz would result in no output, while the 180 Hz claimed would result in no bass output!

"The fact of the matter is that the resonance frequency must fall above the warp frequencies (which are below 10 Hz) in order that they should produce little output, and below the audio band (which is above 20 Hz) in order to produce full audio output. The optimum resonance frequency is thus 10 - 15 Hz, and the Vestigal/ADC-XLM has its resonance in this neighbourhood, and nowhere near 180 Hz. As long as discs have any warps, cartridge compliance must be matched to effective arm mass, and the claim of Transcriptors that infinite compliance is desirable as a goal is patent nonsense.

"(3) Arm geometry: Finally, a few comments on some design inadequacies of the Vestigal arm. The manner in which the vestigial section of the arm is pivoted results in significant cartridge tilt as well as change in vertical tracking angle as the cartridge tracks record warps—much more so than conventionally pivoted arms. Furthermore, the method of counterbalancing the vestigial arm causes tracking force to vary considerably as the arm moves over warps.

"To end, I would like to re-emphasize that these criticisms are aimed at the incredible advertising claims made for the Vestigal arm, and not at its performance. It is, however, difficult to justify that 'nine years of intensive and original research' (Transcriptors' advertising claim) should have been undertaken on this arm by someone with so faulty an understanding of basic mechanics as is demonstrated above. Let me also state that I have no connection with any audio company whatsoever." — S. P. Lipshitz (Waterloo, Ontario)

A "Mod" for the Citation XI

' The article by Peter Mitchell describing a modification for the BSR-Metrotec equalizer (BAS Speaker, March 1974) was of great interest to me as I have a Citation XI preamp with a similar five-band equalizer. I found that the Citation evidently uses circuits like those of the Metrotec, and then concluded that the lower two of the equalizer's controls could be shifted to operate at 28 and 110 Hz (instead of 60 and 320 Hz) by changing C607 and C608 to 33 and 3.3 μF, respectively. The values are from the formula for the resonant frequency for an L-C circuit, where C is in farads and L is in henries:

$$C = \frac{1}{4\pi^2 f_T^2 L}$$

"The change is very easy to carry out, and my results are quite in accord with Peter Mitchell's observations. The sonic improvement with judicious use of the controls now is downright exciting, and has been achieved at very small cost. I have not found any problems with rumble, hum, or other disturbances using Connoisseur turntable, ADC-XLM cartridge, Dynaco 400 amplifier, and double Advents.

"To . . . other topics. I thought the Shanefield article about A-B testing was great—original, clever, lucid, and profound, all at once. I also liked the comparison of the three high-powered amps, which I thought . . . well done.

"I might say I was not surprised by the results, though I was dismayed to see 'Dynaguard' equated with a fuse. I recently built the 400 and the Dynaguard function, in my opinion, is so needed and so effective, that I would not like to use a high-powered amplifier without it. As I understand it, it passes short-duration peaks that would blow a safe fuse, while for lower level peaks, the permissible duration increases, and there are five choices of maximum output. And as to another facet of the discussion, my unit, at the discount price of the kit, costs 85¢ per watt, not counting shipping.

"The range and depth of your publication continue to be exciting. I should hate to be without it." — Bill DeMond (Walla Walla, Washington)

Pairing Loudspeakers

"Reference to 'the occasional feeling that the speakers (used at the October meeting) were bass heavy may have been caused by the pairing of the loudspeakers for each channel' (BAS Speaker, Nov. 1974) reminds me of favorable comments on double Advents in The Absolute Sound and of a demonstration of a system devised by David Beatty (a long time Kansas City, Missouri Audio dealer).

"Possibly the system would not respond well to (laboratory) measurement, but I felt it to be one of the most pleasant sounding I've heard, largely, I think, because the sound didn't seem to come from specific loudspeakers, but from a wide area. It seems to me at this time that two additional Advents will give me the greatest improvement for a reasonable cost that I could possibly get. Possibly The Speaker might include some comments on the pros and cons of four similar speakers for stereo operation.

"Finally, the November and December issues . . . have given me a great deal of information . . . which I would probably have been unable to get elsewhere. I enjoy reading the material and appreciate the effort of the members who contribute time, work, and real expertise to produce an outstanding publication." — O.H. Stewart (Lincoln, Nebraska)

(The Beatty system that member Stewart writes of uses four identical speakers, but places them somewhat farther apart than would be the practice here. For example, on a 16- to 24-foot wall, they would be from 5 to 7 feet apart. Each of the two outer speakers is angled toward the center of the room about 20 to 30 degrees, and the inner speakers by about 45 degrees. For use with this array, Beatty sells what appears to be a switched voltage divider to vary the amount of "center fill." It is not clear whether any ambience information is fed to any of the speakers, though from the description in Beatty's literature it seems unlikely. Price for the "Wall of Sound" control is $79, and we would like to hear more from any member who is familiar with it. Members who want more information should write to David Beatty Stereo, 1616 Westport Rd., Kansas City, Missouri 64111. Mention the BAS.—Ed.)

In the Literature

Audio Amateur, Issue 3, 1974 (Current)

- Articles include a patch panel, a suggestion for accurate frequency response measurement with an audio generator, part II on the preamp console, good advice on grounding techniques in multistage audio amplifiers, more on the PAT-4 phono equalization and on the AR arm, a report on an electret microphone kit using the Thermo Electron capsule, the final installment in the "Learning to Use Transistors" column, and the usual gaggle of interesting letters. Also includes a good commentary by Ed Dell on the big American magazines' practice of publishing only favorable equipment reviews.

Audio Scene/Canada, Feb. 1975

- Damping, Damping Factor, and Damn Nonsense: Argues that since the resistive portion of speaker impedance (about 90% nominal impedance) adds to the amplifier output impedance to determine the actual damping effect, amplifier damping factors over 1.0 are academic—with supporting test data.
- Audio Lab Tests: Includes on-axis and off-axis frequency response curves for the Klipschorn.
- Putting Together a Sub-Woofer System: A 12-dB-per-octave low-pass filter, a 10-watt amplifier, a 15-inch speaker, and an infinite baffle provide unspecified "earthquake" effects.

Electronic Design, Feb. 1, 1975

- Program Multichannel Audio Gain: Use of voltage-controlled FET's for keeping tracking errors low without the use of precision matched potentiometers. Includes discussion of other uses of voltage-controlled resistance (e.g., in an expander) and design hints for controlling distortion and dynamic range. (p. 68)

Electronic Design, Feb. 15, 1975

- Electronic Music Incorporating . . . IC Technology: A brief review of electronic music consoles and electronic organs. (p. 26)
- Easy to Build FM Signal Generator: Use of 562 PLL in a 10.7-MHz FM generator for tuner alignment or test. (p. 94)

Hi-Fi News and Record Review, Jan. 1975

- Equipping an Amateur Hi Fi Workshop, Part I: First part of a series of articles describing the construction of five simple pieces of electronic test equipment, this part being an introduction to the series.
- The Disc—Introduction and The Disc—History and Development: A planned three-month study of records by many well known authorities (e.g., Ben Bauer, John Eargle, Duane Cooper).

IEEE Transactions on Audio and Electroacoustics, Vol. AU-18, 1970

- Transient Distortion in Transistorized Audio Power Amplifiers: One of the earlier references to TIM by M. Otala. (p. 234)

Popular Electronics, March 1975

- Tape Bias and Equalization: A general discussion without too much how-to-do-it instruction, but with mention of the dbx asperity bias method as gleaned through the BAS Speaker (p. 16)
- 4-Channel Equipment Report: Skip it. (p. 58)

Radio Electronics, March 1975

- Switchcraft 621 Dolby FM Compensator: A review. (p. 22)
- This FM Tuner Costs $2500: A review, with partial schematic, of the Sequerra Model 1. (p. 30)
- One Sided Noise Reduction System: A review of the Burwen DNF-1201 consumer version noise-reduction unit. (p. 37)
- . . . Tape [units], Reel-to-Reel vs. Cassette: A very brief survey of tape recorder features. (p. 40)
- FM Tuner Roundup: A survey of FM tuner features, with a two-page chart of published specs (including the Pioneer TX-9500). (p. 44)
- NESDA asks FTC . . . to rule on warranty compensation: A brief announcement of a request for full pay for warranty service performed at local service shops. (p. 69)

Wireless World, Jan. 1975

- Letters to the Editor section includes an item on "settling time" in audio amplifiers, comment on "reducing amplifier distortion," and a comment on the performance claims for the Radford 250 amplifier/preamplifier with response from Radford. (p- 18)

February BAS Meeting

Business and Open Discussion

Despite several inches of wet snow, the meeting drew over 50 diehards. Discovered among the audience was Leigh Pheonix, recent author of a tonearm damping analysis and AR arm modification article (January 1975 Speaker). While waiting for late arrivals, members questioned Leigh in detail about the AR arm modification for decreasing mass and adding damping to the bearings. Among other things, the arm and head shell were drilled out as much as possible and ST P added to the cone bearings of the vertical pivot. Pivot screws were tightened and then backed off 1/8 turn. Leigh felt these modifications, while helpful, did not yet give optimum results.

To aid in the optimization of tonearm/cartridge resonance characteristics when removing (or adding) weight, or introducing damping, Al Foster suggested a test record having a frequency sweep from 100 Hz all the way down to 5 Hz. This is just the thing to use with a scope or VTVM to find out what the arm is doing. The test record is produced by the Components Corporation and may be obtained at Lafayette under their number 24/45070.

According to Peter Mitchell, the Radio Shack sound-level meter compares very favorably with professional units costing much more, e.g., Scott and B&K. In calibration tests, its response was ±1 dB from 150 Hz to 10 kHz. The same comment was made by DGG engineers during the recent BSO recording facilities tour held by the AES.

Peter has also been experimenting with a center speaker, hooking it between the left and right channel hot terminals of the amp. This produces a L - R signal in the speaker and is essentially an ambience-recovery system. An L-pad is used to adjust center-channel volume. Peter claims this hookup, contrary to the normal L + R center channel, adds spaciousness and transparency to the sound.

After rearranging his system to obtain better clarity and altered high-end response, Dick Goldwater withdrew his enthusiastic endorsement of the Micro-Acoustics QDC-1 cartridge. (See note in this issue.) He felt more listening was in order before he could comment further.

Meeting Feature: How Stereo Headphones are Designed and Manufactured

Guest speaker at this meeting was Howard Souther, Senior Engineering V. P. at Koss Corporation. With a long-standing interest in electronics and audio, Souther's background includes work at Langevin and at Electro-Voice, where he designed speakers. At Koss for the last 10 years, he is most recently responsible for the design of the PRO-4AA headphone drivers, the electrostatic headphones, and Koss's newest product, a full-range electrostatic speaker.

Psychoacoustic Effects. Souther began by discussing how psychoacoustic phenomena affect the way we perceive sounds. By drawing an analogy with visual perception, sound can be said to be experienced in the mind through the fusion of the two separate auditory inputs. In addition to providing localization cues the synthesis of a single sonic image in the brain creates the illusion of depth and solidity in the sound source. In sighted persons the sound image is almost never independent of the visual image; the brain relies heavily on optical inputs to interpret the aural input. For instance, a person hearing a plane fly by interprets the sound as coming from above even before looking for the plane. Yet it is difficult (if not impossible) for the ears to differentiate between sound locations which are mirror images in the symmetry plane of the head without some additional cues. These cues are usually from visual inputs and changes in the auditory perspective through slight movements of the head. The head moves, in essence, to "feel out" the sound field and to eliminate the ambiguity of a single perspective.

Because of these multiple inputs it is not commonly recognized that the ear has no inherent front-back discrimination. How then are quadraphonic headphones possible? Here the power of suggestion provides the needed focus for the sound field. For example, when a person wearing a pair of quad headphones reads the instruction manual telling him to turn the control to the right and hear the front channels come up and to the left and hear the rear channels emerge, sure enough, it works. But then he discovers the headphones are on backwards. Turning them around and juggling the controls, he finds, low-and-behold, the front channels are still in the front! They are there because he "knows" that's where they're supposed to be; and so they are.

In attempting to make a case for multiple-miked recordings, Souther again invoked psychoacoustics. Even though we listen to and enjoy live music with two ears, recordings that use only a stereo pair of mikes (he said) tend to sound thin and have less impact because the visual element of the concert hall is missing. Multiple miking is an attempt to make up for this by giving body, fullness, and immediacy, sometimes to the point of making the sound "super-real."

Further justification for individual miking was offered with the observation that concert halls generally show a falloff in response at the high and low ends of the spectrum. To reestablish a balanced sound, bass and treble instruments are enhanced through close miking. The former point was immediately challenged by members of the audience who claimed that this might be true in Milwaukee, but it certainly is not the case in Symphony Hall, Boston.

In reviewing the subjective effects of musical sounds in various regions of the audible spectrum, Souther characterized the deep bass as being the foundation of our musical experience, having tremendous emotional impact, while the upper treble (beyond 9 kHz), although important, is not as readily noticed if missing. This he attributed to our having a "long memory" for bass and a "short memory" for treble sounds. Bass impact is probably attributable to the fact that it is a physical as well as an aural experience. Although musical notes seldom extend below 32 Hz, we sense low-frequency energy more than another octave below this. Wind, thunder, earthquakes, and room effects account for most of the "sounds" in this region.

Proper balance must be maintained between the upper extent of the treble and the lower reach of the bass for natural sound reproduction, he said. It has been a rule of thumb for motion picture reproduction systems that the product of the upper and lower cutoff frequencies should roughly equal 500,000 (e.g., 50 Hz x 10,000 Hz). Otherwise the system is likely to sound either too shrill or too boomy (assuming, of course, a flat frequency response).

Other more familiar psychoacoustic effects were mentioned briefly. Aberrations in the frequency response of a sound system, such as emphasis of the midrange or upper midrange, can lead to prominent presence or enhanced clarity. Certain kinds of distortion result in early listener fatigue.

Headphone Design. Although psychoacoustical factors play an important role in influencing the broad design philosophy for a piece of audio gear, basic physics and engineering determine what can be done practically, and how to do it. This is especially true of transducers.

The first headphones for high-fidelity listening used miniature loudspeakers as the transducer elements. Many still do. However, deficiencies in these elements have led some manufacturers to introduce improved designs. All headphones now on the market can be classed as either velocity type or pressure type. The velocity type are light and rest on the outer ear so that room sounds can be heard through them. Resting on a cushion around the ear and sealed from outside sounds, the pressure type tends to be heavier but is generally conceded to produce the better sound.

The HV-1, a velocity type headphone recently introduced by Koss, utilizes a dome-shaped PVC diaphragm as the driver element. This is felt to be a major improvement over the cone drivers where a midrange dip in response often resulted. Sound waves on the cone reflect from the cone edge support and propagate back toward the coil, tending to cancel the next outbound wave (where are you, Lincoln Walsh?). The PVC dome is stiffer yet lighter than the cone and moves the effect of reflections to much higher frequencies, where it can be compensated for by resonating the driver.

Souther showed the electromechanical equivalent circuit for the HV-1 and described how each of the elements corresponded to mechanical parameters that could be adjusted to vary the response of the headphones. The amount of trapped air in the cavity in the rear of the dome diaphragm is made small by inserting a domed-shaped plastic piece. This equalizes the impedance seen in the front and rear of the diaphragm and creates a cavity resonance that maintains high-end response. The resonance is damped by foam in the rear of the earcup and by judicious placement of slots in the front cover. Proper slot geometry is a result of cut-and-try procedures which select energy for transmission to the ear from different portions of the diaphragm to give a smooth high-frequency response.

Velocity headphones are usually not capable of extremely low bass reproduction because of the tremendous diaphragm excursions required, a limitation shared, in principle, with speakers, which also are velocity transducers. The pressure headphone overcomes this limitation by sealing the diaphragm in the ear cavity so that acoustic pressures can be maintained to near zero frequency. Although Koss makes pressure-type headphones with dynamic drivers such as the PRO-4AA, the top of the line is the electrostatic ESP-9.

The ESP-9 uses a 1/2-mil polyurethane film for the diaphragm. Weighing less than the surrounding few millimeters of air, it produces excellent transient response. The air layer also damps the diaphragm resonance at 1.5 kHz. Two perforated stator plates on either side of the diaphragm drive it push-pull to cancel even-order harmonic distortion. A self-polarizing circuit drawing a small amount of power from the audio signal applies about 700 volts to the diaphragm without the need to be tied to a wall outlet (AC power is optional). In each earcup, a transformer boosts the audio voltage by a factor of 60 and contributes significantly to the 19-ounce weight of the unit. The overall smoothness of response, low distortion, and extended frequency range has made the ESP-9 something of a standard by which all other headphones are judged.

In summary, the most common reason for using headphones, confinement of sound to the vicinity of the head, may not be the most important reason for the audiophile. The 40 dB of ambient noise isolation offered by some models can push ambient disturbances down to near the threshold of hearing, leaving 100 dB or more of dynamic range within which to enjoy music. Finding recorded music with this range is another matter. Also, the quiet background is likely to make ticks, pops, and hiss more disturbing. This quiet background, coupled with the lack of masking effects generated by room reflections, allows more musical detail to come through than with the common speaker setup. Hall ambience can be recreated more accurately with headphones but true ambience is recovered only from binaural recordings, of which very few exist today.

<u>The Koss Electrostatic Speaker</u>. Many felt the highlight of the meeting was Souther's description of the new Koss full-range electrostatic speaker system. Interest in this type of speaker started about eight years ago when Koss took over the AcousTec line of Products including the AcousTec X electrostatic speaker. Similar to the KLH-9, the AcousTec X had a 12-square-foot bass radiator that crossed over to a tweeter panel at about 1200 Hz. With only 80 mils between the stator plates, the peak excursion of the diaphragm was limited and bass began to roll off at about 150 Hz. Because of the bass and problems encountered in production and reliability, the speaker was retired, but not before the seed was planted in Souther's mind for a completely new and improved design.

Full-scale efforts on the Koss Model One electrostatic began about two years ago and have culminated in a rather complex four-way system having very impressive specs. Standing 4 feet high, 2.6 feet wide, and 11 inches deep, the claimed frequency response is flat ±2 dB from 30 Hz to 20 kHz. The Model One is capable of generating a steady-state pink noise SPL of 105 dB with a 50-watt amplifier, with 14 dB more headroom for transients. With efficiency at about 1% (the AR-3A is 0.5%), a 75- to 100-watt amplifier is recommended. Amplifier stability problems with this speaker should be almost nonexistent; the input impedance is pegged at 4.0 ± 0.2 ohms from 20 to 20,000 Hz. Finally, the design incorporates features that are said to minimize or eliminate sensitivity to dust and dirt pickup.

All of this sounds like an audiophile's fantasy. After all, how is it possible to overcome all of the limitations that have plagued electrostatics? Here's how:

One of the greatest limitations of electrostatics has been their inability to produce good low bass, set by the diaphragm radiating area and maximum excursion. The Koss Model One is able to achieve much larger diaphragm excursions than previous designs (while keeping the corona-producing bias voltage to a low 7.5 kV) by stacking three (0.5-mil polyethylene) diaphragms in a sandwich structure between four perforated steel stator plates. Each diaphragm pushes on the others through the intervening air layers, the forces adding to produce the needed excursion.

The entire speaker incorporates five panels, bass (three diaphragms), midrange (two diaphragms), two treble panels, and one tweeter, with crossovers at 250, 1400, and 8000 Hz. Each frequency range is served by a separate stepup transformer (two for the bass) designed to roll off at 12 dB/octave at each end of its operating range, thus serving as a built-in crossover. Connecting these transformers together with a resistive network results in the constant, low input impedance so dear to today's solid-state amps.

Grills have been treated with a conductive compound and grounded to keep electrostatic fields from reaching out into the room to snag unwary dust particles. Although Souther indicated no dust problems have yet been encountered, this failed to alleviate some latent skepticism in those of the audience who have had experience with the KLH-9.

Taking a cue from the ESP-9 headphone, Koss uses a self-powering circuit for the polarizing supply, which not only eliminates plug-in-the-wall inconvenience but also cures the annoying fade-away characteristic during soft passages, when little audio charging power is present. When the input to the speaker terminals drops below 1 volt, a (rechargeable) battery is automatically switched into the circuit to maintain the polarizing voltage. This will sustain the voltage for 4 minutes before it is automatically switched off, to allow time for changing records, etc. During times when power is being derived from the audio signal, the battery disconnects and recharges.

The Koss Model One should be available for listening in the Boston area within 3 months. It will carry a price tag of $745 each, or, if you prefer unit pricing, $7.45 per pound. If the Model One jeopardizes your bank account (or your back), scaled down three-way and two-way versions are planned for later introduction.—John Schlafer

Son of a Witch-Glitch Switch

Dan Shanefield

I've been granted more space to give a fuller description of the LAB (Live versus A versus B) test. But before proceeding, I should mention that I'm not an expert in this field, and I don't have the best available equipment to work with. There might be plenty of room for improvement in the test methods described below, and I hope some of the readers will contribute suggestions.

My procedure involves the following steps:

1. Using a graphic equalizer, achieve a flat response (frequency versus loudness) for each component being compared (A and B).

2. Then, where the sound within one frequency band is slightly distorted or has poor transient response, decrease its volume until the optimum realism is obtained. By optimum, I mean the best compromise between response flatness and a sort of "mellowness." For example, listen to a cheap speaker in a carefully equalized system. Then try cutting the 7-kHz response by about 4 dB. I'll bet it sounds more realistic. The higher order harmonics, which sound particularly unpleasant,[1,2] might be expected to occur often in the 7-kHz region. (I'm not sure this is the explanation, but cutting this band often does improve the sound.) This effect might occur with good components, too, but just not as much.

For the LAB comparison, a live performance (L) in my listening room is tape recorded and then played back through component A, and the live versus recorded sounds are quickly compared. First, the system including A is made very flat (more about that later). Then, if necessary, the graphic equalizer is adjusted to be non-flat until the recording sounds as much like L as possible. After that, the taped performance is played back through B, and is quickly compared to L, with various optimizing adjustments. Then I judge which optimized component, A or B, allows the closest approximation to L. In most cases this is surprisingly reproducible, from person to person and from month to month. The reproducibility arises from the fact that there is something very specific for quick comparison, namely L.

3. The next step is to play a recording of a concert hall performance through A and through B, optimizing further until I hear the best approximation to my memory of the concert hall (C). We could call that the CAB test. The reproducibility is not so good.

4. Finally, each component can be purposely maladjusted from the point of maximum realism to the point of maximum pleasure. This is not necessarily the same as mellowness, for those of us who love breathtaking 40-Hz bass, "presence" at 4 kHz, and sharp attack at 12 kHz. I must admit that I find myself turning up the gain at these frequencies. ("Yuuuch," says the purist. But I'll bet most of you do equally sinful things when nobody else is around!)

Here I find that person-to-person agreement breaks down completely regarding which component is "best." It is very important to recognize this, because it might creep into the tests of those people who don't bother to compare A and B to L.

Let's look at some of these steps again. We can dismiss step 4 from further consideration, because it's strictly a matter of personal taste. Working back to step 3, the CAB test, I suggest the following sneaky trick. Tape record a bugle or trumpet in your listening room. Do it twice, with a long pause in between. During the second recording, decrease the volume of the frequency band stretching from 5 kHz to 12 kHz by about 6 dB. Now play back the tape for a friend, at least a month after he or she has heard a live horn performance. Play back the first recording, the flat one, through component A. Conspicuously switch to component B for the second recording, which has secretly been mellowed. Ask your friend, "Which component, A or B, gives the most realistic approximation to your memory of a horn concert?" The answer will usually be, "Of course it's B. The other one was terribly distorted."

What I am trying to establish here is that our memories for sounds are not very good. We tend to forget that real violins are scratchy and real horns go "blaaaat." If you're pretty sure your friend won't punch you, follow through with this: repeat the playback, but blow the bugle right after each recording, during the pauses. That'll show 'em whose system is terribly distorted!

This implies that the components that some reviewers call "harsh" and "glassy" might really be the accurate ones, and the components that sound good might really be a little bit weak in the treble because of capacitive reactance relationships, etc. I'm suggesting all this as a possible explanation for that basic schism between reviewers. But I can't prove that everybody's memory is poor, or that everybody has an aversion to trumpet treble. It's going to take much more experiment to settle these arguments.

Actually, I would love to be able to do some reproducible experiments regarding CAB comparisons, since I usually listen to disc recordings of concerts, and not to tapes of bugles in living rooms. (Note that the LAB test can't directly test discs or phono cartridges.) If we could first do something reproducible with steps 1 and 2, then maybe we would get a good running start on a truer path for future plunges into the darkness of step 3. But I can't rigorously prove that we should ever agree on judgments made about concert hall sounds heard in the living room. (Darkness is darkness is darkness.) It's just a matter of hope.

Back to the relatively firm ground of steps 1 and 2. Here it is in more detail. First, you have to set up a pseudo-performer. Place an SPL meter at ear height, where you usually sit in your listening room (see Fig. 1). Using a pink noise record or a pink noise generator, play several frequency bands of pink noise through the system, but use only one speaker, and place it where the live performer will be standing later. (Alternatively, you can play "white" interstation FM noise through a graphic equalizer, one band at a time.) Adjust the volume of the 1000- or 2000-Hz band until the SPL meter reads 80 dB. For the other frequencies, write down the SPL readings for each band, through the range of 40 Hz to 14 kHz or wider. This is the pseudo-performer: a calibrated source. Its overall system response (through the SPL meter) does not have to be flat, as long as it is reproducibly measurable for a few hours-

Now set up a stereo tape recorder, with two good mikes. (I use the AKG 202E-1, but of course there are many other good ones.) Place the mikes quite close to the pseudo-performer's speaker (about 2 feet), since mikes seem to have a sort of Fletcher-Munson effect whereby their responses are very sensitive to loudness and distance. Introduce some artificial asymmetry in distance or angle.

To minimize microphone-floor interactions, suspend the mikes over heavily stuffed chairs or inside open boxes made of sound absorbing tile (I use Sears 64H 85852 arranged to make a triangular prism). I don't know why a mike should be worse than an ear when it comes to multi-

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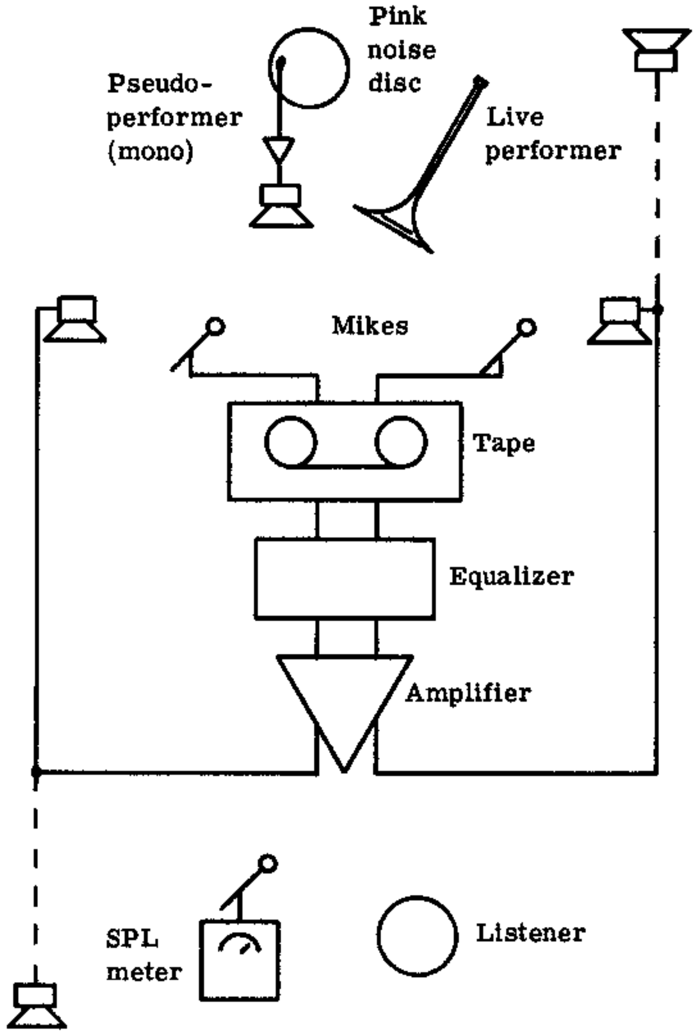

Fig. 1. Configuration for LAB test

path interference with floor reflections, but it seems to be a lot more easily disturbed, [3] especially a cardiod, with its complex little air chambers and vents.

Tape record the pseudo-performer's frequency sweep. Now you can dismantle the pseudo-performer.

Play back the tape in stereo, through two speakers. I like to mount the speakers where the mikes were. (I think that ought to be the best location, but I'm not sure that it is.) Monitor the tape playback with the SPL meter. Adjust the graphic equalizer until you get the same readings as you got with the pseudo-performer directly.

At this point you have a room-mike-recorder/playback-speaker-room link which is flat. But one potential criticism is that the graphic equalizer's bands are too broad for all the narrow peaks and valleys that are present in the overall response curve. However, this makes surprisingly little difference, as long as the worst groups of peaks are squashed.

Theoretically, another potential problem is the fact that while the loudness of room resonances might be compensatable, the transient responses of any resonating system such as the room are almost impossible to damp completely. Amazingly, this also doesn't seem to be an audible problem if you squash those biggest peaks. Because the room enters the system twice, you have to attenuate deeply, and this seems to deaden the bass overhang. In fact, I found that when comparing to live music, realism is strongly enhanced by adding some Bose-like reflection as in Fig. 1 (dashed line) and also by adding some quad-like sound behind the listener. (Who says phase distortion is bad? From this experiment it actually seems to be good, at least when it is added to an ordinary system.) However, rather than the Bose 901's 8 :1 ratio of reflected to direct sound, I prefer about 1 :2 in this situation.

Certainly you already know that not all components can be "flattened." (By the way, I'm assuming that you're not a plug-in freak and that you're willing to match capacitance requirements, etc., to avoid some easily corrected problems. [4]) Even so, many systems will only give you noise when the 21-Hz or 19,999-Hz responses are boosted. My rules for dealing with this dictate that if component A is unflattenable, like an electrostatic speaker, then component B should be given two chances to compete: (1) flat, and (2) purposely unflattened to match component A's response curve. Try turning a Bose 901 around to face you directly and de-equalize it to match the curve of an electrostatic. Being pretty coherent (with full range cone design and no crossovers), it sounds so much like an electrostatic that you need mighty "fine" (meaning close to 24 carat in the gold trade) ears to tell the difference.

And now for the live performance. It is easiest to begin with a piano- Have someone play C^2 loud enough to make the SPL meter read 80 dB, and then start tape recording it. Leave long pauses. Then play back the tape and ask the pianist to play during the pauses, imitating the loudness. Stationing yourself where the SPL meter was, compare L versus A, then L versus B. Which component, A or B, allows the most realistic playback ?

This test can be repeated for low and high notes. Semifinalist components get the bugle, violin, and bell, and finalists get the human voice. If necessary, optimize A and B independently for the best realism.

Don't be surprised to find that medium-quality components (less than $500 each) can sound very much like live performers in the same room. After all, Harry Olson reported fooling people with L versus A tests using very old technology (1947). [5] He was doing it all out-of-doors, which eliminates most resonances. But Edgar Villchur and others did it in concert halls from 1955 to 1964. [6] And sound reinforcement during concerts fools audiences daily!

Suppose you repeat all this, and it ends up being easily reproducible ("true-for-you-too"). Then it remains to be seen whether the over-$ 500 stuff is really better in a rigorous LAB test. There often appears to be a dB-like relationship between quality and price, whereby doubling the price is necessary to produce the minimum audible improvement. Is there a plateau of $500? Julian Hirsch[8] (again) insists that the ESS Model 200 power amp is audibly indistinguishable from much more powerful ones. On the other hand, pages 52 and 54 of Ref. 2 say the same amp is overly "bright" and "edgy." (Such words are rarely used in the shiny mags like Audio, by the way.)

If the super-expensive stuff turns out definitely to be more realistic, then we will understand why it ought to win in a CAB test also. But what if it achieves only a draw in the LAB test, and yet it consistently wins in the CAB test? Then there will be something weird going on. Maybe we must do something to change the sound of our listening rooms in order to mimic the memory of the concert hall sounds. For speakers to do that—OK. But for power amplifiers to do that ???

APPENDIX

I think I should warn you about the witch-glitch switch described in the November Speaker. The diagram was meant to communicate the basic idea, but in most cases you'll find it useless unless you run the current through it backwards (as in testing power amps, etc.). Also, my local dealer-repairman claims that, while inveterate A-B-ers have many power amp failures, the average listener does not. Maybe we are guilty of open-circuiting the inputs too much, even if only momentarily. Static electricity, or RF, or internal leakage voltages might be blasting the amps. The circuit of Fig. 2 allows you to put a low resistance across the input before and during A-B switching, and then take the resistance off, but only after the preamp gets connected in.

Another advantage is that you don't have both power amps' inputs loading the preamp while you're listening to only one power amp. In some cases this could be very important. Because of RC mismatch, one amp could spoil the other one's sound if their inputs remain hooked together. It's probably easier to avoid this (using Fig. 2) than to test for it.

To load the output of the power amps under test, put a 2- or 5-watt resistor of 50 to 1000 ohms across each pair of output terminals. These resistors can be left in place during the test and will not generally affect performance.

When A-B-ing preamps, you can use the circuit of Fig. 3 to avoid open-circuiting the power amp while changing the preamps.

If you don't have a Russound switch box, or have one so wired into your system that you don't want to remove it, you can perform power amplifier comparisons using switches like the one shown in Fig. 4. Use one double-pole, double-throw switch for each channel of each amp under test. Two 4PDT switches are convenient for A-B-ing; simply throw the bat handles in opposite directions simultaneously. Add connectors to suit your needs and mount it all in a metal or plastic box—the application is noncritical.

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2. Bascom H. King, Audio, May 1974, page 57.
3. David L. Josephson, Audio, July 1974, page 34.
4. Julian Hirsch, Stereo Review, January 1974, page 61. This is a good article on matching (which does not necessarily mean "making equal"). I must admit that audible mismatches are not exactly common occurrences, but they still might be a decisive factor in some cases.
5. Edgar Villchur, Audio, November 1964, page 6. This is a letter acknowledging earlier work now out of print.
6. Edgar Villchur, Audio, October 1964, page 34.
7. See recent Acoustic Research Co. advertisements, such as in Audio, November 1974, page 83.
8. Julian Hirsch, Popular Electronics, December 1974, page 73.

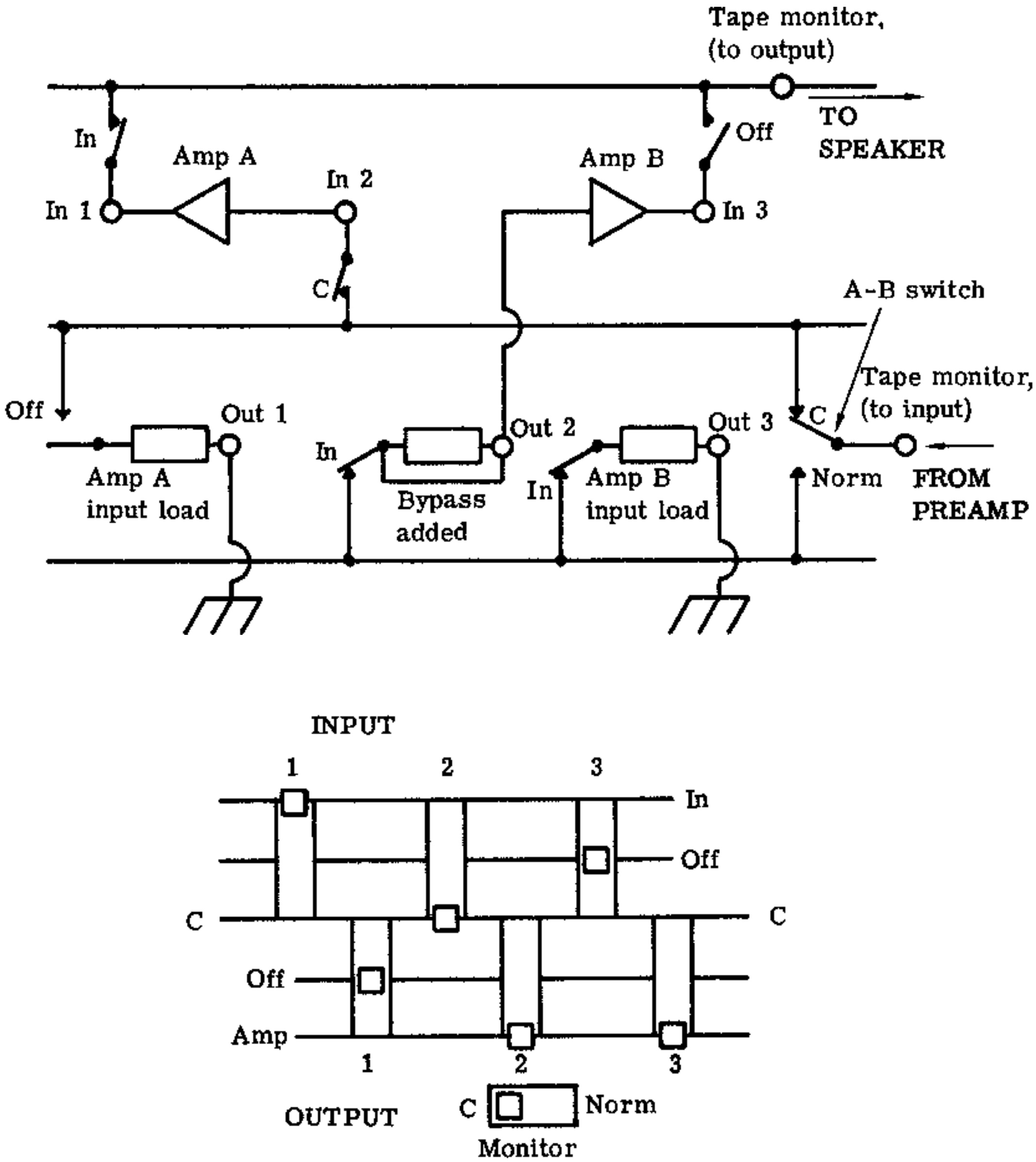


Fig. 2. A-B switch for comparing power amplifiers. (Please note that this is not a W/G switch.) Open circuits at inputs can be prevented with this arrangement. The switches are conveniently obtainable in shielded, silent form by using a Russound TMS-1W or Sony SB-300 tape monitor dubbing box. (The latter is available through Lafayette Radio.) The built-in resistors are each 3.3K, but I suggest bypassing one of them. The lower drawing shows the front panel of a TMS-1 switch box with slide switches in positions correlating to those in the schematic. Not that neither "input 2" nor "output 2" switches are moved. "Input 1" and "input 3" connect and disconnect amp outputs from speakers; "output 1" and "output 3" Connect and disconnect input-load resistors to amps under test.

a

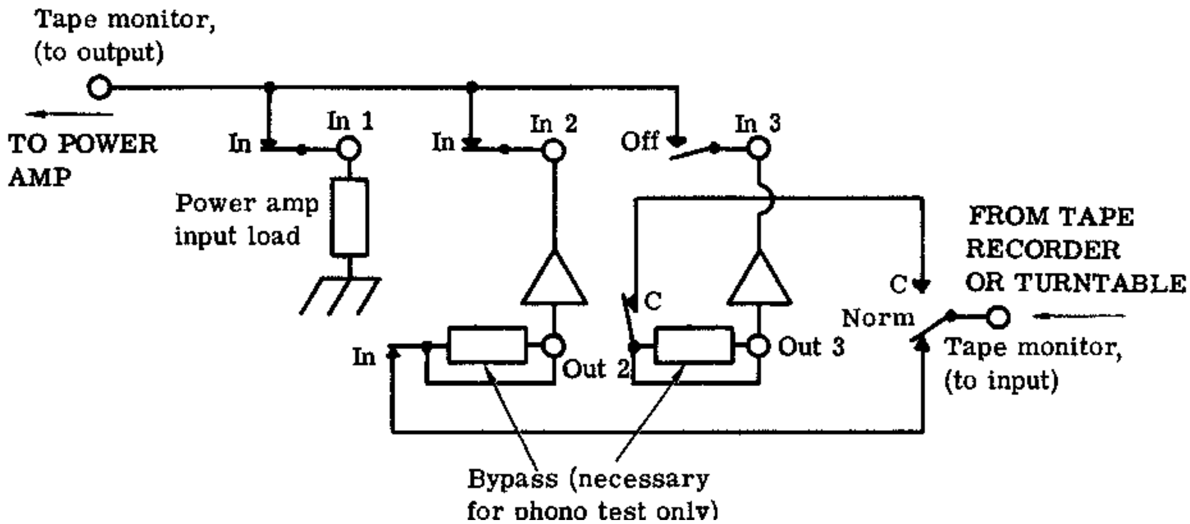


Fig. 3. A-B switch for comparing preamps. The grounded resistor at "input 1" is not disconnected, as it acts as an input load resistor for the power amp. "Output 2" and "output 3" switches need not be moved during A-B comparison.

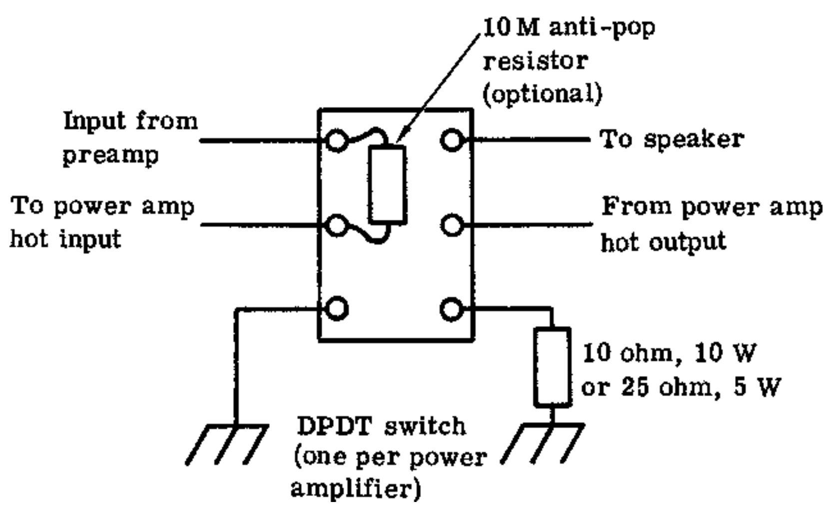


Fig. 4. Alternative switching arrangement

A Case for the 814

Alan Southwick

A lot of enthusiasm has recently been generated about a small electret condenser microphone made by Thermo Electron Corporation of Waltham, Massachusetts (see "A Quasi-Complementary Discussion of Microphones," by Peter Mitchell in the February 1975 BAS Speaker). Originally designed for an ultrasonic alarm system, this capsule, according to recent studies by the designer, Dr. Freeman Fraim, has a flat frequency response to 30 kHz for random directional incidence angles between 60 and 90 degrees off-axis. Also, the capsule has withstood the rigors of military environmental testing and is impervious to a reasonably wide range of temperature and humidity conditions. What this all adds up to is that for about $45, including capsule, you can have a first-class omnidirectional microphone in a rugged case wired for simplex powering from your microphone preamp.

The first step is to decide which capsule to choose for the particular usage you have in mind. The 814 is capable of handling about 120 dB SPL at 3% THD. So if all you plan to record is dual-microphone, true stereo at 20-plus feet from the nearest instrument, you can chop $5 off the previous figure (quoted for the 814C). If, however, you might be using the mike to "spot" an instrument or performer, spend the extra amount for the 814C, which can withstand rock levels or a jet engine at 145 dB SPL for 3% THD, allowing preservation of fleeting musical peak information.

The microphone requires some dc drive voltage for the FET impedance converter. This can be a 9-volt transistor radio battery mounted near the capsule, or if you're already equipped with a few feet of two-conductor shielded microphone cable with Canon XLR type connectors, simply add compatible simplex powering to your microphone preamplifier system.

Simplex (or "phantom") powering is a very simple technique for getting supply voltages to transistorized condenser microphones without resorting to multiconductor cable. A schematic diagram of the system powering technique is shown in Fig 1. With a balanced microphone line, the actual signal is floating above ground and the input is quite similar to a differential input with

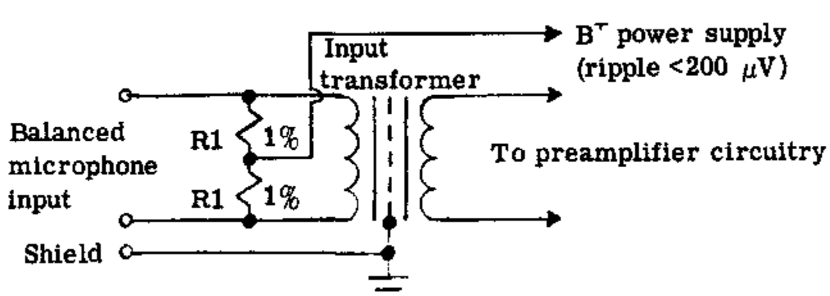


Fig. 1. Simplex or phantom power system

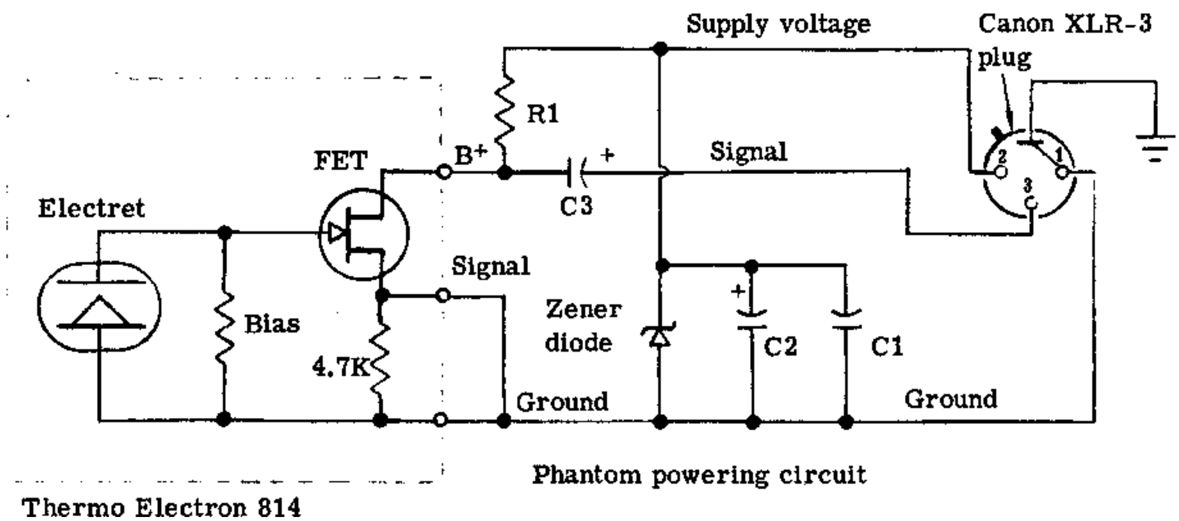
high common-mode rejection, i.e., any voltage common to both sides of the signal line will null at the input so induced hum and other extraneous voltages will not interfere with the audio signal. The 1% resistors (R1) provide a balanced dc feed to the input side of the transformer and provide current limiting so as not to damage either microphone or supply voltage. The ground shield provides the return path for the dc power to complete the circuit when the microphone is connected.

Since the worst-case current drain of simplex mikes is no more than 3 mA, the value of R1 may be determined by application of Ohm's law:

$$R1 = \frac{\text{Supply voltage}}{2 \times 0.003 \text{ amp}}$$

The factor of 1/2 is introduced because the dc resistors are paralleled to the transformer.

For instance, if you plan to use the Advent MPR-1 preamp, which requires +18 volts, R1 = 18/0.006 or 3 kilohms. Even higher supply voltages, up to 50 volts, are useful if you also want to phantom power AKG or Neumann mikes. Voltage regulation in the microphone sleeve takes care of any excess supply voltage (see Fig. 2).



Parts List

Thermo Electron 814 or 814C capsule (see text)
Switchcraft S3FM, machined (see text) or 3/4-in. o.d. aluminum tube, 5 in. long
Zener diode (see text)
C1-0.001 µF, 250 V disc
C2, C3-100 µF, 25 Vdc electrolytic
3 phono-cartridge-type lugs
R1-4.7 KΩ,, 1/2W or 1/4W
Misc.: Insulating sleeving, 3-in. hookup wire, 3/8-in. o.d. x 1/4-in. long plastic sleeve, wax paper, five-minute epoxy glue, AKG wind screen W-3

Fig. 2. Simplex wiring scheme and parts list

To feed supply voltage to the 814 capsule, the drain resistor in the capsule is bypassed and inserted (external to the capsule) in the FET source circuit. Signal is then derived across this resistor and through a bypass capacitor to prevent shifting of the FET's operating point. A 1-watt zener diode chosen to limit maximum voltage to the capsule (18 volts for the 814, 24 volts for the 814C) is tied to the supply lead along with a large dc filter capacitor and a bypass capacitor to reduce radio frequency interference. Fig. 2 is a schematic diagram and parts list for the circuit and Fig. 3 shows the parts layout.

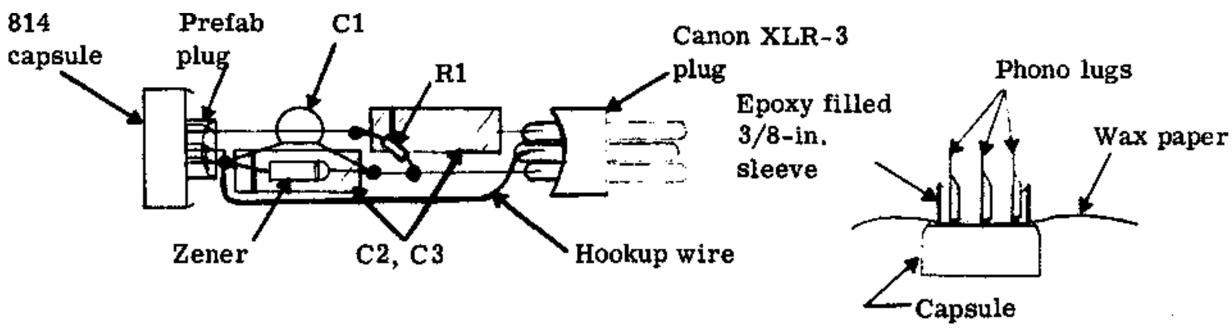


Fig. 3. Parts layout and fabricated capsule plug detail

The capsule plug, not standard, is fabricated by using three phono cartridge lugs, some 3/8-inch sleeving, a piece of wax paper between the capsule and the pressed-on phono lugs, and 5-minute epoxy cement. Using phono lugs avoids damage to the capsule from excess soldering heat; be sure the lugs fit very snugly on the capsule pins before gluing them into a plug (see Fig. 3). After about 10 minutes setting time for the epoxy, the plug should be "coaxed" gently away from the capsule with a knife blade. Once free, peel away the wax paper from the plug and remove any excess paper and epoxy from the capsule pins with emery-paper, file, or judicious scraping with a knife.

If you wish to mount the mike in the Switchcraft S3FM sleeve (see February BAS Speaker meeting summary) the jack end of the sleeve requires some machining modifications. I had two pieces done professionally for $10 to the dimensions specified in Fig. 4. A larger 3/4-inch tube can also be used as a sleeve gently press-fitted with polyfoam inserts.

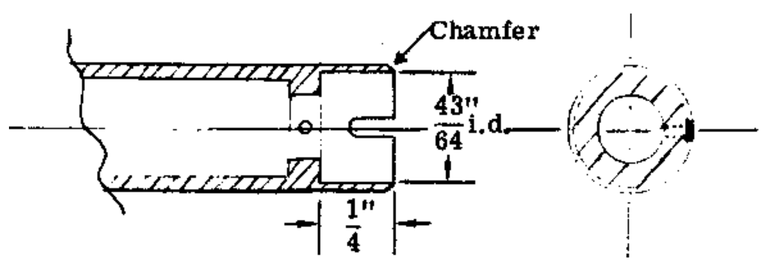


Fig. 4. Machining modifications to jack end of Switchcraft S3FM

Solder the plugs to the assembly as shown in Fig. 3, making sure you get the right connections to the right pins (see the 814 data sheet and Fig. 2) and slide the S3FM paper insulator over the whole assembly. Keep the leads very short. Now check the fit by sliding the assembly gently into the S3FM sleeve from the Canon plug end. If all goes well, the fabricated capsule plug should be flush with the top of the inside shoulder collet of the S3FM sleeve. If it isn't, trim the leads at the Canon plug and resolder until it is just flush.

At this stage, plug the capsule into the sleeved assembly and try the mike out to make sure everything works. The large capacitors may short to the Canon plug. Should raspy noises occur along with audio, the phono lugs are probably not making good contact with the capsule pins. You may be able to adequately crimp the phono lugs using a jeweler's screwdriver—if not, unplug the capsule, unsolder the fabricated plug, and make a new one.

If the mike works, unplug the capsule, use a little 5-minute epoxy around the machined end of the sleeve, reinsert the capsule, and let the epoxy glue set.

Thanks to engineer Alan Woodard of Thermo Electron, I was fortunate enough to run some tests on the capsules, cased microphone, and a comparison professional omni mike (see Fig. 5). The anechoic chambers utilized in the tests are limited to a low frequency of 200 Hz, but the case dimensions will have little or no effect on the low-end response. The top end rises as predicted on-axis and some chamber peaks and dips are noticeable above 15 kHz. The mid-frequency rise of the Sony C-37 is attributable to the interaction of the relatively large case of this particular mike and the disruption of the sound field caused in the small anechoic chamber.

When using the 814 for general pickup, try placing it so the capsule is between 60 and 90 degrees off-axis from the sound source, aimed upward for the smoothest top end. On-axis placement will effectively create a much brighter sound including pickup of increased sibilants and rustling noises. Some placement experimentation will help in selecting the most suitable pickup for the particular results desired.

Although some problems were encountered in fabrication of the initial assemblies, if done carefully, microphone assembly should take a couple of evening's effort at most once the sleeve is machined. Incidentally, the AKG-C451E windscreen, W-3, is a good fit for this 814 mike and adds extra protection to the capsule from dust and debris, albeit at an additional $6.

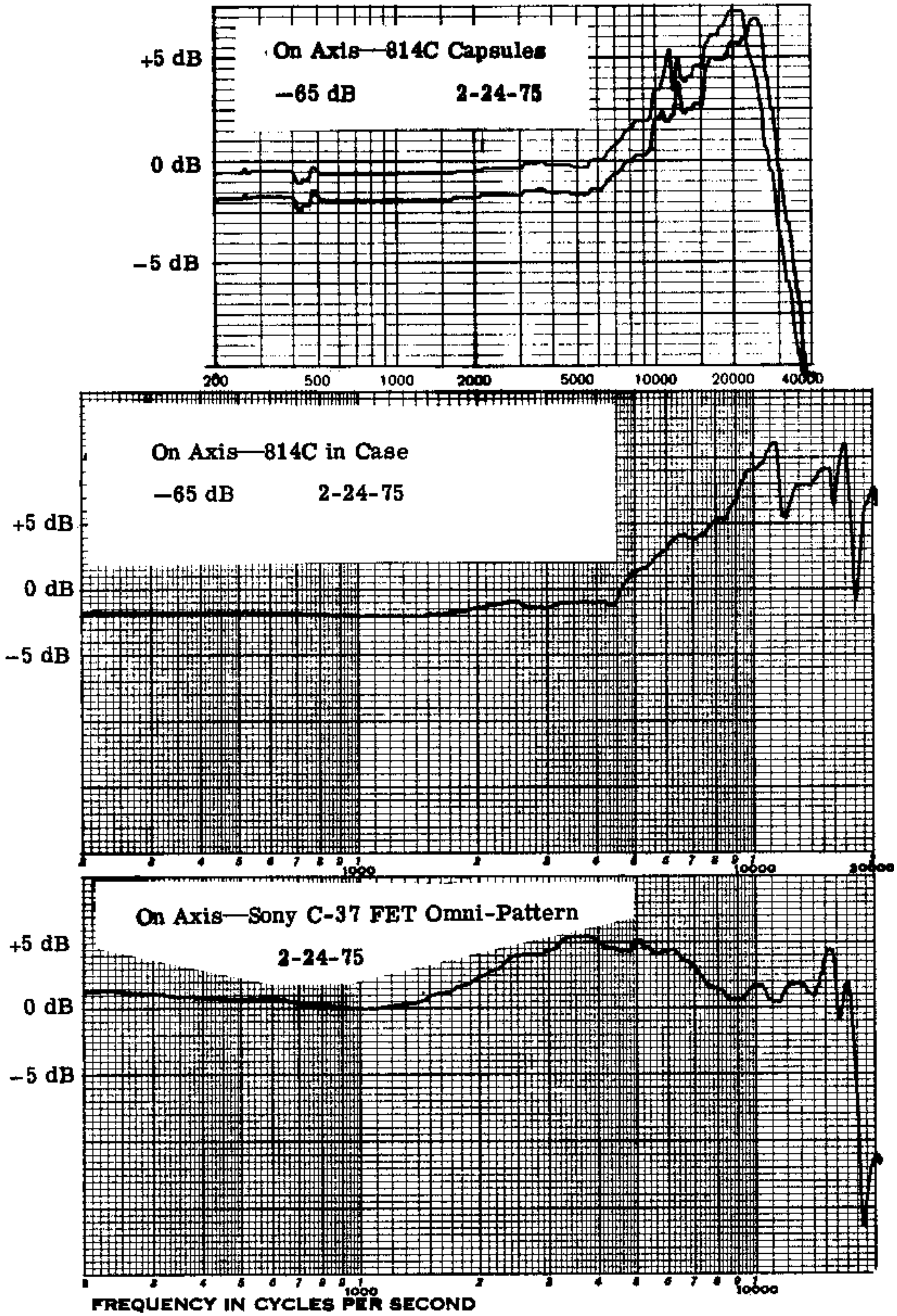


Fig. 5. Mike frequency response