THE B.A.S. SPEAKER

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In This Issue

Would you believe another article about the 814 microphone ? This month's Speaker includes what is probably the last of the 814 articles, and one of the best. This should please everyone. Those who care nothing for live recording will be happy to hear that we have just about exhausted the possibilities for this wonderful little capsule; those who are more positively inclined will be happy to have a design for a line-level 814 microphone.

Designed and constructed by BAS members Rene Jaeger and Jamey Reilly, the unit combines the excellent audio performance we have come to expect of the 814 capsule with "at-the-mike" line amplification that enables users to run fairly long cables—up to 200 feet with far less chance of gross hum or RFI pickup than with ordinary unbalanced lines, and provides the capability of directly interfacing with other line-level devices like noise-reduction units or tape recorders. Performance is good: the unit should overload at about 132 dB SPL, and because some of the capsule's internal parts are taken into account in this design, the residual noise should be a bit lower than Thermo Electron specifies (i.e., about 22 dB versus an A-weighted 24 dB specified). That means a microphone with a weighted signal-to-noise ratio capability (read "usable dynamic range") of about 110 dB, and peak outputs of about '7 volts. More than this you should not need.

Car Pool to GTE

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

Want Ads

For Sale or Swap: Willing to swap records (on a permanent basis), any category considered, prefer stereo. List available by title, label, number. Rene Jaeger, 899-8090 days.

For Sale: Ortophon SL-15E Mk. II cartridge without transformer. Never used, in original box, with warranty. $65. Rene Jaeger, 899-8090 days.

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For Sale: Revox A77, Mk. II quarter-track tape deck. Includes remote control, stainless steel faceplate, dust cover, NAB hub adaptors, owner and service manuals, Ampex test tape (used twice), five 10.5-inch reels of BASF LP-35LH, ten 7-inch reels of BASF LP-35LH. Tape used one time and bulk-erased. Also: Advent 100A Dolby noise-reduction unit with case, Advent mike preamp, owner and service manuals, cassette and reel-type Dolby alignment tapes, all cables. Package price of $700 includes UPS shipping in manufacturers' cartons. John Tooley, RD 2 Box 120E, Milton, Delaware 19968; (302) 684-3442.

For Sale: IMF ALS 40A speakers, McIntosh MC-2105, McIntosh C-28. With all packing and warranties. Almost new. Mike Lulejian, Atlanta (c/o Box 7, Boston).

For Sale: Crown IC-150 preamp, pair of Klipschorn corner speakers (red mahogany, model KDBR, two years old), Pioneer TX-1000 tuner, Crown DC-300 power amplifier, Dual 701 turntable with Micro Acoustics cartridge. Overall, $3250 worth of equipment. Will sell separately or together; will talk price. David Kunz, P.O. Box 304, Marlboro, Mass. 01752; (617) 481-1219.

Wanted: TTS-3000A turntable with base. Dean Slindee, P.O. Box 55, Lansing, Iowa 52151.

Letter

I am interested in corresponding with BAS members who have an interest in FM reception—antennae and general techniques. I'm planning to work with PLL demodulators and to try various PLL tricks with my Heathkit AJ-29. In addition, are any BAS members working on distortion analyzers and/or audio spectrum analyzers? Exotic, I know, but I'm interested in knowing what goes on inside those circuits in a quantitative manner. I've been working on modifications for a Heathkit IM-38 but haven't gotten very far yet. — Damon Hill, Atlanta, Georgia

Ranada Redux

Since the perpetuation of errors reflects badly on both their author and their publisher, I herewith submit corrective commentary on the writeup of my presentation to the BAS. Due to the time limitations imposed, I was not able to give complete or precise "labeling" of the musical examples used.

1) Stravinsky: Le Sacre du Printemps (Danse Sacrale; rehearsal nos. 174-186). Stravinsky's recording uses his "Neo-Classical" rescoring of this dance. The most obvious difference is that the revision does not have a gong during this passage. Stravinsky's is the only recording I know of which uses the revised version.

2) Debussy: La Mer (third movement, 4 bars after No. 59). The horn calls heard in the Froment and Ansermet recordings (among others) were removed in Debussy's later revisions of the work. These particular motives are not (as the writeup would have it) in the Boulez recording.

3) Brahms: Symphony 1 (fourth movement, Letter N). The "traditional" slowdown heard in Walter's recording probably arose long after Brahms' death. Brahms was always careful in such obvious matters and did not write "ritardando" at the place in question. Anyway, a ritard spoils the melodic and rhythmic relationship between this passage and the preceding one.

4) I would disagree with the statement that we would miss "a great deal of musical satisfaction" in works suffering from meaningless "traditions." If I said this, I retract it since we can still be dissatisfied with performances which remove the encrusted traditions. What the audiophile, and the record buyer in general, must realize is that the interpretive liberties taken with some pieces give the non-music-reading listener a distorted impression of what the composer had in mind when he took the trouble to write down how something should be played.

5) Beethoven: Symphony No. 3: (first movement—not finale—9 bars after V, m. 655). Here the horns and winds (minus bassoons) state the theme in the tonic and then the winds (plus bassoons) without horns or trumpets state the theme in the dominant. Beethoven removed the trumpets and horns since the valveless instruments of his day could not play the pitches required. To many people this section, with its mid-19th century additions of trumpets and horns, is the climax of the movement. To me, however, the additions destroy the timbral continuity of the piece. The alterations make this section the only place in the whole symphony where the brass play these pitches. The Monteux recording (not the Leibowitz as reported) is the only one I have heard which plays this section more or less as written.

6) When I first heard the Leibowitz recordings of the Beethoven symphonies I was attracted by his attempts to at least start at Beethoven's own (rather quick) tempo markings. On repeated listening to this set, however, I find it not as noteworthy as I first thought. First of all, he too indulges in the many totally unnecessary modifications of orchestration which have by now become "traditional." Secondly, Leibowitz, though starting at Beethoven's tempi for many movements, slows down perceptably by the middle of those movements to the normal, late-romantic tempi. Both long-term harmonic effects and the rhythmic continuity of the performances suffer from this style of interpretation. There are some revelations for the listener, despite my reservations (particularly the first movement of the 9th symphony). I would not recommend this or any other "complete" set for someone wanting only one cycle of Beethoven symphonies, however.

I must compliment Peter Mitchell for a generally accurate and fair review, especially his summary of my main points. My hasty and rushed delivery must not have helped him sort out everything.

— David Ranada

Peabody-Mason Concerts

Stuart Isveck advises us that the concerts given each year by the Peabody-Mason Music Foundation are both "terrific and free," an excellent combination. Getting tickets is a simple process but it requires a little care. One must apply separately for each concert with a request postmarked no earlier than one month prior to the concert in question, and include a stamped self-addressed envelope. Requests are limited to two tickets; children under ten are not admitted.

Write to: Peabody-Mason Music, Post Office Box 153, Back Bay Annex, Boston, MA 02117.

Here is this year's schedule:

- November 5: Zagreb Pro-Arte String Quartet
- December 3: Israel Piano Trio
- January 28: Il Sestetto di Bolzano—Music for winds, horn, and piano
- February 11: Maurizio Pollini, piano
- March 24: Berlin Philharmonic Octet

All concerts are on Wednesdays at Harvard's Sanders Theatre and begin at 8:30. For additional information call (617) 262-4848, mornings.

Data on Discos and. . .

By now, many BAS members must have heard the AR demo record pressed for them in Spain by Discos Ensayo. I was very impressed with the fidelity and quality of the pressing of this product, but I had never heard of Ensayo.

I got their address from AR and wrote to them for a catalog; they responded with one dated October 1973 that had most of the musical titles in Spanish. I took a chance and ordered several of their recordings just to see if the quality of the AR demo record would be matched.

In about four weeks I had my answer: The discs that arrived were all as good as the demo except for one earlier album that had a slightly higher background hiss level.

Ensayo's catalog is not extensive. They offer mostly classical works, some jazz, and two or three recordings with voice. While all the recordings are excellent, I did receive two pressings out of six that were less than perfect. Ensayo also offers a few cassettes.

Perhaps the best news is that Ensayo charges $3.00 per album for these discs; to me they would be a bargain at a much higher price. Unfortunately, shipping costs do inflate the price. Specifically, shipping on six discs was $8.22, raising the average cost per disc to slightly over $4.00-still a bargain to me.

The company's address is: Mr. Antonio Armet, Discos Ensayo, S.A., Zaragoza, 16, 3º, 5.ª, Barcelona - 6, Spain.

I made payment by international money order as obtainable at the Post Office. Packaging seemed a bit flimsy to me (by United States standards), but all the records arrived intact.

— Jerry Johnson, Oklahoma City

...A Line on Lyrita

The Lyrita releases listed in the August compilation of good recordings are available in domestic pressings from the Musical Heritage Society for a song. I have not compared the Society's discs with their British counterparts, but I doubt that there's any difference in sound that's worth the price (and inconvenience). The MHS versions are superb in every respect save one; unfortunately the Society's pressings aren't what they used to be. All of my Bax recordings are a trifle noisy. Not bad, but not Philips either. — Mike Riggs, Boston

BAS Oscillator Feedback

David Roudebush assembled one of the first BAS oscillators delivered and found that in order to prevent clipping with the signal generator in its highest output setting, he had to trim the 39-ohm resistor (a 10% tolerance item). All it took in his case was placement of a 390-ohm resistor in shunt with the original one. If the 39-ohm resistor had been 10% high in value (42.9 ohms) and the 390-ohm resistor exact, the equivalent resistance would have been very close to the specified 39-ohm figure at 38.6 ohms. If you have a similar problem, you might either try a similar shunt resistance or either a 5% or 1% tolerance resistor. — Jim Brinton

Capacitance and Your Phono Cartridge—Revisited

The June 1975 Speaker contained what was meant to be a helpful note on the capacitance requirements of several cartridges and the inherent capacitances of several turntable/arm combinations and a couple of separate arms. Unfortunately, as local member Mike Riggs points out, because of the labelling of the tables, it is possible to misinterpret some of the data. (Worse, there may also be errors in Dynaco's information—the BAS's source for this data.)

Without going into the grisly details, let it be said that the information on the Decca International and SME/Shure tonearms is correct (as supplied by Dynaco . . .), but the figures are said to refer to the capacitance of the arm alone, and do not include that of any interconnecting cable. Apparently . . . .

Thus, the capacitance of the Decca—without hookup cables—is 300 pF, according to Dynaco. If this is accurate, one would be wise indeed to use only the lowest capacitance connecting cable with the Decca in a CD-4 application, or any other for that matter. What is probably more nearly true is that the figure given includes connecting cables supplied with the arm, and since Decca serves markets requiring various cable lengths and connector types (i.e., DIN as well as RCA phono), the number probably is an average for the arm and several different cables; unfortunately we know not which.

We are in better shape with the SME/Shure through the good offices of Delaware member John Tooley who has supplied us with SME Data Sheet no. 13. This sheet lists the capacitances of all the varied cables supplied by SME for use with their arm. Since few if any BAS members are likely to be using their SME/Shure in a system with DIN connectors, we will not reproduce that information here, though it is available to any who want to write us for it.

Part No.	Length, feet	Nominal Capacitance Per Channel, picofarads	Total Capacitance With 3009 Arm, picofarads
L Series: Connects SME Series II and Series II Import arms to phono sockets (male output)			
L2	2	59	74
L4	4	112	127
L5	5	139	154
L6	6	168	183
LCL Series: Connects arm to phono sockets (male output); "ultra-low" capacitance for quadraphonic use			
LC L4	4	60	75
LCL6	6	89	104

If you now are wondering about the credibility of Dynaco's other listings, you are not alone. Even well-intentioned firms like Dynaco often have to use information supplied by people who either fail to understand the inquery or are inexpert enough to cause errors. If any BAS member has data that he can quote with absolute confidence, like the SME data sheet, please send it to us so that we can reproduce it. Especially, send it if it contradicts data we already have published. Meanwhile, thanks to Riggs and Tooley for jarring our complacency.

— Jim Brinton

Optimum Damping for Your Tonearm

Since January 1975, the Speaker has been advocating viscous damping of tonearms for improved reproduction of stereophonic discs, increased warp tolerance, etc. Several members have developed methods of damping various arms, most notably Bob Graham's dashpot approach, also mentioned last January. But until now there has been no way of determining whether optimum damping had, in fact, been achieved. Most of us, therefore, applied damping trying to err on the underdamped side—most of the advantages were obvious despite our lack of test methods, and we realized that overdamping could be as serious a condition as the undamped state, though for different reasons. Now, New Jersey member Dan Shanefield has developed a technique of elemental simplicity which we publish herewith.—Ed.

When a "critically damped" mechanical system is disturbed by a unidirectional impulse, it returns to its equilibrium position at the fastest rate possible without quite overswinging in the opposite direction. And theoretically, the optimum amount of damping is applied to a tonearm that is critically damped.

Here's a method for determining when a tonearm is critically/optimally damped. Feed the left output of the phono preamp to the vertical input of an oscilloscope, preferably a "dc-coupled" scope. Set the scope's horizontal sweep rate for about 4 Hz (equivalent to about 250 milliseconds full-scale). Put a disc on your turntable, but leave the motor off.

Using the fingerlift on the side of the tonearm head, hold the stylus about 1 millimeter (about 0.039 inch) above the record and near its center. Now drop the tonearm onto the disc while watching the scope. The scope trace from an undamped system will probably look something like that in Fig. 1. It may be inverted, but this isn't important—if it bothers you, switch to the other channel.

Increase damping until the curve looks more like the ones in Fig. 2. Curve B shows critical damping, while curve C shows an overdamped condition—the apparent frequency is changed.

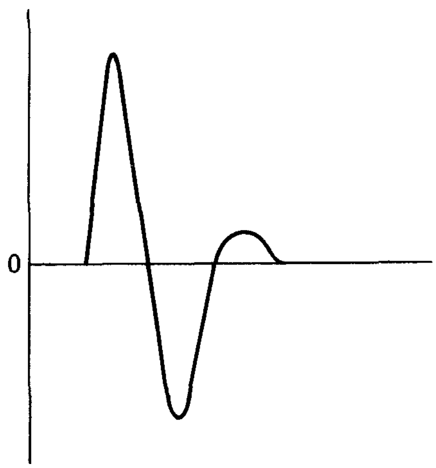


Figure 1

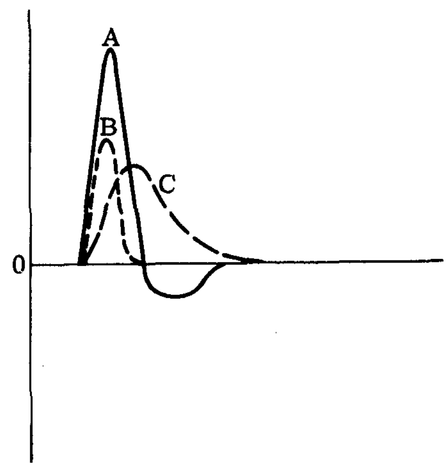


Figure 2

I prefer to stay slightly underdamped (i.e., the curve would lie somewhere between curves A and B of Fig. 2), because it is hard to tell B from C in practice, and extreme overdamping can be as bad as no damping at all.

It was fairly simple for me to adjust the damping in my system as I had mounted a Bob Graham-style (variable area) paddle on my SME 3009/non-detachable-head arm. A 2-square-centimeter horizontal paddle with Dow-Corning 200 (450 cp) fluid provided a type-B curve and made my worst warped records playable even with the highly compliant ADC XLM cartridge.

While you have the scope hooked up. To estimate the resonant frequency of your arm-cartridge system, adjust the damping to give a type A curve, then increase the scope's sweep speed until the tail end of the disturbance barely starts to appear at the beginning of the trace, overlapping the true beginning. Then back off the sweep speed until you see only a clean beginning again. The total time for one positive-going hump plus one negative-going hump (slump?) will indicate the frequency. For example, 50 msec up plus 50 msec down indicates 10 Hz, since frequency (Hz) = 1000/time (msec). A rough calibration of your sweep is possible by viewing 60-Hz line frequency; it should make 16 $^{2}/_{3}$[6]complete cycles in 100 msec.

Also . . . The warp frequencies of the records in your collection also can be estimated if a graphic equalizer is available. Set all frequency bands to maximum cut except for the lowest one, which may be set at neutral. Play a record through this filter, and observe the output on the scope.

The warps will show up with essentially all the music filtered out. The highest amplitudes are generally observed in the range of 2 to 6 Hz.

The stereo setting of the preamp, using one channel's output to the scope, will allow display of both veritical and horizontal warps. But the mono setting discriminates against vertical information, and thus shows only horizontal irregularities. Therefore, rapdily switching between stereo and mono will show that most very low-frequency oscillations are entirely vertical, as the shift to mono obliterates them. — Dan Shanefield

More on Tonearm Damping

Although I have to agree that my damper is ugly, a coat of flat black paint does help a great deal to hide it and I am redesigning it for less mass and a nicer appearance, if that's possible (see BAS Speaker, April 1975).

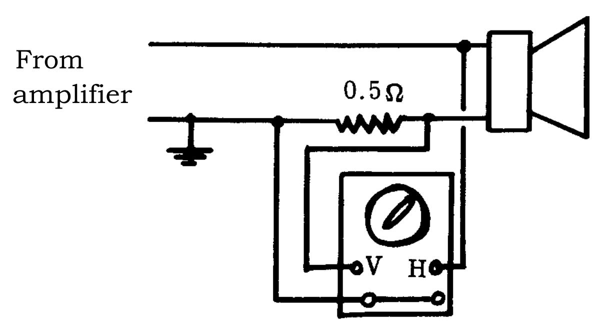
At present I am using a small piston in castor oil, which gives minimal damping—just enough to prevent the continual headshell flutter that occurs without a damper. Woofer-cone flutter is also eliminated, and the bass is tight and well defined, instead of mushy. The improvement is striking on Rick Wakerman's "Six Wives of Henry VIII" (side 2, band 1).

I would compare the audiophile's use of tonearm damping to the "autophile's" use of Koni shock absorbers—the tendency is to adjust the Koni's to an overly stiff setting, which may be fine on a smooth race course but is actually detrimental to handling on uneven pavement; and the ride is terrible. I'm afraid most of us may be doing the same thing and are overdamping our tonearms. — Tom Mashey

[No excuse for that anymore. See above. —Ed.]

Speaker Measurements Mod

A circuit for evaluating loudspeaker current-versus-voltage relationships contributed by Peter Mitchell was shown on p. 15 of the July 1975 Speaker. In the semiconductor industry, this is well known as the "curve-tracer" circuit, and of course, it works fine in loudspeaker work too, providing that one has an oscilloscope that can operate with a floating (ungrounded) chassis. However, many audiophiles might encounter noise or other problems when trying to unground their scopes and their shielded cables (or if they try to use two "grounds"). In such cases, the circuit shown here is recommended.



This modification introduces a slight but really negligible error in the voltage measurements. It is sufficiently accurate to distinguish clearly between the peculiar quirks of ordinary crossed-over two- or three-way speakers (impedance hump in the bass), electrostatics (impedance dip and usually a Lissajous ellipse in the treble), and the fantastically ohmic Magnaplanars (invariant straight line from 30 to 20,000 Hz).

When an impedance hump occurs at some given test frequency, the sloping line on the scope display tilts toward the horizontal; an impedance dip causes a swing toward the vertical.

Other items of interest include the facts that, 1) the Bose 901 shows plenty of Lissajous ellipse in its curve in spite of the fact that it has no crossovers, and 2) most electrostatics act like other-than-pure capacitances. Electrostatics almost always include step-up transformers in their drive circuits.

So let us not expect full-range, crossoverless speakers like the Bose or Dayton-Wrights to offer (I-V) phase-coherent loads or generally to be beautifully simple animals. Beautifully simple designs don't necessarily give beautifully simple results. —Dan Shanefield

What's Up at Phase Linear

Rumors have been circulating around Boston that some major firm (some said CBS, others Pacific Stereo) had bought into, or bought out, Phase Linear. The rumors went on to suggest that the consequent inflow of new capital explained how and why the company had introduced so many new products lately, i.e., the Model 2000 preamp and Model 1000 accessory autocorrelator. The rumors also suggested that Robert Carver, one of the corporation's whiz-kids and its president had been moved or removed.

I spoke with Carver by phone recently and he immediately assured me that the rumors were just rumors. He still heads the corporation and it remains totally owned by its founders.

Carver has been on an extended leave of absence from his executive role at the firm, spending time in research. He and his staff are working on a perfected autocorrelator intended for broadcast use, which would sell for about $1000. They also are developing a time-delay system that would recreate the ambient effects of an actual hall.

Phase Linear's new product introductions were made possible by plowing back into the company- the profits from prior sales. And there have been some. Carver also reports that the top-selling amp and preamp in the United States are the Phase Linear 400 and 4000. He notes that this has been verified by a major audio magazine survey.

Our conversation drifted to design and test of preamp phono-input stages. Carver said that the 4000 was designed using a technique unique here, although pioneered by Quad in the United Kingdom.

Carver compared his preamp's output on a dual-trace oscilloscope with the outputs of other leading and well-reviewed preamps, among them the Audio Research SP-3A, Marantz 7C, the Citation tube preamps, the Citation 11A, the Crown IC-150, several Pioneer units, and the Marantz 3300.

He also used the scope's differential input (a "null" function) to cancel identical signals from any two preamps under test, and thus leave only the difference signal on the screen. This difference signal was also auditioned via separate amp and speaker.

By using this procedure, Carver was able to adjust his prototypes' RIAA curves to within about 0.1 dB of any of the other preamps tested. The advantages to this technique, he claims, are that first, if any unit is producing distortion or behaving differently from another, the difference shows up immediately on the scope and in the audio channel, and second, that it allows the designer immediately to predict how his unit will sound in comparison with competition, and to take this into account.

In summary, Carver, and Phase Linear are alive, well, and thriving in Washington.

— Alvin Foster

Epistemology Department

Epistemology—or the determination of how we know things—appears to be on the verge of hot-topic status. Two BAS members noted Larry Klein's brief discussion of level matching in A-B testing; Les Leventhal of Winnipeg replied in the August Speaker, and both he and New Jersey member John Sprague enter the lists again this month with discussions of appropriate aspects of "theory of cognition."

If this seems a bit afield to you, take Dr. Leventhal's words to heart: "I think these issues are important because BAS members who have no access to test equipment may nevertheless make useful subjective A-B tests of equipment if subjective test methodology is refined." I am sure that Dan Shanefield and many others would agree. —Ed.

Leventhal. My reply to Larry Klein's position on adjusting volume levels of A-B'ed components appears in the August 1975 Speaker. Here I would like to reply to Peter Mitchell's editorial note, i.e., "This is becoming an epistemological inquery."

Not only is it becoming one, it started that way! Epistemology is the study of the nature and origin of knowledge. Larry's position amounts to an epistemological assertion about how we can acquire knowledge about audio components. The reason I take space to reply this month is that the sort of statement made by Mitchell often is used to put down non-empirical inquery and leaves the unfortunate impression that epistemological issues are not important. (You hit a nerve, Peter —I teach a doctoral level course in the philosophy of science and research methodology to budding research psychologists who cannot see the connection between what they do in the laboratory and fundamental epistemological decisions made by psychologists 30 and 40 years ago.)

Peter goes on to ask in what sense can amplifier A be held superior to amplifier B if "by making amplifier B just 0.2 dB louder than A we can make them sound completely indistinguishable."

One could reply that amplifier A is superior in the sense that when loudness is equated using instrumentation, A sounds better. That was easy to answer. But Peter's real question, if I may be permitted some mind reading, is more difficult and provocative. I think he is asking, 'What theoretical or practical importance arises from the fact that A and B sound alike when B is 0.2 dB louder rather than when their loudness is equal?" From the theoretical point of view, the fact that the two amps sound the same only when one is louder (as established by measurement) indicates that there are at least two psychoacoustic processes going on that in some way "cancel" or counteract each other.

For example, process number one may be something like Larry Klein's suspicion that, within certain limits, making music louder makes it sound better rather than louder. Process two may be that the more odd-harmonic distortion, the worse the sound.

So two amps may sound the same only when B is louder because: 1) B is louder, and 2) B has more distortion. Investigation into these psychoacoustic processes may ultimately benefit everyone because of what may be discovered about the connections between component design and the way we perceive things. In addition, if such counteracting processes are at work, they may provide a method of scaling the annoyance value of component misbehavior.

For example, we might say that the annoyance value of harmonic distortion in any amplifier is twice as high for the fifth harmonic as for the third harmonic, if, for two amps, B sounds indistinguishable from "distortionless" amp A when: 1) B is 0.2 dB louder (by meter), and 2) amp B is adjusted for either 1% fifth-harmonic distortion or 2% third-harmonic distortion.

From the practical point of view, the fact that two amps sound the same only when one is louder (by meter) may turn out to indicate several things about the amps valuable to audiophiles. First, even though for one or two listening sessions the amps can be made to sound alike, they may sound different over the long term—perhaps there will be more listener fatigue with one of them. Second, there may be certain conditions where they cannot be made to sound alike, as for example, with improved program material or improved associated components. [Dan Shanefield would also note changes in impedance match with the components in question.—Ed.l

To summarize, I think that, in general, components should be A-B'ed when loudness is equal by meter, with audible quality differences providing evidence of the worth (to the audiophile) of the components. If loudness is adjusted so that quality differences are eliminated (as suggested by Larry Klein), and this produces unequal loudness by meter, then audible differences will be eliminated which might otherwise have provided useful information.

There are, of course, problems associated with equating components on loudness by meter. For example, in which frequency ranges should the equating take place? What weighting curves, if any, should be used? How do you best equate components for loudness when one is producing audible noise (like tape hiss) along with signal? But these are different issues.

Finally, since I am neither an engineer nor a psychoacoustician, all this is offered as the meanderings of an audiophile, not the considered judgment of an expert. I hope that a dialogue will continue in the Speaker with our resident experts contributing on the epistemology of A-B component testing.

— Les Leventhal

Sprague. "The fidelity of stereophonic sound should not replace the intrinsic value of the music recorded." James Wm. Gayner (then Commissioner, New York State Division of Housing and Community Renewal), speaking on May 1, 1968, before the Central City Council of the Urban Land Institute in Milwaukee, Wisconsin.

How's that again? Is the above just another tedious example of bureaucratic doubletalk, doublethink, or non-think? Did the speech writer know what he was talking about? Did the speaker? And what was the reaction of the listeners, if any? It is easy to off-handedly dismiss the statement as verbal garbage. But the easy way out leaves a nagging doubt that there might have been some hidden, deeper meaning.

The statement seems to imply that increases in fidelity could lead to the listener becoming less aware of the message of the music. This contradicts the frequent admonitions that equipment has been improved to a degree that now, finally, it allows listening as if to the original performance, rather than to its own deficiencies and aberrations. Electronic components routinely have distortion far below 1%. Loudspeaker systems are now rated in percent of accuracy with the best about 90% (which is not the same as 10% distortion, as this also includes smoothness of frequency response). Phonograph cartridges and tape recorders, the other transducers, are better than loudspeakers but not as good as "pure" electronics. So, the overall fidelity of a good quality playback system might be rated as about 85%. Another 5% might be taken off at the recording end, for tape recording and microphones. Subtract more for the current state of the art of quad, at least on discs, but add some for improved dimensional effects.

These rough figures emphasize that the reproduction of recorded music produces an illusion of the original performance. How then, can the illusion replace the reality? If it at best approaches it, how could it possibly exceed it?

Fidelity is faithfulness to the original, and the higher the fidelity, the closer the reproduced imitation is to the original. Any exaggeration or diminution, however spectacular, is a lowering of fidelity. This includes such common "hi-fi" practices as dynamic range compression, reduced frequency bandwidth, differences in sound level, overemphasis in one or more parts of the frequency spectrum, and alterations of the acoustic perspective (depth, spread, etc.). Such things can certainly detract from intrinsic musical values; they are not higher fidelty even though they may sometimes clarify musical details.

Is it possible to achieve greater realism than was present at the original performance ? Clearly it is not. There cannot be greater than 100% accuracy. However, it is possible to strive for the illusion of greater realism as technological limitations are rolled back. There are those whose interest in technological ultimates exceeds their interest in music. They seek to hear the swish of the conductor's baton, the turning of score pages, and the operation of the condensation valves in the brass section. Taken in this sense, the illusion of fidelity can indeed replace the intrinsic value of the music. As any reproduced performance is an illusion, does it matter whether one's interest is in imitation of the original or in creating something beyond the original? That may depend, finally, on whether one's interest is in intrinsic musical values or in super-technology. Is there a valid reason for the latter?

At the original performance, there may be two basic types of music appreciation. One is that of the audience. The other is that of the performers. They interact and often are mutually enhancing, but they are fundamentally different. Part of the gap may be bridged when the listener sings along, pretends to conduct, or stamps and yells. Still, for most, music is something in which the role is that of a spectator. And for some, that is not enough. Some want to feel they are really a part of the performance. And years of study and practice to become even a competent amateur, much less a professional, is the hard way. The acoustic illusion that live musicians are playing next to you, or all around you, as difficult as it is to achieve technologically, is an easier solution.

There is a basic difference between creation and re-creation. The composer is the original creator, but performers are also in a creative role, even if it is one of interpretation. Virtually all composers and conductors have backgrounds as performers. One whose instrument is a "hi-fi" should not expect transformation or transportation into a creative or interpretive role.

— John F. Sprague

The Idea File

The Speaker is always looking for new articles and new authors. This note is an outgrowth of the suggestion that one of the problems with trying to write is simply not having something to write about. Herewith we present a few ideas for experiments that could be interesting and informative. We welcome any articles that these ideas might stimulate. Additionally, we welcome ideas for inclusion in this column in later issues.

1) Electronic Crossovers. There is a lot of talk about electronic crossovers and the improvement that they make in the sound of a speaker. At least, every article I have read states that electronic crossovers are the way to go for the ultimate sound. I wonder if this is necessarily true. I would like to suggest an A-B comparison between a good, familiar loudspeaker—perhaps an AR-3a—and another AR-3a adapted for a three-way electronic crossover and three separate amplifiers. The crossover frequencies, levels, rolloff rates, and amplifiers would have to be identical for the experiment to be valid, but it should be interesting.

2) Stylus Wear. What does a phono stylus really look like as it wears? The stylus inspection microscopes in stores allow you to look only at the reflection of light off the wear surfaces on the side of the needle. If someone out there has access to a scanning electron microscope, that would be the ideal tool to investigate the problem. In the same groove (sorry), what does the record surface really look like as it wears?

3) Auto Hi-Fi. One of the problems with add-on auto loudspeakers is getting enough power. Power means voltage, and a car amplifier is usually limited by the 12-volt supply. ADS gets around this in their amp/speaker setup by providing a very sophisticated (read expensive) power supply to provide the high voltage the amp needs. However, another solution would be to keep the low-voltage amp and step up the output with an autotransformer, the way MacIntosh does in their amplifiers. Has anyone tried this?

4) Cartridge Versus Preamp. It has been suggested that some problems inherent in many phono preamps are caused by the interaction between the reactance of the cartridge itself and the capacitors in the equalization feedback loop of the preamp. Could a very low-noise prepreamp used as a buffer without equalization between the cartridge and the RIAA equalization stage improve the sound? Is the prepreamp the reason that moving-coil cartridges often sound better (to some people) ?

These ideas are expressly designed to stir up some commotion. Comment and criticism are specifically desired.

— Mark Saklad

A Review of a Review— *High Fidelity* on the Allison:One

The distortion and frequency-response curves for the Allison:One speaker given in High Fidelity's October 1975 test report seem terribly at variance with the quality of the Allisons tested by the BAS and reported on in the July Speaker.

This made one or two of the listening panel a bit defensive, but the impact of High Fidelity's written review is generally positive—a stark contrast to the reproduced measurements. It would be hard to imagine a reviewer lauding a speaker as ". . . among the best systems available," if he had heard the 9.2% second-harmonic distortion (80-Hz fundamental, 100 dB SPL) noted in the table.

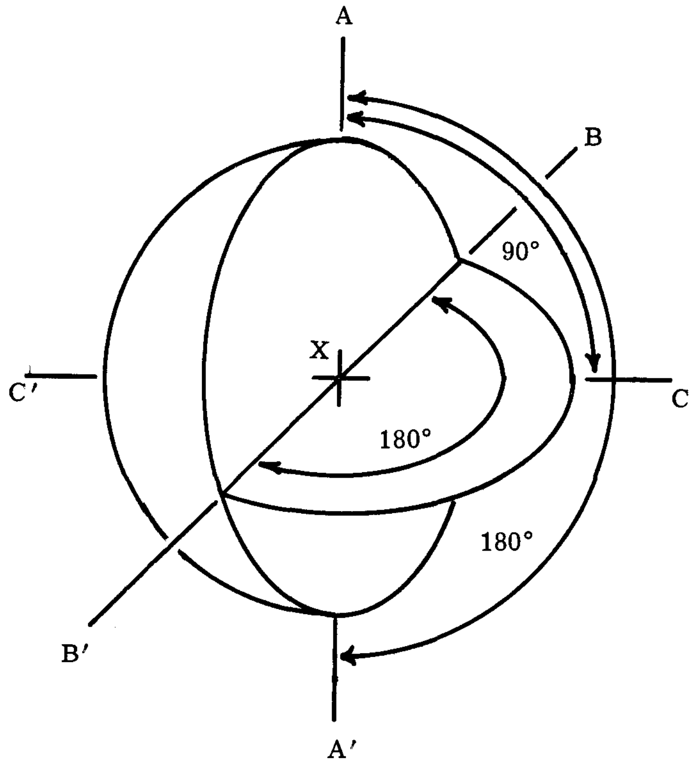
If Allisons measure "that bad," the BAS's panel (and High Fidelity's reviewer) should have heard and said something. As is, some of us feel we owe the membership some explanation and interpretation. Obviously, something went wrong somewhere, but it took a bit of thought to spot the place (s). Now we think we are at least close to an explanation, and it lies with oversight on the part of CBS Laboratories and High Fidelity.

As best as can be determined, almost everything can be traced back to the (reasonable) use of an anechoic chamber to test the Allison, and the failure to take the Allison's particular design into account when writing up the test results.

Under normal circumstances there's a lot to recommend the anechoic chamber approach: Many speakers are designed using such chambers and their performance is optimized in them; tests conducted within them offer a basis for performance comparison—but only if the reader can properly interpret the results of such tests, and that's not the case here.

The Allison was optimized for performance in a listening room with a wall behind it, and its woofers at the floor-wall intersection. The system as a whole "sees," and is loaded by, a solid angle of π steradians or one-quarter of a sphere (Fig. 1).

Placing the Allison in what is called a 4π anechoic chamber, therefore, places it at a distinct disadvantage relative to the application it is designed for. Some quick and dirty solid geometry shows why. A 4π anechoic chamber simulates a sphere of infinite radius; there are no reflections from the wall, floor, or ceiling, and for this reason measurement is simplified and usually made more accurate. Suspended in the center of such a chamber, any tested speaker must broadcast its energy into a spherical solid angle. Thus, the 4π chamber is great as a reference, but bears little resemblance to a listening room.



$\angle A \rightarrow C + \angle B \rightarrow B' = \pi$ **steradians**
$\angle A \rightarrow A' + \angle B \rightarrow B' = 2\pi$ **steradians**
The full sphere = 4π steradians
Speaker is at "X"

Figure 1

Few speaker designs take driver loading and radiation angle into account so thoroughly as does the Allison; thus when forced to operate into a solid angle four times that of its design optimum (4π versus π steradians), performance is bound to nosedive.

At first glance, the Allison would appear to have to work four times as hard—require four times longer cone excursions—to achieve a given sound pressure in a 4π chamber as in the π steradian situation. And if distortion rose and fell linearly with cone excursion (it doesn't, quite), we would be able to divide High Fidelity's 9.2% figure by four to get the "real" distortion the Allisons would yield at 80 Hz and 100 dB SPL. Such a first approximation would indicate that for the Allison, 9.2% in an anechoic chamber translates roughly and with great effort into 2 to 3% in a listening room situation (see table). But that still seems high; what else is going on?

Boundary Effects. Well, overall sound pressure for a given input power at all frequencies will increase as the radiation angle "seen" by the system is reduced. Energy density will increase linearly as the solid angle is reduced. Moving the Allison from the 4π situation into a π setting like a listening room boosts such "apparent efficiency" by a factor of four. But this doesn't take into account the wavelength-dependent nature of (wall or) boundary augmentation on speaker efficiency and therefore on distortion and low-frequency response. Here energy density rises as the square of the reduction in solid angle (see Olson, Acoustical Engineering).

Allison:One Distortion Corrected from CBS Labs Data

Output Level, dB SPL	Frequency 80 Hz		300 Hz	
	% 2nd	% 3rd	% 2nd	% 3rd
70	0.11	0.26	0.14	0.16
75	0.14	0.23	0.14	0.18
80	0.23	0.18	0.13	0.19
85	0.34	0.11	0.13	0.21
90	0.64	0.16	0.14	0.23
95	1.2	0.34	0.16	0.37
100	2.9	0.73	0.44	0.50
105			1.6	0.38

This boundary augmentation is greatest at lowest frequencies, and adding its effect means (in theory) a 16-fold total reduction in necessary cone excursion for a given SPL at zero hertz (if that's not a contradiction). This augmentation effect falls off with increasing frequency so that in the upper audio band, it is back to the fourfold reduction achieved by just narrowing the radiation angle.

This effect ought to be significant (though not approaching a factor of 16) at 80 Hz given that wavelength equals about 13.75 feet at that frequency. So it seemed safe to assume that something near 2% or less would be more appropriate for that 80-Hz harmonic-distortion figure. Allison claims to have measured 1.5% second-harmonic distortion at 25 watts input, while THD was said to have been about 1.7%. The latter casts an interesting light on High Fidelity's 1.5% third-harmonic distortion figure, much less its 9.2% second-harmonic figure for the same SPL.

That lower figure sounds more nearly right, and we can get a feeling for it by comparing the performance of the AR-LST-1, a speaker optimized in a 2π environment, with that of the Allison.

The low-frequency distortion curve for the LST supplied with the speaker (Fig. 2) shows a THD of about 1.8% at 20 watts input. Given that the Allison includes two 10-inch woofers, and takes more direct advantage of boundary effects (including an artificial one in the vertical plane perpendicular to the back of the speaker's cabinet as a result of the two woofers' in-phase movement), if equally efficient, the Allison would show somewhat better distortion in that its woofers' cones should have to move less for a given SPL (the work is shared and eased by boundary reinforcement). Since the cones themselves are smaller, there may be less likelihood of breakup, and perhaps better impulse response.

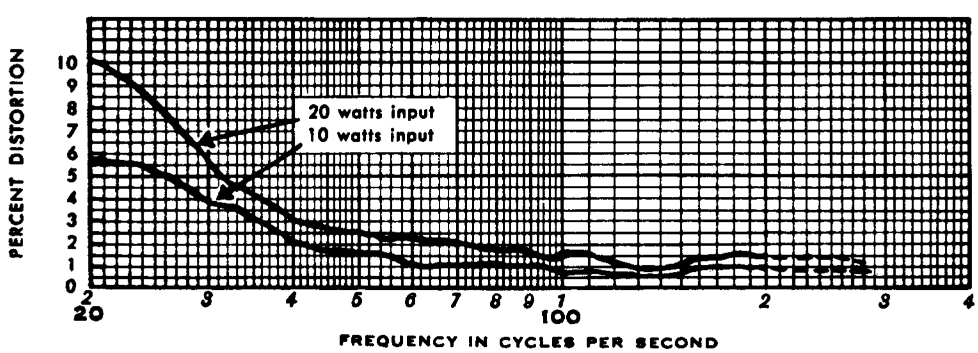
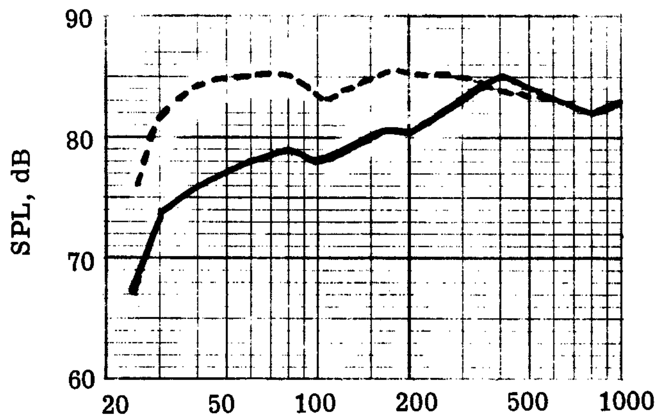


Fig. 2. AR-LST-1 low frequency distortion versus frequency

Also, relative to the LST curve, that for the Allison should be somewhat flatter with falling frequency for each of the above reasons, but mostly because of frequency-dependent boundary-reinforcement effects.

Frequency Response. Which, finally, brings up the low-end droop in High Fidelity's response curve. Note that the magazine's reviewer says that the speaker keeps right on pumping down to 20 Hz, which is just about what the BAS stated. If that's the case, the curve should be much flatter in the low end. In fact, in a listening room it is; again, it is a result of taking data in an anechoic chamber—raw—that causes the misunderstanding.

By this time, any reader should understand what's at work here and realize that boundary-effect and radiation-angle correction both work to flatten response in the Allison's low end. We will simply reproduce a curve mathematically derived from High Fidelity's and note the gain in response below about 500 Hz; it often reaches as much as 8 or 9 dB (Fig. 3).



— CBS Labs "Average Omnidirectional Response"
Allison:One in 4π anechoic environment

--- CBS Labs data plus calculated augmentation
for Allison:One at floor-wall intersection,
3 feet from adjacent wall

Figure 3

Parting Shots. Does all this mean that the BAS is vindicated by the uncompromising eye of physics? Well, partly. But more to the point, does all this have the effect of asking for special testing treatment for the Allison:One. The answer is, "No, but. . ."

The anechoic chamber obviously is here to stay as a test and measurement tool, and its use in this case was not an "evil" thing. But, the data reduction could have been much more carefully performed.

Some loudspeakers are designed to work more intimately with the room than others. The Allison is obviously one of these, but so are the Bose units, the AR-LST to a degree, and of course, corner horns like the Klipsch. For that matter, bipolar radiators like the KLH-9, the Quad, and the Dayton-Wright electrostatics are much affected by the environment as are all of the planar-magnetic loudspeakers.

Taking only the most obvious case, the measurements of the Bose 901's performance would say little about the experience of hearing its direct-reflected sound, and indeed this was taken into account in most of its reviews. But the Allison is no less dependent on the position of the walls and floor relative to its drivers than the Bose, and this was not allowed for in this review.

When the reviewer at High Fidelity heard what to him was a good-sounding speaker with outstanding bass response, a bell should have rung when he saw the CBS Labs data. If it had, the data could have been mathematically corrected and this piece might not have been necessary; perhaps next time, it won't be.

— Jim Brinton

Proposed Amendments

Several proposed amendments to the BAS Constitution and By-laws were presented at the September meeting. Each of the following proposed amendments deals with procedural changes dictated by the large percentage (over 40%) of out-of-state members in the Society.

1) Election of Officers. Election of officers takes place at the September meeting; a slate is sent to all members as a part of the September meeting announcement so that signed absentee ballots may be submitted. It is recommended that the By-laws be amended so that nominations from the floor are solicited at the August rather than at the September meeting in order that a complete slate can be distributed to all members in advance of the September meeting.

> Current wording: "A slate of proposed officers shall be included in the notice of the September Annual General Meeting. Nominations also will be accepted from the floor at the September meeting and will be followed by a vote of the membership in attendance. A quorum of one-fourth of the number of members residing in New England shall be required. Signed ballots from non-attending members will be accepted at the meeting. Election will be by a majority of those voting; if no majority is received, a run-off will be held between the two leading candidates."

> Proposed wording: "Nominations will be accepted from the floor at the August meeting and these will be included in the notice of the September Annual General Meeting. The election will take place at the September meeting and a quorum of one-fourth of the number of members residing in New England shall be required. Signed ballots from non-attending members will be accepted and election will be by a majority of those voting; if no majority is received, a run-off will be held between the two leading candidates."

2) Amendment of By-laws: The By-laws amendment procedure currently does not include a specific provision for mail balloting, and it is recommended that such a provision be inserted.

> Current wording: "Proposed amendments to the By-laws shall be presented to the membership in advance of the general meeting at which they are to be considered. By-laws may be amended with the consent of two-thirds of the members present at any general meeting, having a quorum of one-fourth of the number of members residing in New England, plus a majority of the Executive Committee, including at least three elected officers."

Proposed wording: "Proposed amendments to the By-laws shall be presented to the membership in advance of the general meeting at which they are to be considered to permit signed absentee ballots to be submitted. By-laws may be amended at any general meeting with the consent of two-thirds of the members voting, having a quorum of one-fourth of the number of members residing in New England, plus a majority of the Executive Committee, including at least three elected officers.

3) Amendment of Constitution: The Constitution's amendment procedure indicates that proposed amendments should be discussed at the meeting preceding the voting so that they can be included in the "written precis of events mailed to each member." Since such a precis now appears in the newsletter which is distributed at or after the next meeting, this would necessitate at least a two-month delay. It is recommended that the procedure be changed to one similar to that for amendment of By-laws.

Current wording: "This Constitution may be amended with the consent of two-thirds of the members present at any general meeting consisting of a quorum of one-fourth of the number of members residing in New England, plus a majority of the Executive Committee, including at least three elected officers. Proposed amendments shall be presented to the membership at the general meeting preceding that at which vote is to be taken so that they may be included in the written precis of events mailed to each member."

Proposed wording: "Proposed amendments to this Constitution shall be presented to the membership in advance of the general meeting at which they are to be considered to permit signed absentee ballots to be submitted. This Constitution may be amended at any general meeting with the consent of two-thirds of the members voting, having a quorum of one-fourth of the number of members residing in New England, plus a majority of the Executive Committee, including at least three elected officers."

In discussing the proposed amendments and the current election, the whole issue of setting a quorum was discussed at length and a number of alternatives were suggested. Since these various quorum alternatives could affect the final wording of the proposed amendments and may necessitate further amendments, the consensus of the meeting was to present the quorum alternatives to the membership, and discuss them at some future meeting. From that discussion it should be possible to reach consensus on which alternatives are feasible and these will be presented formally to the membership in the form of amendments.

The current quorum definition requires that the number of members present at a voting meeting be equal to one-fourth of the number of members residing in New England (in September 1975 this quorum was 54). A list of the proposed quorum alternatives follows:

1) A quorum present at the meeting of 35 members or one-fourth of the number of members residing in New England.

2) A quorum equal to one-fourth of. the number of members residing in New England, of which (no more than) one-third may be in the form of signed absentee ballots.

3) A quorum based on a percentage (e.g., 50%) of the members present and voting at the last meeting at which a vote was taken. (Note: This could result in a steadily declining quorum. It could also present difficulties if the membership declined drastically.)

4) Base the quorum on "active members" and have each member define himself as "active" or "associate"—the difference being the right to vote. Those members who do not wish to participate in voting would be classed as "associate" and would not be included in quorum computations. (Note: Record-keeping difficulties could result as members decide to change their status.)

5) Base the quorum on "active members" as above and have this be defined by having voted, contributed to the newsletter, or served on some BAS committee. (Note: New members would not immediately be active by this definition and record-keeping difficulties would certainly result in determining each member's status.)

6) No quorum—merely require that all votes be announced in a meeting notice sent to all members and permit absentee balloting. (Note: No voting would be possible without prior notice to all members and currently notice goes only to local members and those out-of-state members who request it. A quorum may be required for eventual non-profit incorporation.)

Comments on these various alternatives are solicited from all members. We will set aside some time at the November meeting to discuss them—the discussion period will be limited so that it will not detract from the program. If necessary, discussion may continue at later meetings. If you cannot attend the meeting but wish to comment, you may send your comments to Joyce Brinton, c/o P.O. Box 7, and they will be summarized and presented at the meeting.

These sorts of decisions regarding matters of parliamentary procedure are often very time-consuming to reach and in fact rarely affect most members. But the procedures must be agreed upon in order to assure orderly organizational functioning.

A New Address for *The Audio Amateur*

Ed Dell writes from bucolic New Hampshire with the new address for The Audio Amateur: P.O. Box 176, Peterborough, NH 03458. This magazine is widely recommended within the BAS, especially but not exclusively for those who like to build equipment of their own. If you do not subscribe to this quarterly, but are curious, write either to Ed Dell or to the BAS at Box 7 for a prospectus. (The BAS's prospectae are a better deal as with them comes a small discount. . .)

Book Reviews

Acoustic Techniques for Home and Studio, F. Alton Everest, Tab Books, $7.95

Many of us have found that the physical characteristics of the room in which we listen significantly influence the sonic illusion we perceive. In some instances there is little that can be done to improve a given room. However, in most instances, worthwhile improvements can be made with minimum expenditure of time and money. Major improvements can be obtained by either performing structural modifications on an already existing room or by building a listening room from scratch.

In either case, Everest's Acoustic Techniques for Home and Studio can serve as a trusted guide. The book is well-organized, easily digested, and relentlessly practical. The material is essentially non-mathematical and presumes no prior knowledge of room acoustics. However, numerous references are provided for those interested in pursuing any given subject in more depth.

In his discussions of the various qualities of a good home listening room or studio sound room, Everest describes how these qualities can be achieved. For example, in one chapter the concept of reverberation time is considered—how it is computed and how it can be measured. (It turns out that a knowledge of reverberation time as a function of frequency is one of the major ways to characterize a given room.)

A good portion of the book is devoted to the sound absorption characteristics of different kinds of materials and various wall and ceiling structures. For example, Everest notes that one of the few ways to absorb energy in the troublesome mid-bass range without affecting energy in other parts of the spectrum consists of judicious use of standard wood paneling mounted on 3-inch studs spaced 16 inches apart.

This book is unique in many ways and is well worth the investment. Copies may be obtained directly from the publisher ('TAB Books, Blue Ridge Summit, PA 17214). — Gary J. Rancourt

Audio IC Op-Amp Applications, Walter G. Jung, Howard Sams Publ.—Bobbs-Merrill Co., $4.95

Walter Jung has been a prolific writer about op-amp applications for several years now. He has recently written a multipart series on audio applications for The Audio Amateur. This new 144-page book is extracted in large part from a much larger work of his, The IC Op-Amp Cookbook. The Cookbook was published earlier this year and has received universally excellent reviews.

This book is not a primer for the first-time user of op-amps. It assumes that the reader has some prior knowledge of what an op-amp is and what it can do. The organization is directed toward both theory and applications.

Jung first discusses general considerations of stability, bandwidth, slew rate, compensation, and offset adjustment for a variety of common op-amps. However, the National Semiconductor LM381, which was designed specifically for audio use, is conspicuously absent. After this introduction, he moves on to specific circuit configurations useful in audio applications.

In his applications chapters he thankfully does not just give a circuit with a quick description and suggested component values. He also includes selection criteria for the amplifier involved, compensation suggestions to increase stability and reduce distortion, and component value recommendations when alternatives are available. He states clearly why something is done; he doesn't just say do it this way. I find I now better understand several circuits with which I had previously been familiar.

Besides standard circuits phono preamps, mike preamps, power boosters—he includes such unusual designs as shelving tone controls, signal summing amps, equalizers, and a stereo pan-pot circuit (one signal in, two out which can be panned from left to right).

It is a good book, easy to read and easy to understand if you have some minimal op-amp knowledge. — Mark Saklad

In the Literature

Acoustical Society of America, Journal of the, Aug. 1975

- Spiral Membrane Speaker Enclosure: If you thought Yamaha's ear-shaped speaker was far-out, take a look at Fig. 1. (p. 446)

Audio, Oct. 1975

- Build an Audio Generator: More elaborate than the BAS generator. Like most low-distortion units it requires stepped frequency selection. (p. 22)
- If you like reading phone books, the October Audio Equipment Directory is for you. Tables of manufacturers' specs can be misleading, but this is easier to scan than is the Stereo Review Buyers Guide.

Electronic Servicing, Sept. 1975

- Servicing Stereo Audio Systems, Part 1: Pickups, plus a review of the electronics and controls that follow them. Pretty basic, but this is only the first installment.

Electronics, Aug. 21, 1975

- CBS Returns to the Basics: One-page report on the reorganization of CBS Labs. (p. 72)
- Linear Pot and Op-Amp Provide Tapered Audio Control: Simulation of log-response. (p. 83)

- Assuming that many audiophiles are also photobuffs, see "Smart Cameras Clicking with Electronic Functions," a review of IC and light-sensor technology in the camera and flash market. (p. 74)

Electronics, Sept. 4, 1975

- Electronic organ IC's reviewed. (p. 110)

Popular Electronics, Oct. 1975

- Stereo Scene: What's New for Hi-Fi in 1976: Ho-hum. (p. 22)

Radio-Electronics, Oct. 1975

- QS Matrix Simplified: Encoding and decoding in the QS system. (p. 16)
- Build a Four-Channel Synthesizer: Enhancement system for stereo sources. (p. 33)
- Bookshelf Speakers, Part 1: Mainly dispersion data on a collection of speakers. Reproduced from data supplied by manufacturers, i.e., not measured by R-E. (p. 41)
- Super-Fi Testing: Review of top-of-the-line audio testing and servicing gear. (p. 50)
- Add-Ons to Improve Your Hi-Fi Rig: Compilation of accessories with comments. (p. 52)
- Test Reports: Soundcraftsmen RP2212 preamp/equalizer (very short) and Pioneer CT-F9191 cassette. (pp. 36, 39)

Sky and Telescope, Sept. 1975

- Astronomy and Music: This one is really spaced-out, but in addition to discussing the usual music-of-the-spheres, a discography of astronomy-related music is included.

WCRB Guide, Oct. 1975

- Announces what we knew already, that AM is going to "Big Band Music" and FM is going to separate programming, with the promise that FM will maintain its classical format.

Wireless World, Aug. 1975

- Peak-Reading Audio Level Meter: Discrete LED stereo indicator system. Not for the novice builder and no "kit" is offered. (p. 357)
- New Domestic (British) Equipment, and Consumer Electronics in the USA: Reports of the new summer equipment announcements.

September BAS Meeting

Business Meeting

The September meeting started 25 minutes late after a fruitless wait for a quorum. The quorum requirement in the by-laws requires that 25% of the New England membership (a total of 216) must be present for a business meeting, and only 43 paidup members were present of the 54 required. The absentee ballots were not counted for quorum purposes. As a result, the election of officers was postponed to the next meeting. More details and a number of suggestions made to deal with the quorum problem appear in another part of this newsletter. The September business meeting normally has the smallest attendance, and we have reached the quorum requirement in the past with usually only a few over the required number.

The quorum requirement operates as a protective device for the members so that the aims, purposes, and funds of the society cannot be controlled by a very small coterie. It can also be an impediment. One organization I belong to has business meetings four times a year. At least one of these four meetings can do nothing because of the quorum requirement, even though the requirement has been lowered twice and requires less than 50 members out of 1200 to be present! The matter of setting a quorum requirement should be given some careful thought before coming to a final conclusion.

In addition, our By-laws and Constitution need changing since we have changed almost everything about the newsletter this year, including the timing of writing, printing, and distribution. This means the newsletter no longer carries the meeting announcement, but the By-laws and Constitution require any proposed changes in the Constitution or By-laws to be announced in the newsletter before the next meeting. The proposed new sections in the By-laws and Constitution would require written notice to the members before the meeting at which a vote is to take place on the proposed change in the Constitution or By-laws in sufficient time that those members unable to attend may vote by mail. Also, the By-laws need to be changed so that nominations for officers are made at the August meeting so that absentee ballots can be printed and distributed with the next meeting announcement. More detail on these proposed amendments appears elsewhere in this newsletter. Further discussion will take place at the next meeting.

Jim Richardson took more tape orders but announced that these would be the last until sometime next summer. Peter Mitchell delivered oscillator kits at $17 for a kit without a circuit board and with an unpunched box or $19 for a kit with a printed circuit board (an experimenter's board from Radio Shack) and a punched box. We are all lazy, because only the latter were sold at the meeting. For those mailing in orders, please remember to include $2 additional for postage. Insurance is also strongly recommended.

Peter Mitchell reported that Rene Jaeger had commented to him that the best book on tape recording was "Modern Recording Techniques" by Robert E. Runstein, published by Sams as their number 21037 with a list price of $9.95. Sams' description follows: "Provides the information necessary to prepare pop music recordings. Explains the equipment, controls, and techniques found in the modern recording studio and how to use them not only properly but creatively to produce a desired result."

Meeting Feature

Joseph Hostetter, head of the Recording Arts Department of the Berklee College of Music, was the featured speaker. He gave a very brief history of the college, including the founding by Lawrence Berk in 1945 and its evolution into a college granting Bachelor of Music degrees with its reputation based in the areas of modern commercial and jazz music. When he joined the faculty in 1967 the school had about 300 students, whereas it now has about 2500, a rather rapid growth which required their acquiring their present quarters, a former hotel and adjoining former movie theater, still in the process of remodeling and renovation. It is his opinion that the current enrollment makes this music school one of the largest in the country, including departments of music at universities.

Hostetter played trumpet as a child and formed a small band while in high school. In the mid-fifties he became interested in audio, building a Heath Williamson amplifier and several loudspeakers. Then he became interested in tape recording and recorded the small group he played with. While in the Navy, he made use of the low Hong Kong prices and bought two Neuman U47 microphones and a mixer. Later he bought a used Berlant Concertone tape recorder, which he turned into a recording business while in college.

When he first came to Berklee he used the Roberts recorder and microphones that the school then had, but he soon brought his own equipment to use, and from this evolved the present setup including a small recording studio, control room, 8-track Scully, and a very good control board. The quality of the control board is necessitated by the daily operating schedule, which usually runs from 9 a.m. until after 11 p.m.

The recording studio is used to make many of the teaching tapes used by the students and teachers as well as the master tapes dubbed onto cassettes for use in the cassette laboratory, which has 40 machines. It is also used to record students in the process of advancing their playing techniques. In addition, it is used to teach students the proper "behavior" at recording sessions, for if he goofs up in a commercial recording session, he may be fired and blacklisted for the rest of his career. Finally, it is used to teach students the art of recording, but not specifically to turn out recording engineers. He uses Runstein's book as the course text.

Tapes were played demonstrating some of the types of music recorded for student study, including isolated two-track tapes with a rhythm section on one track and suggested accompaniment on the other and student compositions. Also demonstrated were concert recordings from the late sixties to the present. Joe said he preferred to record acoustical bass and most of the selections used acoustical bass.

He talked about miking setups used both in concert recording and studio recording. The mike setups in concert recording that were presented are those he uses. Since over 100 concerts are recorded in a year, many of them must be done by student recordists who use a Sony tape recorder and an X-Y microphone pattern. Simple and effective for concert recording, but as our speaker pointed out during the question and answer period, it is not usable in studio recording because of the excessive pickup of nearby instruments at the wrong levels and phasing.

Throughout his talk, Joe discussed mike setups that he had used for each tape recording sample that he played and the mike setups he had used back to his high school days. The mike positioning was optimized if only one group was to be recorded, but a compromise setup had to be used for multiple groups. All of the concert recordings could have been duplicated by an amateur recordist, there being no ceiling-hanging or exotic microphones used. Microphone dynamic range is important because some of the musicians play so loud, in particular the saxophone players. The distances the microphones were set from various instruments to achieve loudness and instrument balance were determined by sense from within rather than a rote formula. Microphone gains in the mixer box were also set the same way, but sometimes a musician would surprise the recordist by putting the bell of his instrument very near or even over the microphone. Attenuator pads were sometimes used with some microphones to bring the output within convenient range of the mixer controls and to prevent input stage overload.

In the Boston area, there are many musical groups who are non-union and who are willing to record. As has been said many times at BAS meetings, it is not an expensive proposition to start live recording if you already have a tape recorder. Two microphones, some long mike cables, a tape recorder, a pair of headphones, and some willing subjects are all that is needed. Those members who are out of the Boston area and have never heard some of the recordings made by BAS members using Advent 201 or equivalent reel-to-reel recorders and two 814 or Advent mikes have no idea how superb such recordings can be. Our guest speaker used more expensive mikes than those since his teens only because such good, low-cost mikes were not then available.

No groups available? Try your church organ and choir. They have rehearsals and I know several people in northern New England who got their feet wet recording at church rehearsals.

— Keith North

A Line-Level 814 Microphone System

Rene Jaeger and James Reilly
dbx, Incorporated

Introduction

All prior implementations of the Thermo Electron 814 microphone capsules have had some characteristics that, while not failings, might have made them less convenient or useful to audiophiles than they might have been. Among these have been the difficulty, admittedly minor, of getting power to the microphone capsule, and in some cases the use of low-level, unbalanced outputs that could lead to pickup of radio-frequency or power-line interference.

A simple and effective method of sidestepping these potential problems is to place within the microphone assembly an amplifier with enough gain to eliminate the need for relatively costly and complex balanced lines by raising the microphone's output level high enough to make interference and hum inaudible in most situations where cable length doesn't exceed 100 or so feet.

That's what has been done here. By way of a further departure from earlier implementations, the 814C capsule has been used rather than its opposite number. Also, an effort has been made to reduce the apparent self-noise of the capsule by lowering the source resistance "seen" by the preamplifier.

The design is fairly successful: frequency response flatness is a function of the capsule selected, and even with double the recommended lengths of cable (Belden 8412 or 8422 is recommended) frequency response remains within ±0.5 dB of nominal from 20 to 20,000 Hz. Overload with this system occurs at about 132 dB SPL, while residual noise is approximately 20 to 22 dB; this gives a dynamic range capability for the system of about 110 dB.

The system divides neatly into three parts: the at-capsule preamplifier, a current-sink circuit located in a remote chassis, and in the same package, a power supply, which may be either a battery-type or ac-line-powered.

A final note regarding convenience—even though standard high-quality microphone cable and Cannon XLR connectors are used to mate the microphone with the remote chassis, the output from the chassis to the recording system is unbalanced and at line levels. This means that it can be plugged directly into the wide variety of mixers, tape decks, and noise-reduction units (including dbx's own 120-series units) that use RCA-type phono connectors. Thus, not only is there no need for separate microphone preamplification, but the system also is connector-compatible with almost any of the semi-pro/amateur recording equipment with which it is likely to be used.

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Technical Description

Unlike most of the earlier implementations of the 814 capsule, this one is built around the 814C. A resistor, R1, is added to the drain circuit of the capsule's FET, which now becomes a common-source amplifier instead of a source follower (see Fig. 1). This feeds the base of a pnp common-emitter stage, Q1, who's collector load is another common-emitter stage, Q2, connected as a current sink. This means that it will always draw a fixed amount of current regardless of the potential at its collector. This assures an adequate voltage swing in the negative direction.

A resistor could have been used, but that would have required doubling the negative supply voltage and would also have reduced the gain of the output stage. Q1 must add or subtract current from the preamp's output to produce a voltage swing. It must also drive the feedback resistor, R4, as a load. This resistor, in conjunction with R3 at Q1's emitter, sets output gain to 20 dB (R4/R3 = 10).

C1 allows us to dc bias this stage with R2 and set gain with R3. The feedback resistors R4 and R5 in parallel with the FET's source resistance set overall gain to about 22 dB. Higher preamp gains are possible if R4 is increased in value, since this will increase the load for Q1, raising stage gain proportionately as overall gain is raised. For example, increasing R4 to 10K will yield 30 dB overall gain.

It might seem possible at first glance to eliminate R5 and C2 entirely and simply use a larger value for R4, thus increasing gain before feedback, and giving lower distortion. However, initial calculations indicated that the 5K resistor in the 814C capsule was responsible for about 4 dB of the capsule's stated self-noise. Therefore, it was decided to lower the source resistance by paralleling it with the external R5. C2 ensures that the dc bias point of the FET is maintained primarily by the 5K source resistor. Some current does flow in R4 as a result of the potential difference between the FET source (about +3 volts) and the feedback point (at about -1.5 volts). R6, by the way, is an isolation resistor in series with the current sink and helps isolate capacitive loads (i.e., long cables) from the feedback loop. C3 is a small disc for RF bypass; C4 takes care of RF on the B+ line, while C5 and C6 are the power supply bypasses to ensure a low impedance even as the batteries run down.

R7, R8, and R9 set the operating point of Q2. C7 is a blocking capacitor used in case of small dc offsets. Note that the current sink and the heavy bypass capacitors are removed from the mike by however many feet of cable you use. This means that only two wires plus ground are needed for the run.

This part of the circuit, plus power supply (either batteries or an ac design noted below) and connectors, should be mounted in a suitable box. Any number of mikes can be driven from a common supply so long as the supply can deliver 6 to 7 milliamperes per mike. Three 9-volt transistor radio batteries (such as 2U6 or 2N6) may be used; the 2U6 will yield about 4 to 6 hours of operation per mike while the 2N6 will give about 30 to 50 hours per mike. Divide these hour figures by the number of mikes being powered for the operating life of the batteries under actual conditions. Finally, 8.4-volt mercury batteries could be used with a small sacrifice in output swing.

For those with more enthusiasm, a power-supply design is shown in Fig. 2. A 24-volt transformer delivers about 35 Vdc to a large filter capacitor through a bridge rectifier. This voltage will be typical of a 24-volt, 1-amp rated secondary with very light loading. Thirty volts is the minimum allowable to keep the regulator going and we need a safety factor for low line voltages. Forty volts is the maximum allowable and again we need a safety factor. Thus 35 volts is best if you can get it.

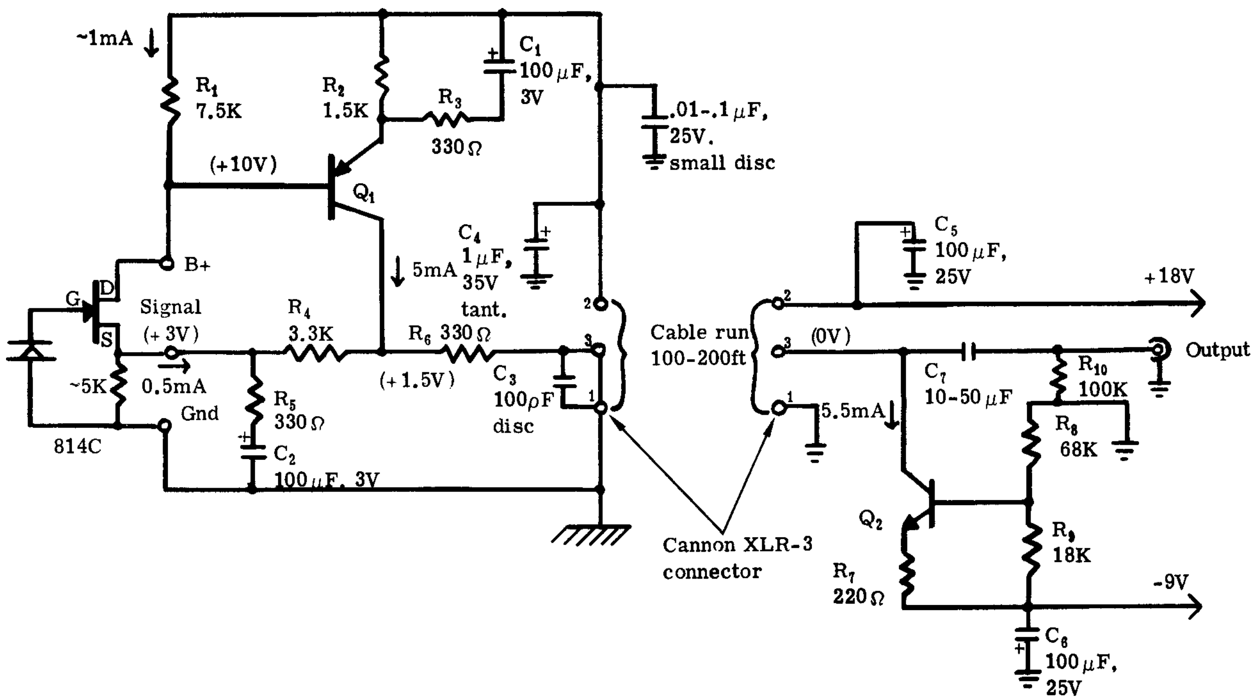

Fig. 1. Mike preamp and current sink

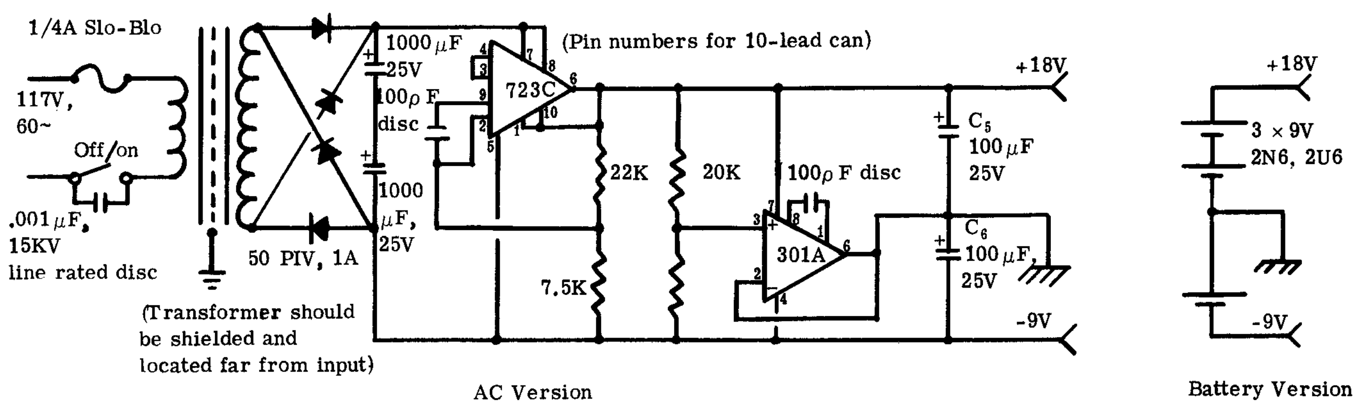

Fig. 2. Power supply

This voltage is regulated at 27 Vdc by a 723C integrated-circuit voltage regulator. This regulated voltage is then referenced to ground by means of an LM 301A-type IC connected as a voltage follower and referenced to a voltage one-third of the way from the negative to the positive supply (note the 20K to 10K voltage divider at the IC's "+" input). The 301A's output is connected to what now becomes ground or common. Since its supply is the regulated 27 volts itself, its job will be to source or sink current in an attempt to keep the ground 9 volts away from the negative supply or 18 volts away from the positive supply. Of course it can sink or source only about 20 mA, but since these mike preamps draw approximately equal currents from both polarities, the 301A need make up only a relatively small difference.

When driving a load (such as the feedback resistor, R4), the filter capacitors C5 and C6 provide the peak currents required, but the average of positive- and negative-going currents remains zero.

Note that if more than eight mikes are to be powered, one should use larger values for C5 and C6.

Construction

Assembly is possible in a 5/8-inch-diameter thinwall aluminum tube and practical in a Switchcraft housing. (See "A Case for the 814," by Alan Southwick, BAS Speaker, March 1975.) The main difficulty in packaging will be the 100-microfarad capacitors, which must be small. At dbx, we used Siemans B41313 style units. Tantalum capacitors will work if you can afford them.

Q1 may be any popular low-noise pnp such as the 2N4250 or 2N5087. Q2 may be most any npn with good collector saturation characteristics such as the 2N3904, 2N3565, or 2N5088. R2 may have to be trimmed in value to suit the particular 814C FET. The desired result is nearly zero volts at the output pin (Q2 collector) ±1 volt.

R1 should be a low-noise carbon-film or (better) metal-film resistor. Ten-percent tolerance can be used for all resistors, but 5% is preferred for R1, R4, R5, R7, R8, and R9. Five-percent resistors also should be used in the power supply.

Additional RFI protection in the presence of (say) 100-MHz energy fields such as might be encountered near an FM station can be had by putting a 0.01- to 0.1-microfarad, 25-volt disc capacitor in parallel with C4. This is within the mike package, so size is important; use a small disc. Note also that C4 should be a tantalum capacitor.

Further RFI protection can be had by paralleling another similar disc capacitor with C6 between the -9-volt bus and ground.

Acknowledgement

The authors wish to thank Alan Southwick for his aid with the original manuscript and schematics for this article.

Cautions, Addenda, and Miscellany

The output from this mike system can reach 7 volts on peaks. It should therefore be obvious that this version of the 814 should not be used with a microphone preamplifier or plugged into the "mic" inputs on your tape recorder. Its signal should be fed to line-level inputs only, whether on a noise-reduction unit, mixer, or tape deck.

To maintain the best spurious signal rejection, plan your installation so that the run of phono cable (typical in amateur installations) between the system mainframe and your noise-reduction unit, mixer, or tape deck is as short as possible.

Given that line-level signals are present in the mike cables, it becomes possible to err through overconfidence, ignoring hum and other fields, and running either overly long cables or locating them creatively to get interference in spite of the design. One-hundred-foot cable lengths are easily tolerated by the system, and 200-foot lengths have little effect on frequency response (say, 0.5 dB, 20 to 20,000 Hz). But longer cables do increase the risk of interference becoming audible given the system's low residual noise of about 20 dB.

With the occasional need for long cable, and the possibility of unintentional abuse in mind, Mark Davis suggests the possibility of using what is basically a unity-gain common-mode (noise) rejection stage between the system's output and your mixer, tape recorder, or noise-reduction unit. The design, shown in Fig. 3, incorporates a variable low-frequency rolloff of 6 dB/octave below either 10, 100, or 200 Hz. This should be helpful in location recording situations troubled by subway, elevator, air-conditioning, or traffic noise.

In addition, there is a switchable gain control at the output of the op amp. By switching between the three positions, it is possible to get either unity gain, -10 dB, or +10 dB of gain relative to the basic 814 system's output.

This feature might be of importance, for example, to owners of Revox A77 tape recorders; these machines have no attenuators prior to the first active stage beyond the line input. Seven volts at the line input of some Revoxs can send this first stage into clipping distortion. Thus, the -10-dB gain setting might prove useful in high-level or close-miking situations.

Similarly, if the 814 system is to be used to drive an unfavorable load, the +10-dB setting might be useful for opposite reasons.

The Davis modification could be built in a separate box and be battery powered or could be included in the 814 system's chassis.

— Jim Brinton

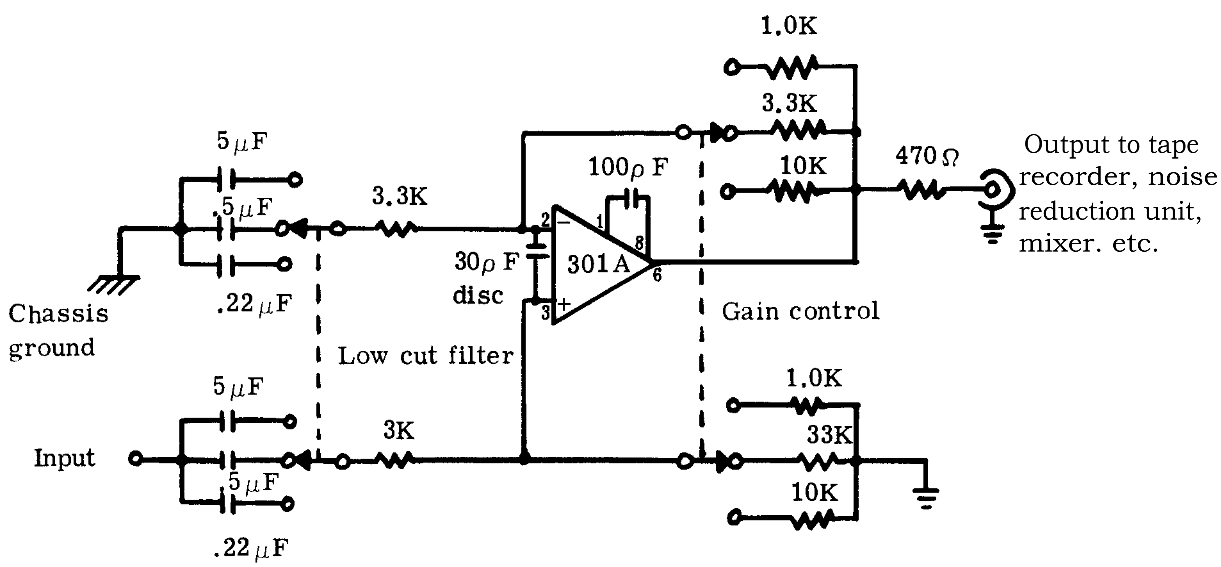

Fig. 3. High common mode rejection add-on (one-channel shown)