THE B.A.S. SPEAKER

Editorial Board: James Brinton, Peter Mitchell
Coordinating Editor: Peter Mitchell
Production Manager: Robert Borden
Copy Editor: Joyce Brinton
Staff: Richard Akell, Stuart Isveck, Lawrence Kaufman, John Schlafer, Peter Watters, Harry Zwicker

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

VOLUME 3, NUMBER 11
AUGUST 1975

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

In This Issue

The content of the Speaker this month reflects the interest of audiophiles in both technology and music. The July meeting featured interesting comparisons of recorded performances of a range of music, and also featured demonstrations of two time-delay systems for recreating in the home the reverberant ambience of the concert hall. The development of such systems may prove to be the most important event of the year (perhaps of the decade) in audio. We will follow this up in future issues and will review purchasable systems as they become available.

Returning to music, we all yearn for consistently excellent recordings since the sound of our records remains a principal limitation on the sound our stereo systems produce. But the erratic quality of the discs in local shops often deters us from buying at all, and inevitably we miss some great ones. Some of us have adopted the philosophy that the only discs we can buy with confidence are those on the Philips label, but that approach can lead to a surfeit of Haitink. British audiophiles have it better: their reviewers provide more reliable comments on technical quality than do the writers in American magazines, and there are many excellent records released in England which are hard to find in the USA. Obviously we all should be ordering British (and other European) records, and this is easier to do than you may have supposed. In upcoming issues we expect to publish procedural suggestions from members who have bought from British dealers. (Members who have done so, please write!) Of course you then need to know what discs to order, so in this issue we have the first of several lists of recommended records.

As usual, however, equipment dominates the issue. Peter Mitchell has supplied two technical contributions. One is a note on the BAS remote-powered 814 microphone and an inexpensive preamp which is an ideal mate for the mike. The other is a collection of suggestions on useful things you can do with the BAS oscillator or any other good sine-wave oscillator. If you have other ideas for using the oscillator, pass them on. To round out this issue we have some important data on open-reel recording tape, some interesting letters, and news of a new magazine for audiophiles.

In Future Issues

We have several do-it-yourself projects in various stages of preparation for publication this fall: a very wide-range audio voltmeter (a companion for the BAS oscillator), an active A-weighting noise filter, a filter to eliminate subsonic and ultrasonic garbage, a phono preamp, a square-wave generator, a sharp notch filter for reducing hum or the Tanglewood whistle, a PLL stereo decoder

Copyright © 1975 The Boston Audio Society

for your tuner, and simplified procedures for improving the AR turntable. But we don't want the Speaker to become a journal just for the solder-and-spaghetti crowd, so we hope some of you are preparing to send in equipment reviews, notes on new products, d comments on music or recordings. If you don't want to write even something brief, how about telling us what you want to read about?

A couple of issues back Dan Shanefield mentioned his high opinion of the Magnepan speakers. A few of us have now heard a pair of these and our first impressions are very favorable (especially since their price is not exorbitant). We hope to A-B them against Allison Ones, AR 10π 's, and more expensive competition such as multiple KLH-9's.

Wanted: Letters

You may have read this message before. We've certainly written it before, but evidently it bears repeating. The BAS Speaker is intended to be the publication of the entire Society, not the product of just a small clique of favored writers. We really mean that. The same four or five of us can continue to provide material for the Speaker, of course, but is that really enough? These pages should represent more diverse points of view. For example, if you are like most of us you invest in at least one new component each year, either to expand your system or to replace an older item. Whenever you obtain something new, give us two paragraphs describing the best and worst features of the product. If it's a genuinely satisfying product, we all would like to hear about it, and if in some way it's not all it is cracked up to be, we certainly want to learn that too. What are you waiting for? Do it now before postage rates rise again!

Last Call for the BAS Tuner Clinic

If you want to have your FM tuner or receiver tested, call or write Joyce Brinton immediately. The final clinic sessions are being scheduled now.

Erratum

In the article on headphone amplifiers in the June issue, it was not made clear that the layout diagram in Fig. 2 shows the PC board as seen from its upper, nonclad side.

Used Equipment

For sale: Dynaco Stereo 120 amp in mint condition, with performance graphs and original packaging. $100. Dennis Boyer, 566-5972.

The BAS Oscillator Kit—Available at Last

After a year's delay, the BAS sine-wave oscillator kit finally is available. It is slightly improved over the original version, as noted in the appendix to the article on using oscillators in this issue. This is the only low-cost oscillator we know of which produces an undistorted sine wave sounding absolutely pure; other inexpensive oscillators produce audible distortion. The price for the basic kit is $17. Since some members are not equipped to drill the large (3/8-inch) holes in the box chassis or the small (no. 60) holes in the circuit board, the kit is also offered with pre-punched chassis and predrilled PC board for $19.

— Peter Mitchell

Options with the 814

Members now have so many available options for packaging and using the 814 microphone capsule that an organized summary seems appropriate.

1) The least expensive approach is to install the mike (in a socket) on the end of a cable without a casing, and to battery-power it, as shown in the July "814 Column." This can be done for either the 814 or 814C capsule. Under this heading there are several suboptions: (a) a kit of parts is available for $10 (consisting of socket, capacitors, battery connector, 20 feet of Belden cable, and a steel three-circuit phone plug); (b) this kit is available in assembled form for $20; (c) you could buy the parts and cable elsewhere.

With the simplicity and economy of this system you also buy some disadvantages. The wiring to the back of the socket is not encased and so is somewhat fragile; if you use the mikes often and carelessly, you may someday, as I have twice done, set up at a concert only to discover a dead mike because of a broken connection. Without a casing the mike is also inconvenient to hand-hold or mount on a stand. The only practical ways to use it are: (i) tape the battery/cable assembly to a string stretched between balconies; (ii) if stands are required, tape the cable and battery to a boom with the mike end of the cable hanging over a couple of inches beyond the end of the boom; (iii) install the mikes in a binaural head. Finally, since the connections to the mike socket are not shielded, the mike is relatively prone to RFI—notably hum if hand-held (because the body is an ac antenna) and a buzz in the vicinity of SCR light dimmers.

2) The next step up is to install the socket, battery, and connector in a grounded casing to provide mechanical protection, RFI shielding, and a convenient physical form for stand-mounting. This requires the parts listed in option 1(a) above plus aluminum or brass tubing, a battery holder, an on-off switch, and some mechanical ingenuity. Since tubes of convenient length may be difficult to obtain, I have procured 8- and 12-foot lengths of aluminum tubing in 5/8-, 3/4-, and 1-inch outside diameters and can supply short lengths (specify) for $1 each. I also have lots of cheap battery holders which accept either a type AA 1.5-volt penlight battery or a type 126 8.4-volt mercury battery. Since we have not settled on a satisfactory mechanical arrangement for mounting the socket, battery holder, switch, and cable connector in the tube, an assembled version is not offered.

With either of the above approaches you may either (1) wire the cable "unbalanced" for use with whatever mike inputs you presently have in your recorder, or (ii) wire the cable "balanced" for use with professional-type mike preamps and mixers. If you do want to use a better mike preamp than the one in your recorder, the balanced-line Advent MPR-1 is the best bet ($35 at your dealer). (The essentially identical Wollensak preamp can be obtained for $25 in a group purchase via Jim Richardson.) The input transformers of these preamps produce a mild but acceptable bass rolloff with the 814; the transformers in some other professional-class preamps produce even larger variations in input impedance. Incidentally, if you plan to use the 814C capsule to record extremely loud sources close-up, you will also need an input attenuator to prevent preamp overload. This is easily built with a few resistors.

3) Ira Leonard has described another battery-powered approach in which the mike socket is mounted at one end of a pencil-sized tube about two feet long, with the battery, capacitors, switch, and cable connector in a mini-box at the other end of the slim tube. Contact Ira for details. This can be used with any of the preamp options mentioned above. By making the mini-box a little larger you could include a mike preamp in it (using the LM381A IC for instance), thus yielding line-level signals for transmission down the cable to your recorder.

4) The mechanical problems of mounting a battery and on-off switch in the mike case are solved by rewiring the mike for remote powering. Several options are available here too: (a) an assembly described by Alan Southwick in March, in which everything is artfully squeezed into a modified Switchcraft connector; (b) a kit of parts described in July, with cable connector pre-installed in a 6-inch unfinished aluminum tube of 5/8-inch o.d., costing $20; (c) the above kit assembled, for $50 until some enterprising member offers to do it for less.

These remote-powered mikes are intended for use with balanced-line preamps. If you choose the Advent or Wollensak preamp, it can be modified to phantom-power the mike, or you can build an external phantom-powering battery box which would connect between the mike cables and the preamp (such a battery box could also be used to power the preamp itself). Alternatively you may use the preamp described in this issue which has provisions for powering the mikes (but requires reversing the polarity of a capacitor in the mike assembly); so if you choose to buy or build remote-powered mikes you should decide in advance on the preamp you will use so that the mike wiring can be chosen accordingly.

5) These remote-powered mike assemblies are suitable only for the 814, not the 814C. Rene Jaeger has developed a circuit for the 814C in which additional transistors are added at the mike, yielding a system capable of extremely wide dynamic range. This will be described in a future issue.

— Peter Mitchell

Some Excellent Records

Recently we were requested to suggest a list of some records having excellent sound and good performance, discs which would be suitable for demonstrating a good playback system. A requirement was that the records be purchasable in an ordinary domestic record shop, ruling out special-order discs and European pressings such as EMI. The following list was produced by Dick Goldwater and Peter Mitchell. The list is not exclusive; doubtless there are numerous other records as good as these. Note that these discs were chosen for natural, not necessarily spectacular, sound. They are listed in random order.

Rossini: String Sonatas (Marriner), Argo S-506 and ZRG 603
Mahler: Symphony No. 4 (Horenstein), Monitor S-2141
Weill: Suite from Three-Penny Opera/Milhaud: Creation of the World (Weissberg), Nonesuch 71 281
Beethoven: Symphony No. 6 (Jochum), Philips 839782
Berlioz: Overtures, miscellaneous (Boulez), Columbia M31799
Berlioz: Symphonie Fantastique (Davis), Philips 6500 774
Mozart: Symphonies Nos. 25 & 29 (Marriner), Argo ZRG 706
Handel: Concerti Grossi Op. 6 (Leppard), Philips SC71AX 302 (3 discs)
Haydn: Symphonies Nos. 22, 39, & 47 (Leppard), Philips 839796
Scarlatti: Sonatas (Kirkpatrick), ARC 2533 072
Bach: French Suites (Dreyfus), ARC 2533 138/39 (2 discs)
Rimsky-Korsakov: Scheherazade (Haitink), Philips 6500 410
Mahler: Das Klagende Lied (Haitnik), Philips 6500 587
R. Strauss: Also Sprach Zarathustra (Karajan), DG 2530 402
Mozart: Piano Concertos Nos. 12 & 17 (Brendel), Philips 6500 140.

Another list, compiled by David Ranada, was not received in time for inclusion; we hope to publish it next month.

— PWM

British Record Recommendations

It has become a truism that European-made records are generally superior to American pressings, and one of the reasons for this was discussed by Bruce Maier of Discwasher in the July issue. But since many of the best European discs are not sitting in the racks at our local record shops and are not reviewed in the magazines which most of us read, we face two problems: learning about the good records and then finding out how best to obtain them. Last month's recommendation of Maildisc & Co. of England was the first of what we hope will be a series of pointers from experienced members on where to get European discs without fear or pain.

To identify the good records, the first thing to do is to make a habit of reading the better British record-review magazines. British reviewers tend to pay more attention to the quality of recorded sound than do their American counterparts, and in reviewing a new disc, they also often include valuable comparisons with other recorded performances of the same music rather than treating the new release in a vacuum. In addition, two magazines (Hi-Fi News and Record Review and the Gramophone) publish quarterly re-reviews of the best discs of the preceding three months to select those offering the best sound and performance. If you have an unlimited budget, there are other review magaiznes which are well worth reading (Records and Recording, for instance), but the big two are the best bet for a record buyer's dollar. An annual subscription to Hi-Fi News and Record Review costs 7.5 pounds (about $18.75 at current exchange rates) from Link House, Dingwall Avenue, Croydon CR9 2TA, England. If you buy more than a dozen records a year it's well worth the money, because HFN/RR pays closer attention to the quality of recorded sound than any other magazine I know of. A subscription to Gramophone costs $16.50 from General Gramophone Publications, 177-179 Kenton Road, Harrow, Middlesex HA3 OHA, England. They also publish the Gramophone classical catalog, which is decidedly more useful than the Schwann catalog because of the limitations which Bill Schwann has deliberately placed on the content of his catalogs.

Since most of our readers have not been subscribing to these magazines, we publish below the first of a series of lists of recommended records. These recommendations are based on reviews in the British magazines together with judgments of performances (not sound) by American reviewers; i.e., the discs in these lists have been judged good in performance by reviewers on both sides of the Atlantic and have also been rated excellent in sound by British reviewers (e.g., given an "A" rating for sound quality in HFN/RR). Since some discs are available in essentially identical form in both the U.S. and the U.K., they are given in one list, with discs not generally available through local shops listed separately. The first installment includes records released between mid-1972 and late 1973, and more recent releases will appear in future installments.

Records Generally Available in the USA

Handel: Ballet music from "Alcina" et al. (Marriner), Argo ARG 686
Bruckner: Symphony No. 5 (Haitink), Philips 6700 055 (2 discs)
Dvorak: Symphonic poems & overtures (Kertesz), London 6543
Mozart: Symphony No. 38/Schubert: Symphony No. 8 (Britten), London 6539
Prokofiev: Sonata No. 7/Stravinsky: Petrouchka (Pollini), DG 2530 225
Hindemith: Mathis der Mater (Steinberg), DG 2530 246
Mozart: Piano Concertos Nos. 19 & 23 (Brendel, Marriner), Philips 6500 283
Beethoven: Symphony No. 6 (Jochum), DG 2530 142
Beethoven: Violin Sonatas Op. 30, Nos. 3 and 47 ("Kreutzer") (Menuhin, Kempff), DG 2530 135
Beethoven: Sonatas Nos. 24 & 29 (Brendel), Philips 6500 139
Mozart: Oboe Concerto K.314/R. Strauss: Oboe Concerto (Holliger, de Waart), Philips 6500 174
Shostakovitch: Piano Concerto No. 1 (Ogdon, Marriner), Argo ZRG 674
Beethoven: Symphonies Nos. 1 & 2 (Marriner), Philips 6500 113
Bonporti: Four concertos (I Musici), Philips 6500 182
Liszt: B Minor Sonata (Arrau), Philips 6500 043
Mozart: Symphonies Nos. 25 & 29 (Marriner), Argo ZRG 706
Schubert: Octet (Berlin Philharmonic Octet), Philips 6500 269
J. Strauss II: Polkas & Waltzes (Karajan), DG 2530 027
Carter: Quartets Nos. 1 & 2 (Composers Quartet), Nonesuch 71249, also available on Advent CR-70 cassette
Handel: Water Music, Fireworks Music (Marriner), Argo ZRG 697
Bach: Passacaglia & Fugue in c, BWV 582; Toccata & Fugue in d, BWV 565, et al. (Chorzempa), Philips 6500 214

Schutz: Saint Matthew Passion (Norrington), Argo ZRG 689
Beethoven: Piano Trios Op. 70, Nos. 1 & 2 (Kempff, Szeryng, Fournier), DG 2530 207
Handel: Concerti Grossi Op. 3 (Leppard), Philips 6700 050 (2 discs)
Brahms: Paganini Variations, 4 Ballades, etc. (Earl Wild), Vanguard VCS 10006
Ives: Concord Sonata (Szidon), DG 2530 215
Mendelssohn: Sextet Op. 110, Quartet Op. 3 (Haas et al.), Philips 6500 070
Haydn: Symphonies Nos. 52 & 53 (Marriner), Philips 6500 114
Scriabin: Piano Sonatas (Szidon), DG 2707 053 (2 discs)
Beethoven: Trios Op. 9, Nos. 1 & 3 (Grumiaux et al.), Philips 6500 227
A. Scarlatti: "Endymion" (Grist, Troyanos, et ay, DG ARC 2533 061
J. C. Bach: Six Symphonies Op. 3 (Marriner), Philips 6500 115
Schubert: 12 Piano Sonatas (Haebler), Philips 6741 002 (7 discs)
Monteverdi: Madrigals (Jurgens), DG ARC 2533 087
Schubert:. Symphonies Nos. 1 & 2 (Kertesz), London 6552
Stravinsky: The Rite of Spring (Tilson Thomas), DG 2530 252
Tchaikovsky: Manfred Symphony (Maazel), London 6562
Vaughan-Williams: Tallis Fantasia, et al. (Marriner), Argo ZRG 696.

Records Not Widely Available in the USA

Mozart: Violin Concertos Nos. 3 & 4 (Kantorow, Gulschbauer), Erato STU 07079
Walton: Facade (Marriner), EMI ASD 2786
Brahms: Clarinet Quintet (Michailik et al.), Philips Universo 6580 057
Vaughan-Williams: Symphony No. 2 (Previn), RCA SB 6860
Beethoven's Sketchbooks (Matthews), Discourses ABM 1/3 (3 discs)
Mussorgsky: Bare Mountain/Borodin: Symphony No.. 3 (Lloyd-Jones), Philips Universo 6580 053
Bax: Symphony No. 5 (Leppard), Lyrita SRCS 58
Hoist: Somerset Rhapsody, et al. (Boult), Lyrita SRCS 56
Hovahness: Symphony No. 11 (Hovahness), Unicorn UNS 243
Ireland: Sextet, Cello Sonata, Clarinet Sonata (Melos Ensemble), Lyrita SRCS 59
Stanley: Six Organ Concertos (Jones), Oryx 1742
Bartok: Music for Strings, Percussion, & Celesta (Barenboim), EMI ASD 2670
Ravel: La Valse, Alborado del Gracioso, et al. (Karajan), EMI ASD 2766
Vaughan-Williams: Job (Boult), EMI ASD 2673
Balakirev: Piano Sonata, et al. (Smith), EMI HQS 1259
Mozart: Symphonies Nos. 29, 30, & 34 (Barenboim), EMI ASD 2806
Mozart: Wind Serenades (Czech Philharmonic Ensemble), Supraphon 1081/82 (2 discs)
Mozart: Cosi fan Tutti (Klemperer), EMI SLS 961 (3 discs)
Delius: Paris, et al. (Groves), EMI ASD 2804
Shostakovitch: Symphony No. 12 (Durjan), Philips Universo 6580 012
Alwyn: Quartet, Trio (Gabrieli Quartet), Unicorn UNS 241
Bach: Italian Concerto, et al. (Malcom), EMI SXLP 30141
Mozart: Violin Concerti (Oistrakh), EMI ASD 2839/42 (4 discs)
Bax: Four Tone Poems (Boult), Lyrita SRCS 62
Italian Harpsichord Music (Agana), Philips 802 898
Viennese Music (Boskovsky), Columbia TWO 368.

— PWM

A New Magazine

Sound Advice is a new quarterly magazine of reviews and commentary on perfectionist audio components. The first issue is devoted to amplifiers and phono cartridges. Amplifiers tested are: Audio Research Dual 75, Dual 76, Bozak 929, Crown DC-300A, Dyna 400, Epicure Model One, ESS Eclipse 500A, Ampzilla, Citation 12, Citation 16, Infinity Class D, Accuphase P-300, McIntosh 2300, Paoli 60M, Phase Linear 400, Quatre DLH 100, Quintessence PA II, Sony TAN-8550 (VFET), and Yamaha B1. The cartridges are: ADC-XLM Mk II (to be retested in a later issue), ADC Super XLM Mk II, B&O MMC-6000, Decca V, Decca V Export, Denon DL 103, Denon DL 107, Fidelity Research FR1 Mk II, Grace F8F, MicroAcoustics QDC-le, Ortofon SL-15E Mk II, Satin M15L, Stax, Supex SD-900E, Supex SD-900E Super, Supex SD-901 Super, and the Win Labs cartridge.

The testing procedure depends mainly upon a listening panel but is fascinatingly different from that employed by The Absolute Sound or by the Stereophile. They use double blind A-B comparisons of components into a variety of associated equipment and, in the case of the amplifiers, make A-B comparisons with a piece of wire (literally—needless to say they haven't found the means to do the equivalent with phono cartridges). The method and the terminology are clearly explained at the beginning of the magazine, with discussions of the advantages and shortcomings of the comparison techniques. Said techniques appear to be about as airtight as it is possible to make them, and the tests are repeatable. The results are as interesting as the methods used to get them, especially as regards the amplifiers. They conclude that the Phase 400 is clearly the most accurate of the amps tested and that others have disliked it because it fully reveals disc distortion. Ampzilla, the Dyna 400, the Citations, the Infinity, and the VFETs so beloved elsewhere take a beating. The favored cartridges are the Win Labs, the Denon 103, the Supex 900E Super, and (maybe) the XLM.

I doubt that the magazine will end the fighting, but it does provide a new slant.

— Michael Riggs

Sound Advice is an attractive publication in several respects. For one thing, it does not worship at the feet of High Technology. The "new is better" and "exotic is better" syndromes are endemic in audio, yet the magazine found that the novel Infinity switching amp and the Yamaha and Sony VFET amps sounded less accurate than the plain old Phase 400. While Sound Advice does not worship technology, neither does it ignore technical issues. The reports include some technical data in addition to the subjective evaluations; more important, perhaps, the subjective reviews themselves reflect an alert awareness of how carelessness in technical matters can invalidate a review. For example, the first issue contains an eight-page essay on pitfalls in amplifier evaluation which by itself is worth the magazine's subscription price. This is in sharp contrast to The Absolute Sound, whose totally subjective approach has occasionally led to errors (such as the "veiling" they found in the dbx 119, apparently because of a failure to match levels exactly in A-B testing).

Sound Advice does have its own shortcomings. The amplifier tests relied solely on objective methods of matching levels; it would be interesting to see whether some of the reported sonic differences would disappear if Larry Klein's level-matching procedure were used (see the June Speaker, p. 7). The phono cartridge report did not indicate any awareness of the effect which tonearm damping can have on the sound of high-compliance cartridges. All of the response curves in the cartridge report were printed upside-down, a curious mistake. Perhaps the most annoying aspect of the first issue of Sound Advice is that some of the most widely distributed and familiar products were conspicuously absent (BGW 500 and Phase Linear 700 amplifiers, Shure V-15/III and Audio-Technica cartridges), making it difficult for readers to relate the magazine's findings to the sound of components which we are familiar with. Sound Advice is written in a sober, well-organized, logical style which is superb for conveying information without ambiguity, but it is decidedly less entertaining than The Absolute Sound's personalized and often rhapsodic style.

Indeed it is striking how sharply contrasting (and complementary) Sound Advice and The Absolute Sound are, not only in approach and style but also in geographic perspective and in the conclusions they reach. If you read one, you should read both because of the way they offset and illuminate each other's views. We welcome Sound Advice and recommend it. As we said when we recommended The Absolute Sound in these pages 18 months ago, the value of such a magazine is not that it will tell you what equipment to buy (though it might), but that it stimulates fresh consideration of what characteristics are important in equipment and suggests new ways of hearing and identifying the sonic personalities of components. Sound Advice costs $9 per year for four issues, contains no manufacturers' advertising, and is obtained by subscription from Suite 200, 225 Kearney Street, San Francisco, Calif. 94108. The first issue contained 30 fully packed 8½ - by 11-inch pages. — PWM

Letters

Measuring Speaker Impedance

I believe there is a slight error in the mathematics of Joel Cohen's article on loudspeaker impedance in the June 1975 issue of the Speaker. On page 3, Fig. 2, two voltmeters, V_o and V_r are shown measuring the voltage and current across a network composed of a 1-ohm resistor and a loudspeaker. On page 4 is an equation for deriving the impedance of the loudspeaker as a function of V_o and V_r, namely,

$$\text{Impedance at frequency in question} = \frac{100\ \text{mV}}{V_r} \times V_o - 1\ \text{ohm.} \qquad (1)$$

This equation is not accurate. The basic equation for the magnitude impedance of a network at a given frequency is, from Ohm's law,

$$|Z| = \frac{|V|}{|I|}\,, \qquad (2)$$

where Z is the impedance, V is the voltage across the network, and I is the current through the network. In general all three quantities are complex numbers; that is, they can be thought of as having a magnitude and a phase angle. Equation 2 states that if we divide the magnitude of the voltage by the magnitude of the current, we get the magnitude of the impedance. However, all information about the phase angle is lost from Eq. 2, and this information is necessary if we are going to do any calculations with the network, even subtracting off the effect of a 1-ohm resistor.

Impedances do not add as scalars; they add as vectors and must also be subtracted as vectors. [Ed. note: The letter goes on to show the correct vector method of subtracting using "complex" numbers, assuming that the phase angles are known, and illustrates that by using the incorrect equation one might attribute to a true 4-ohm speaker an impedance of barely over 3 ohms. It also points out a factor-of-ten error in the discussion of method 2.]... While the above mathematics illustrates the proper method of calculating network impedances, it is not necessary for the experimenter to have to deal with it to measure the impedance of his loudspeaker. All that is necessary is to remove the voltmeter V_o from across the amplifier terminals and connect it instead across the loudspeaker terminals, so that it directly measures the voltage across the loudspeaker. V_r will still measure the current through the loudspeaker, so the impedance of the speaker will just be

$$Z_s = \frac{V_o}{V_r} \times 1\ \text{ohm.}$$

Note that with this method, if we are interested only in the magnitude of the loudspeaker impedance, we do not have to deal at all with complex numbers or phase angles. — Mark Davis

Joel Cohen Replies. Mark Davis' comments on method 2 are well taken. It is possible to neglect the realities of complex impedances and vector analysis only if, as in method 1, the circuit impedance is either very much higher or very much lower than that being measured. I actually used a 0.1-ohm resistor for my method 2 testing, but at the last moment I substituted 1.0-ohm in the text to lower the required speaker power level when correctly warned of potential tweeter damage from sustained high-frequency power. The maximum error caused by neglecting complex impedance considerations is the value of the test resistor (1 ohm as illustrated). The factor-of-ten error, of course, resulted from the last-minute change in recommended resistor size. The top line of p. 4 should have said that the impedance in ohms is equal to ten times the output level in volts, minus 1.0; thus a reading of $V_o = 0.9$ volt indicates a speaker impedance of 8 ohms (give or take the 1-ohm uncertainty due to phase angle). Mark's suggestion of measuring V_o across the loudspeaker will give the correct value without the uncertainty, at the cost of somewhat more complicated measurements and calculations since both the speaker and resistor voltages will vary with frequency. —JC

Adjusting Volume Levels

Larry Klein's position that volume levels of A-B'ed components should be adjusted until quality differences are minimized is, I believe, mistaken. Larry's fundamental assumption is that small loudness differences are identified by the ear as differences in quality, not loudness. If this is true, and I suspect it is, then when comparing two pieces of equipment truly having small (but audible) quality differences, Larry's procedure would mask the difference by having us make the inferior piece slightly louder until we produce the illusion (and conclusion) of equal quality.

— Les Leventhal

Ed. note. This is becoming an epistemological inquiry. Suppose that amplifier A is "superior" to amplifier B—whatever that means. Suppose that we compare them and we find that by making amplifier B just 0.2 dB louder than amp A we can make them sound completely indistinguishable. Since B sounds fully as good as A at the same subjective loudness level, in what sense can A be held to be superior? — PWM

Phono Load Capacitance

I have some information and a couple of niggles. In the manual for the KMAL arm, the manufacturer claims a "lead capacity, core to shield," of 80 pF; in their promotional literature, they claim 100 pF. Which claim is correct I do not know, and it is not clear whether the published figures include the arm wiring. I am interrogating the distributor and will report his answers. Some time ago, I asked Harman/Kardon the phono input capacitance of the Citation 11A. I have lost their reply, but I recall that it was either 67 pF or 87 pF.

The niggles are just that. The first has to do with the Speaker's spelling of "Ortofon" several issues ago: the Speaker consistently rendered it "Ortophon." The second concerns the use (or misuse) of the word "subsonic." Not that "subsonic" is strictly improper as audiophiles use it, mind you. It's just that everyone else uses "infrasonic." In the common parlance, "subsonic" refers to speeds below the speed of sound. — Michael Riggs

Ed. note. We plead guilty, never having become accustomed to "infrasonic," a word which has always seemed to have a faintly Pentagonese flavor—like "infrastructure." As for "Ortophon," the responsibility for correcting spelling errors belongs to the Coordinating Editor; members who read the Speaker with a microscope will have noticed that we have a different Coordinating Editor each month in order to avoid burdening any one member with all the work, and some members of our revolving editorial panel have a sharper eye for spelling than do others. — PWM

Mobile FM

I would like to add my voice to Victor Campos' and others requesting that the BAS perform meaningful tests of automobile FM radios. After much frustrating research recently, mainly consisting of a review of available test reports and listening to fixed demonstration units at a variety of dealers, I came to the conclusion that all of the test reports and manufacturers' specifications are worthless or, worse, downright misleading.

I have had six different car FM radios, including two stereo units, over the past ten years. Based on this experience with a variety of equipment I can only further Clark Johnson's comments on "Shop Talk" that traditional parameters of importance for home hi-fi equipment are nearly irrelevant for judging performance in automobiles. In particular distortion, ultimate quieting, power output, and even frequency response are of no importance or, at least, only secondary importance in automotive applications.

I believe that freedom from fading is the most important parameter in judging car radio performance. By fading I mean dropout in relatively strong signal areas, as opposed to lack of signal strength from distant or weak stations. I also believe that overload capability is extremely important. If one is listening to a low or medium strength station and drives by another station, the poorer radio will receive both simultaneously. Antenna attenuator switches are of little use ... their presence suggests a front end with limited dynamic range and susceptibility to overload problems.

The most valuable comparative test of car radios would be an in-car evaluation over a test route selected to provide worst-case multipath and signal-strength variations. Should there be enough interest in the BAS to pursue such a project, I would be willing to donate time, test car, and reference tuner (Pioneer TX-9100) with inverter for A-B comparisons.

Incidentally, a problem I find most disconcerting is the difficulty in listening to wide dynamic range material in a car. The problem suggests the desirability of an adjustable compressor built into the radio. It would be interesting to try a dbx 117 if road tests are undertaken. I would expect that road noise may mask any "breathing" effects even at rather high amounts of compression. — Tom Horrall

Ed. note. Obviously the comparisons suggested by Tom would involve a substantial investment of time and effort, but the result might be much more useful than Audio's recent lab-only tests. Of course, those tests were interesting, especially in documenting the exaggeration and dishonesty in some car stereo advertising. But they did not reveal such common faults as vulnerability to ignition noise and strong-signal overload; my Pioneer KP-300 looks good on paper and in some respects is a delight, but it exhibits both of these faults. The August 1975 Consumer Reports has some useful information on car stereo tape players based on use tests as well as lab data. Tom's suggestion about dynamic-range compression is right; I make cassette recordings using dbx compression (1.2) plus Dolby (which in effect adds further compression at high frequencies); the resulting cassettes sound fine in the car. Dolby FM broadcasts ought to be good in the car (undecoded) for the same reason.

Volunteers for an evaluation project should call Tom or write him c/o Box 7. Incidentally, I suspect that the Advent 400 FM radio, used with an inverter, will run rings around any car radio not only in sheer sound quality and acoustic power output but also in resistance to overload on strong signals and fading on weaker ones. — PWM

Are Some Red Apples Really Green?

It was with great joy that I read in the June Speaker Peter Mitchell's statement that the Koss Pro-4AA headphones are medium-impedance devices. Plugging my pair (bought September 1972) into Presto 800 and Ampex 350 600-ohm line driver outputs resulted in a drastic decline in output signal. Faced with reality, I measured my pair and found one driver slightly below 18 ohms and the other slightly above 18 ohms at 1 kHz. Moral: In spite of the math I was taught, all a's (where "a" = Koss Pro-4AA) are not equal.

And speaking of equal objects not being identical, I would like to relate the following true story. An engineer for a local company designed a circuit using a transistor operating in an area not covered by specifications. The circuit went into production and for over 6 months everything was fine. Then suddenly the production line started producing circuits that failed to work. Investigation revealed that the transistor manufacturer had changed the process used to make the transistor and it no longer worked in this specific circuit. The 741 op-amp circuit curves given on page 5 of Peter Mitchell's article represent operation in an area unspecified by the manufacturers. It is possible and probable that one or more brands and/or production lots will fail to drive 250-ohm loads in a low distortion mode. Moral: You cannot hold the manufacturer responsible for anything not explicitly stated in the specification sheet of that manufacturer.

— Keith North

Ed. note: Keith has a good point. I can't rule out the possibility that some 741-type op amps may not be completely suitable for the headphone amp. But I think that the chances of success are pretty good. My prototype of the headphone amp actually contains a Signetics 5558 IC, sold as a "dual 741" by Radio Shack, and it works fine driving Pro-4AA's when I substitute 5558's from another source (PolyPaks) or Motorola 1458's. Laurie Cote built one using two 741 mini-DIP's from Radio Shack, of unknown manufacture, and it drives his Pro-4AA's well. And I have learned from an old issue of Electronics that an engineer had discovered that Fairchild TO-8 741's produce maximum power into 270 ohms.

But Keith's measurement of an 18-ohm impedance in his Pro-4AA is very surprising. Howard Souther of Koss assured us when he was here that the original Pro-4 had a 50-ohm impedance, increased to 100 ohms in the Pro-4A, and to 250 ohms in the Pro-4AA. If you are concerned, check the dc voice-coil resistance of your own phones with an ohmmeter. — PWM

Addendum to Allison : One Review

Because of space restrictions, many items were cut from individual reviews of the Allison: One that appeared last month. One paragraph cut from Peter Mitchell's and which may be of general interest is reproduced below. As is often necessary in the preparation of such reports, Peter's material had to be cut somewhat and rearranged. For those interested in reading his entire review, Peter offers to send a copy of his original manuscript in response to a self-addressed stamped envelope to Box 7. — Jim Brinton

"It should be noted that I used a Phase Linear 700 amp throughout my evaluation (with, in most cases, no preamp—the tape deck was connected through a dbx 117 directly to the Phase). Some writers have suggested that the 700 may not be an ideal mate for the AR-3A and LST, possibly because of the low and complex reactive impedance which they present to the amp. I cannot rule out the possibility that if they had been driven by a BGW or Dunlap-Clarke amp, the LST and 3A might have fared better in the comparisons. The Allison, though low in efficiency, appears to be an 'easier' load to drive; in any case the 700 drove it beautifully. While the LST and 3A take on an irritatingly 'loud' quality at sustained levels above about 95 dB SPL, I found that I could play the Allisons a full 10 dB louder with full pleasure and without listening fatigue. Whether this is due to the amplifier's behavior or to the smoothness and low distortion of the Allisons, I cannot say." — PWM

Some Interesting Test Data on Recording Tape

Jim Richardson and David Satz have called to our attention a very interesting test report on open-reel recording tapes which was prepared by British recording engineer Angus McKenzie and published in the February 1975 issue of Studio Sound. The tests, including some not usually performed, appear to have been done intelligently and carefully, and they produced some important and surprising results. Since Al Foster's test report on tapes was one of the most popular articles that we published during the BAS Speaker's first year, we presume that there *is* a continuing interest in the relative qualitites of recording tapes. So in the accompanying table we reprint a selection of the data from Studio Sound.

The items in the table are as follows.

1. Optimum bias current, expressed in dB relative to an average; found by adjusting the bias to maximize the record/playback output at 10 kHz and then increasing the bias until the playback output at 10 kHz drops by 4 dB.
2. Sensitivity at 1000 Hz, i.e., the playback level resulting from a 0 VU recording level.
3. Frequency response at 15 kHz, corrected for differences in sensitivity at 1000 Hz; measured using the same bias current for all tapes.
4. MRL (maximum recording level) at 1000 Hz relative to the NAB 0 VU level; specifically, the signal level which produces 3% third harmonic distortion.
5. THD at +4.7 VU; this odd-looking level is DIN 0 VU.
6. MRL at 10 kHz relative to NAB 0 VU; the level at which high-frequency saturation sets in, specifically the level at which the measured IM distortion at 10 kHz reaches 10%.
7. The CCIR-weighted tape hiss level relative to NAB 0 VU; the use of A-weighting would produce numbers a few dB better in each case than the CCIR-weighting.
8. The useful dynamic range of the tape; it is the difference of the preceding two' columns, chosen on the assumption that high-frequency saturation due to recording pre-emphasis is what limits the maximum useful recording level in practice.
9. The print-through due to a 1000-Hz tone recorded at the 1000-Hz MRL followed by a three-day storage time; compare with the "dynamic range" column to judge the audibility of the print-through over the background of tape hiss.
10. The modulation noise, relative to the MRL at 1000 Hz; the threshold of audibility is believed to be about -40 dB. The -51 dB figure given for TDK Audua is very impressive if correct, but one wonders whether it might be a misprint for -41 dB; there were several other definite misprints in the Studio Sound table which are corrected in the table printed here.
11. "Stability," a measure of the smoothness of the oxide surface; a 15-kHz tone is recorded, and on playback it exhibits very rapid amplitude variations. The rating expresses the severity of this fuzz: poor, fair, average, good, very good, or excellent.
12. Dropout, i.e., the incidence of larger and longer amplitude variations in playback, rated as above.
13. Winding smoothness in fast wind on a high-quality transport; poor wind raises the risk of tape damage and accelerated dropouts.

Tape Type	Bias (Relative), dB	1000-Hz Sensitivity, dB	15-kHz Response, dB	1000-Hz MRL (3%Distortion), dB	THD at +4.7 VU, %	10-kHz MRL re 10%1M), dB	Hiss re NAB 0 VU, dB	Dynamic Range, dB	Print-through re MRL at 1 kHz, dB	Modulation Noise, dB	Stability	Drop-out	Wind
Ampex 406	-0.5	+0.5	+1.5	+11.2	0.60	+10.7	-49.0	59.7	-64.5	-40.5	F	G	VG
Ampex 407	-0.8	+1.0	+1.0	+11.5	0.50	+11.0	-48.3	59.3	-63.5	-39.5	G	VG	F
Ampex 9472 GrandMaster	-0.2	+3.0	+1.3	+17.5	0.10	+12.7	-49.5	62.2	-60.0	-43.5	F	G	F
BASF LP35LH	+0.3	-1.3	+2.5	+11.5	0.55	+11.7	-51.5	63.5	-65.5	-41.5	A	F	F
Maxell UD50	+0.3	0.0	+2.5	+12.0	0.45	+11.7	-50.6	62.3	-68.5	-41.5	F	A	P
Maxell UD35	+0.3	+0.3	+2.5	+12.0	0.40	+11.7	-49.8	61.5	-67.5	-42.5	A	G	P
Memorex 1.5 mil	-1.0	+0.3	+2.8	+10.7	0.65	+12.0	-50.0	62.0	-66.0	-41.0	F	A	P
Memorex 1.0 mil	-0.7	0.0	+2.0	+10.5	0.75	+11.7	-50.1	61.8	-66.0	-41.0	A	F	P
3M 206	0.0	+0.8	-0.5	+12.2	0.45	+9.5	-51.3	60.8	-62.5	-40.5	P	P	G
3M 207*	+0.3	+0.3	+0.3	+11.7	0.55	+9.7	-51.3	61.0	-58.0	-41.0	A	G	P
3M 209	-0.7	-0.5	+0.3	+9.7	0.85	+9.5	-51.3	60.8	-62.5	-40.0	A	G	VG
3M 250	+1.5	+2.0	+0.1	+16.2	0.10	+11.7	-51.8	63.5	-54.0	-36.0	F	P	G
3M Classic 1.5 mil	+0.3	+0.8	+2.3	+12.5	0.45	+12.2	-51.8	64.0	-54.5	-40.5	P	A	G
3M Classic 1.0 mil	+0.5	+0.8	+2.3	+12.7	0.45	+12.2	-52.0	64.2	-53.0	-43.5	A	F	P
TDK Audua	+0.5	0.0	+3.0	+12.7	0.40	+12.2	-50.6	62.8	-63.0	-51.0	VG	G	P

*Identified as 307 in the report, presumably a misprint.

Summary. In some respects these tapes are quite competitive with each other. Perhaps the most important news in this table is the discovery that the excellent dynamic range of Scotch Classic tape is obtained at the price of rather bad print-through. There are two tapes in the group which have astonishingly high midrange overload levels (Ampex 9472 and Scotch 250), but few recorders contain recording and playback preamps which could accommodate such high levels. Apparently the best tapes in the list (if you can afford them and can adjust your recorder to use them correctly) are TDK Audua, Maxell UD, and BASF LP35LH, the same tapes which have been recommended previously in these pages by Al Foster.

Incidentally, Studio Sound is a monthly magazine for broadcast and recording engineers. It is available by subscription from the publishers at Link House, Dingwall Avenue, Croydon CR9 2TA, England. At current exchange rates a subscription costs about $10 per year, and it is free to people involved professionally in music or recording. — PWM

In the Literature

Acoustical Society of America, Journal of the, June 1975, Part I

- This issue, long in the offing, is a special devoted to the work and students of Frederick V. Hunt, a professor of physics at Harvard, who specialized in diverse audio matters. Definitely worth the trouble of searching out in a local university library.
- Acoustics and the Concert Hall: by L.L. Beranek, one of Hunt's students. Short article with some history and some science, beginning with Sabine's design of Symphony Hall (he was also at Harvard) and mentioning the formation of B.B.&N . (pp. 1258-1262)
- Behavior of Sound in Bounded Space: Theory (pretty lengthy) about sound in a room (tone bursts and spectral analysis). Note the conclusion on p. 1290 that objects in a hall that give a spurious reflection to the listener's ear are annoying and will show up in analysis of tone burst data. (pp. 1275-1291)
- The Acoustical Qualities of Concert Halls: Psychoacoustical evaluation of halls with a rating scheme more complex than Beranek's. Interesting, but hard to apply to one's audio life. (pp. 1292-1299)
- Effective Length of Horns: Another paper from B.B.&N. Describes the mathematical "length" of a brass instrument's horn. (p. 1309)
- F. V. Hunt and the Disc Recording Arts: About Hunt's work, this is by Ben Bauer of CBS and is a brief history of "low" tracking force pickups (5 gm to 0.1 gm). (pp. 1327-1331)
- Theory of Groove Deformation in Phonograph Records: Another from CBS, this one modern theory and a comparison with experiment, emphasizing the interaction between stylus tip and vinyl surface. (pp. 1332-1340)

Acoustical Society of America, Journal of the, June 1975, Part II

- Optimal Acoustical Design of Sandwich Panels: Panels for sound absorption, perhaps of use for apartment dwellers (p. 1481).

Audio, Aug. 1975

- Two good articles on loudspeakers, one a review of Thiele's landmark review of electronically equalized vented loudspeaker design (e.g., the Electro-Voice Interface-A) and the second a short review of motional-feedback-controlled speakers, which just may be the "acoustic suspension" revolution of this decade. Equipment reviews include the Yamaha B-1 VFET amplifier and the B&O cassette deck. Mention is made of the dbx-Sheffield Volume 4, and a full page is devoted to TIM in the letters section. Finally, Giovanelli answers some of the most stupid questions (yes, there are stupid questions) in the history of Audio; where does he get these, at the GE appliance center ?

db, July 1975

- Architectural Acoustics, Part II: Noise control for auditoriums. Points out that electronically delayed signals are sent to loudspeakers at the rear of some halls (e.g., Royal Festival Hall in London). (p. 33)
- Handy Black Boxes: Variety of devices (mike line tester, ground checker, cable tester) useful to audio recordists. Written by Don Davis, a well known audio author and instructor. (p. 28)
- For those not familiar with db, this monthly is a commercial publication of interest to recording engineers. There are occasional articles useful to audiophiles, plus regular columns (The Sync Trac by John Woram, who is VP of the Eastern Region of AES, and Theory and Practice by Norman Crowhurst). Subscriptions are $6/year from Sagamore Publishing Co., 980 Old Country Rd., Plainview, N.Y. 11803.

Electronic Design, May 24, 1975

- CMOS Audio Amplifier Features 115 dB Bass, Treble Control Range. (p. 98)

Electronic Engineering Times, July 14, 1975

- John Fink reviews the Audio Craft AC-300 damped tonearm and finds it not suitable for high compliance cartridges. This is a $200 Japanese viscous-damped unipivot design with a thumbscrew to squash grease more tightly into the pivot cup to control damping. Fink finds the damping setting critical for best sound quality. (p. 45)

Electronic Products, July 1975

- Are You Making the Right Connection: Article on choosing connectors by Switchcraft's chief engineer.

Electronic Servicing, July 1975

- Principles of Video Tape Recorders, Part II: For the videophiles among us. (p. 30)

Electronics, April 17, 1975

- Quaking Sensation Comes to the Movies: Description of Earthquake sound track. (p. 34)

Electronics, June 26, 1975

- Solid State Power: New high power devices, including Nippon Electric's 200-watt (200-volt, 10-amp) VFET. (p. 81)

Radio-Electronics, Aug. 1975

- Looking Ahead: Sony TV projection system (12-inch Trinitron and projector) and Muntz 30- by 40-inch system again announced. (p. 4)
- Reviews of the Kenwood 5400 receiver (perhaps a good gift to nonaudiophile friends) and the Crown VFX-2 electronic crossover. (pp. 36, 39)
- Inside Today's Tape Transports. Emphasis on the electronic motor control section of the Crown 800, with schematic. (p. 45)
- All About Oscilloscopes, Part III: Mostly on triggering, which has always been an operational mystery to me. (p. 52)

Recording Engineer Reproducer, June 1975

- A freebie magazine to those in the industry. Contains a review article on broadcasting, very general in scope, following the signal through the sources, studio, and electronics in the pre-transmitter signal path (limiters, quad matrixers, and broadcast distortion specs).
- Spectrum Analysis Applied to Audio System Diagnostics: Expensive gear (the HP 3580A) that does a lot (e.g., see the Dyna meeting summary).
- We will not publish subscription information for this journal, so you must find a professional friend who has access to it.

July BAS Meeting

Over 70 members gathered on July 29 for a busy meeting in which both the room and the agenda were fully packed. Peter Mitchell reported that the BAS oscillator kit was not available due to Lafayette Radio's extreme lateness in supplying promised parts; he also discussed packaging options for the 814 microphone capsule (see note elsewhere in this issue).

The first feature of the meeting was a discussion and demonstration of the developmental prototype of a device for synthesizing realistic concert-hall reverberation at home. The original purpose of quadriphonic sound, of course, was to recreate in the home listening room the acoustic ambience of the concert hall surrounding the listener. But since this goal is not being met by commercial quad recordings, the best alternative is a new kind of audio component: a device which will accept two stereo channels of information and will feed to two or more side/rear speakers the multiply time-delayed signals required for a convincing re-creation of the ambience of a large acoustic space. (For background, see the August 1974 BAS Speaker, the Sound Advice supplement to the Oct. 22, 1974, Boston Phoenix, or the October-December 1974 issues of the WBUR Folio.) Hybrid Systems of Burlington, Massachusetts, is developing such a device, and it was described and demonstrated by Sam Walinsky and Richard DeFritas.

The incoming audio signal in each channel is converted to digital form using a "delta-sigma modulator" which overcomes some of the cost-versus-quality limitations of ordinary binary analog-to-digital converters. The digital signal is then time-delayed by translation through shift registers having 24,000 bits per channel and is finally converted back to an audio signal. Since a simple time delay like this cannot provide a convincing simulation of real concert-hall reverberation, the Hybrid Systems device incorporates a more complex mode of operation in which the two delayed signals are filtered, partially cross-mixed, and multiply recycled through the delay process so that the final audio output contains a matrix of multiply-delayed signals whose amplitude decays away over a period of nearly a full second—rather like the reverberation of an actual large hall.

In the experimental prototype, various additional options are switch-selectable, including the lengths of the time delays, the amount of high-frequency rolloff, and the re-insertion of synthesized reverberation into the front channels of the four-speaker array. This last option does not appear to contribute to realism. The other options were demonstrated at length using four Smaller Advents driven by a Phase Linear 700 in front and an AR amp in the back. In a BSO broadcast tape of Stravinsky's "Firebird" the preferred option was for the longer delay times with the multiple recycling. But on spoken voice the synthesized reverberation sounded grotesque, indicating that lesser amounts of reverb are desirable with solo material (not surprisingly).

The demonstration of the Hybrid Systems prototype engendered a good deal of interest and discussion, and we anticipate opportunities for extended at-home evaluation of later prototypes. If the device is marketed, it is expected to retail for under $400, not much more than one might expect to pay for the simpler time-delay units being developed.

For comparison, at the end of the meeting Joel Cohen briefly demonstrated a prototype of a time-delay system being developed by a few BAS members. It employs the Matsushita analog bucket-brigade IC which is finally in production. Since Joel's experimental unit was mono and limited to a maximum delay of about 50 milliseconds, it sounded much less spectacular than the previous unit; yet some listeners felt that it sounded more natural. In any case the quality of the delayed signal is impressively free of response aberrations, distortion, or noise, making the prospect of further experiments with these IC's quite attractive. It is hoped that we may be able to develop a BAS kit time-delay unit for $150 or less.

The central part of the meeting was I begun by David Satz, who is a musician, semi-pro recordist, and teaching assistant at the New England Conservatory. He discussed some facets of the relationship between music and recordings, stating at the outset the thesis that when you buy a record of a piece of music you really are not getting a recording of the music. Rather you are getting a recording of a single performance of the music—a trivial-sounding but crucial distinction because even in an ideal case a performance can represent a crystallization of only some of the possibilities inherent in a composer's score. And in the real case, because of the nature of the commercial recording business, the performance you get may have only the remotest relation to what the composer envisioned. Since many aspects of performance are not fully spelled out in scores, the performing musician needs the help of a historical and musical perspective to judge how to deal with the many things which are not specified. This perspective is often lacking, and "tradition" and "common practice" are inadequate, often seriously misleading, substitutes. A gross example is the nearly universal practice of playing Beethoven's music much slower than his metronome markings specify, and various excuses are invented to justify the habit. Yet when rare performances have been given at the specified tempos they have demonstrated that Beethoven knew what he was doing. When so played the music acquires a very different character than the bombastic Beethoven we usually hear, and its musical validity is apparent to non-musicologists as well as to partisans in the tempo debate. As another example, David noted that there is a point in the Beethoven Violin Concerto where performers always slow down and later speed up again, though such a retard is neither specified in the score nor justified by musical sense; the corruption probably was the invention of some Russian violin teacher who trained many of this century's virtuosi, and having learned it they keep passing it on. Satz said that when the passage is played correctly as written it clearly is one of those inspired Beethoven moments that seem overwhelming in their inevitability—a moment we have been cheated of.

Yet true fidelity to the composer's score, even if we could find it in commercially recorded performances, does not guarantee that a recording will let us hear what the composer envisioned. The score is an abstraction, and expressivity in music (as in stage drama) depends not on the notes (words) but on their phrasing. Satz illustrated the crucial importance of musical phrasing by playing excerpts from two recordings of Mahler's 5th Symphony: Solti's performance with the Chicago orchestra (of all recordings the one most scrupulously faithful to Mahler's instructions in the score), and Mahler's own performance as recorded on a Welte piano roll. David suggested that the Solti performance is a stunning display of orchestral virtuosity, but Mahler's spirit is missing. He stressed that musical phrasing should illuminate the hierarchical structure of a symphony and communicate the structural meaning of each passage; in Mahler's own phrasing one hears the implications of all that has come before and all that will follow in the symphony. In the comparison it was clear that, despite the dynamic and tonal restrictions inherent in the piano reduction of a richly orchestrated symphony, Mahler's piano-roll performance is a far richer and more unified musical experience than the Chicago recording. Mahler suspends the laws of strict rhythm. His tempos flow imperceptibly from one speed to the next; his phrasing, alternately hesitant and impulsive (but involving conscious rhythmic displacements too small to measure accurately) lends the music the expressive quality of anguished human speech. Solti's performance, as impressive as it is on first hearing, seems cold by comparison.

Following an intermission, another aspect of the relationship between recordings and music was explored by David Ranada, a music student at Harvard and an assistant catalogist at Schwann, currently working on a new Artist Issue to replace the edition last published in 1970. He summarized his central proposition in two points.

1) The main elements of a performance (dynamics, pitch, tempo, phrasing, articulation, rhythm, instrumental balance, and tone quality) are little affected by the quality of the reproducing equipment—once a minimal quality level has been achieved, of course. Only the tone quality, instrumental balance, and dynamics can be substantially altered by either the recording producer or the audiophile. David discussed this point in greater detail in his previous talk to the BAS a year ago.

2) Recorded performances often do not reflect the intentions of the composer. In order to know and judge a piece of music one must listen carefully to many different performances of it, or read the score—preferably both. The concept of the "definitive" recording is inherently absurd, and the record collector should make a point of obtaining diverse performances to compare and learn from. The thesis that further improvements in our playback systems can lead us closer to the music is without basis in the facts of musical life, and so the audiophile's quest is not a musical one. Full musical understanding and appreciation require a more attentive ear and the development of a good musical memory rather than the acquisition of more extensive, expensive, or exotic equipment. An audiophile who has invested $1500 in equipment and only $300 in records is a victim of distorted priorities. A medium-grade audio system is adequate to let us hear the essential musical differences among recorded performances—if we really are listening to the music and not just bathing in the sound. And when we undertake to recommend records for their musical merit (rather than simply as sonic demonstrations), we are on very shaky ground unless we know all of the competing versions well enough to characterize their differences.

Having thrown down the gauntlet, Ranada gave members an opportunity to test their musical ears on comparative recordings of well-known works. In the first example, the "Danse Sacrale" of Stravinsky's "Rite of Spring," a gong plainly heard in Mahta's recording was absent from Stravinsky's own disc—not because of a recording fault but because the composer re-orchestrated the music in 1947. Next, in Debussy's "La Mer," trumpet and horn figurations were audible in the Boulez performance and not in Froment's (from a VoxBox), again representing the composer's revised and original orchestrations, respectively. David next compared two recordings of the final movement of the Brahms First: Walter doing a now-traditional slowdown midway in the movement, Swarovsky achieving a quite different effect by playing the music as written. David pointed out that if we don't read scores, the only way we will learn about such possibilities is to listen to diverse performances. If we buy just one recording of a work and assume that it conveys everything in the music that the composer intended, we will miss a great deal of musical satisfaction especially in works subject to corrupt performing traditions. As a further illustration, in the finale of Beethoven's "Eroica," most recorded performances have trumpets blaring the restatement of the theme, while in the score and in the Leibowitz recording the strings and winds carry the music far more attractively. (The Leibowitz performance is from a set in which he attempted to conduct all of the Beethoven symphonies according to the score, free of encrusted tradition. The set is available at a budget price from Reader's Digest.) Another Romantic tradition was illustrated in the scherzo of Beethoven's Ninth: galumphing French horns doubling the string parts in most performances, while Solti's and Leibowitz's achieve a lighter sound free of awkwardness by playing what Beethoven actually wrote. David concluded with assorted examples of unauthorized aforzandi and tempo retards, the comparisons indicating in each case that the performances which more accurately reflect the score seem to be more satisfying to the ear as well. Evidently those composers know what they were about.

At the conclusion of the meeting, as source material for Joel Cohen's brief time-delay demonstration, David Satz played excerpts from his superb on-location tapes of Bach cantatas performed weekly during Sunday services at Emmanuel Church on Newbury Street. — PWM

A Transformerless Balanced-Line Preamp for the Phantom 814 Microphone

Peter W. Mitchell

This article continues and concludes the experiments reported in last month's "814 Column." The discovery of the mild impedance mismatch between the 814 mike and the Advent MPR-1 preamp suggested the possibility of designing a preamp having an input impedance better matched to the 814. Normally, trying to design a preamp to replace the MPR-1 would be a waste of time due to the difficulty of equalling the MPR-1's low input noise level in another design. However, I have found the combination of the 814 and the MPR-1 satisfactorily quiet in live recording despite the fact that the MPR-1's input impedance attenuates the 814's signal by 10 dB. This means that a new preamp, if it had a high impedance which would not load down the 814, could be up to 10 dB noisier than the MPR-1 and still produce as good a signal-to-noise ratio in practice with the 814 mike.

To explore this possibility I set up a calibrated 400-Hz, 94-dB SPL sound source and measured the sensitivity of some microphones, both while they were driving an MPR-1 and while they were driving a high-impedance preamp of identical gain (40 dB). Given the voltage which each mike delivered into the preamps at 94 dB SPL, subtraction of 70 dB yielded the preamp noise level which would be equivalent to a hall background sound level of 24 dB SPL with that mike. For reference, a typical mike preamp has an input noise level of about 1 microvolt (-120 dBV), and the MPR-1's specified input noise level is -128 dBV. The figure of 24 dB SPL was chosen because it is the specified equivalent noise of the 814 capsule, which I have found amply quiet for live concert recording, since in monitoring the live mike feed via headphones, what one hears is the hall ambience rather than the mike hiss.

The results, listed in Table 1, are quite interesting. In view of the relatively low output of ordinary dynamic mikes, most mike preamps are not quiet enough for perfectionist recording; even the MPR-1, whose noise is close to the theoretical limit, is barely quiet enough. The battery-powered 814, with a high-impedance preamp, produced an output level of -49 dBV, within 1 dB of the sensitivity specified by Thermo-Electron. Into the MPR-1 the 814's signal level drops 10 dB, as predicted on the basis of last month's impedance measurements, with the result that the MPR-1's noise level is just low enough to correspond to the desired 24 dB SPL.

The startling measurement is that of the phantom 814. Into the MPR-1 its signal level is 9 dB higher than that of the battery-powered 814, and when feeding a high-impedance preamp the phantom 814's output rises an additional 14 dB! In other words the phantom 814 feeding a high-impedance load has 23 dB more output than the battery-powered 814 used with the MPR-1. With so high an incoming signal level, a preamp would not need to be extraordinarily quiet to deliver an excellent signal-to-noise ratio with this mike. However, the preamp will have to have an impedance of several thousand ohms in order not to load the phantom 814 down; I measured the impedance of the phantom 814 mike and it turned out to be 2600 ohms.

Copyright © 1976 Peter W. Mitchell

Table 1

Mike	Preamp	Signal at 94 dB SPL, dBV	Desired Preamp Noise (24 dB SPL), dBV
Typical dynamic	High-Z	-56	-126
Typical dynamic	MPR-1	-60	-130
Battery-powered 814	High-Z	-49	-119
Battery-powered 814	MPR-1	-59	-129
Phantom-powered 814	High-Z	-36	-106
Phantom-powered 814	MPR-1	-50	-120

Last month we noted that the phantom-powering circuit for the 814 altered the behavior of the FET in the capsule, raising its current drain. Apparently we have here another manifestation of that change: the FET's gain is increased in the phantom-powered mode, yielding the high output signal measured above. The change evidently is due to the placement of the 4700-ohm resistor in the source circuit rather than in the drain circuit of the FET (see the March Speaker for details). It might reasonably be supposed that the higher gain would cause the capsule to be more susceptible to overload due to loud sounds, but this turns out not to be the case. I tested for overload using a 400-Hz sine-wave signal. At 125 dB SPL the waveform observed on an oscilloscope was still a sine wave. At 130 dB SPL mild distortion was visible, though I could not be certain whether the distortion was due to the mike capsule or to the loudspeaker which was producing the test tone (it did not sound quite clean). Actually, in fact, the overload levels in these capsules can be predicted on the basis of dc voltage measurements across the FET and its associated resistor; the predicted overload level is 127 dB SPL for the phantom 814, compared to 122 dB for the battery-powered 814 and 140 dB for the battery-powered 814C.

Since the phantom 814 seems to work fine despite its magnified output sensitivity (and sounds OK in casual at-home tests, monitoring via headphones), there is no visible barrier to designing a preamp for it, one having a high enough input impedance so as not to load down the mike. The first challenge is to eliminate the input transformer.

In order correctly to design a transformerless mike preamp we must understand why a professional unit normally employs an input transformer. Stated simply, there are two reasons: (1) to convert from "balanced" to "unbalanced" line and (2) to step up the signal voltage to overcome the input noise of the circuit. The second of these is not crucial to us since we have a high signal level coming out of the mike. But the first requires a close look.

In an "unbalanced" circuit all voltages are measured relative to the system ground, as is done in all ordinary consumer audio gear for instance. Thus ordinary audio cables have a single conductor in the center, with the grounded cable shield serving as the signal-current return path. In a "balanced" line, on the other hand, there are two conductors surrounded by a grounded shield; the undesired audio signal is carried as the voltage difference between the two conductors, which have a floating or undefined potential relative to ground. Now, perfect shielding in a shielded cable is impossible, so an unbalanaced line can pick up hum or radio interference (especially in live recording where mikes must be 40 feet or more from the recorder). You can demonstrate this for yourself just by passing your phono signal cables close to an amplifier's power transformer or running them along an ac line cord. If a balanced line passes through an ac hum field or an RF field, the hum or RF voltages will be induced identically on both signal conductors relative to ground. This is where the input transformer does its thing: it is wired so that it passes the voltage difference between the two conductors (the desired audio signal) but rejects any "common-mode" signal appearing identically on both conductors (see Fig. 1). This property of rejecting hum and RFI picked up in mike lines is why balanced lines and transformers are considered imperative for serious recording.

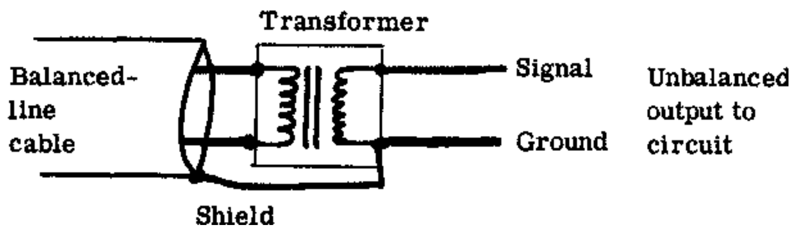

Figure 1

So in order to make a transformerless preamp we want a high-quality "differential" amplifier circuit with good rejection of common-mode signals. Many readers will immediately recognize this as a description of an operational amplifier (see "IC Op Amps," September 1974 Speaker). An IC op amp is inherently a differential amp, though for audio use we usually ground one of its inputs, and good IC op amps also have excellent common-mode rejection. Normally, of course, an IC op amp would not be quiet enough to serve well as a mike preamp; cheap op amp IC's commonly exhibit an input noise level of about -105 dBV, and the better op amps achieve -115 dBV. The LM381AN IC can achieve -120 dBV or better, but it is not an op amp and apparently cannot be wired for balanced-line input without a transformer.

Since the phantom 814 has such a high output signal level, the input noise of an op amp IC actually is low enough to be acceptable, and since a differential-input op amp can be used for balanced-line operation with good common-mode rejection, an IC op amp can be the heart of a successful preamp for the phantom 814. Figure 2 shows the basic schematic of: the circuit. The

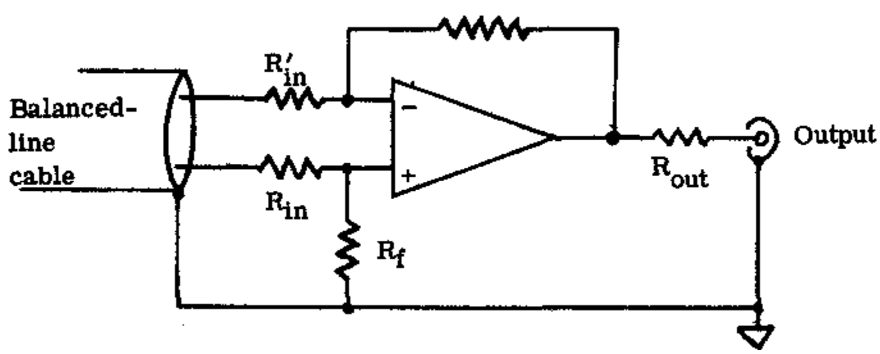

Figure 2

input resistors R_{in} and R'_{in} are matched in value; similarly R_f and R'_f are a matched pair. R_{out} can be any moderately small resistor, such as 1000 ohms. The effective input impedance of the circuit is twice R_{in}, and the differential signal gain of the preamp is $G = R_f/R_{in}$.

The rejection of interfering common-mode signals depends on how accurately the pairs of resistors are matched in value. According to Jung (IC Op-Amp Cookbook, Sams 20969, p. 18) the voltage gain of the differential amp for common-mode signals is given by the expression

$$G_{cm} = \frac{R_f R'_{in} - R_{in} R'_f}{R'_{in}(R_{in} + R_f)}$$

This can be simplified and converted to decibels, yielding

$$G_{cm}\,(\mathrm{dB}) = 20 \log (P) - 34\ \mathrm{dB},$$

where P is the percentage error in the matching of the resistor pairs. So if the resistor pairs are matched within 5%, the common-mode gain is equal to -20 dB. Then if we make the amp's differential-input gain equal to +20 dB, the common-mode interference is attenuated a total of 40 dB below the desired audio signal. This is ample for even relatively difficult recording situations; however, if you want to, you could use a resistance bridge to match resistors to within 1%, pushing the preamp's common-mode gain down to -34 dB and thus increasing the common-mode rejection to 54 dB below the desired audio signal.

We have mentioned a differential gain of 20 dB for the preamp. That is a convenient value given the phantom 814's sensitivity. To accommodate very soft or very loud sound sources, you may wish to provide switchable gain by altering the matched pairs of resistors R_1 and R'_f. To accomplish this change simultaneously in two stereo channels, a four-pole switch is required, and the Lafayette three-position rotary switch (no. 99-61566) serves nicely. Table 2 suggests appropriate resistor values providing 10-dB increments in gain, assuming that the input resistors are 3300 ohms (appropriate in view of the impedance of the phantom 814 mike). The table also lists the corresponding preamp overload with the phantom 814, based upon a preamp output clipping level of 5 volts rms.

Table 2

Gain, dB	R_f,R'_f	SPL at Clipping, dB
10	10K	134
20	33K	124
30	100K	114

We have neglected just one item. This preamp is designed to work only with the phantom 814, so we must provide a positive dc feed up the signal cable to turn the mike on. Figure 3 shows how

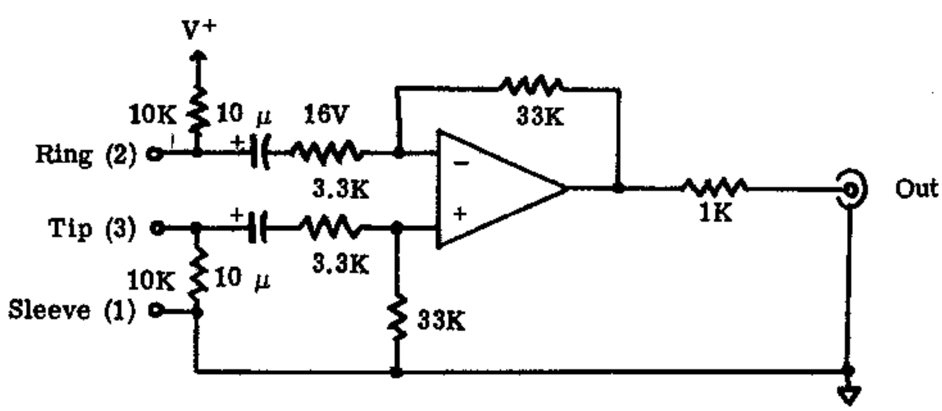

Figure 3

a

this is done. The 10K resistor isolates the V^+ dc supply from the signal cable so as not to short out the audio signal (remembering that a dc supply is an ac short circuit). A similar 10K resistor then goes from the other signal lead to ground in order to preserve the symmetry of the balanced line, and of course coupling capacitors are required to isolate the dc mike voltages from the op amp inputs because the IC's dc gain is as great as its audio gain. Incidentally, this arrangement is not a true "phantom" supply since dc is applied to only one side of the balanced line; but this approach provides better polarizing voltages across the capacitors in the phantom 814 mike. However, the polarity of capacitor C3 in the mike should be reversed to place its positive end at the B+ terminal of the capsule; Figure 4 shows the revised wiring of the phantom 814 mike. The

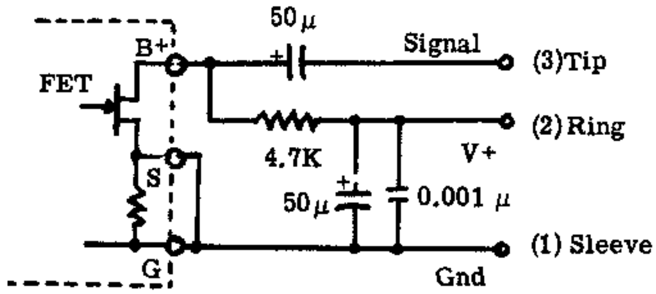

Figure 4

zener diode in Alan Southwick's design can be omitted so long as the mike will be used only with this preamp. The pin numbering here and in the preamp schematic must be scrupulously observed; so must the phasing of extension cables.

Another useful option now presents itself. The input coupling capacitors in the preamp combine with the input resistors to form a low-cut filter. For wide-range recording, the cutoff is placed at a subsonic frequency, of course, but with the aid of a switch we can change the capacitors to introduce a gentle bass cut when desired. For instance, if you have to place the mikes on stands rather than hanging them, a bass cut can minimize the floor rumble which will be picked up by a mike with a truly flat low end. This option is particularly handy when recording music which has little or no deep bass energy of its own. Also, when recording the spoken voice, a judicious bass cut nearly always provides a more natural-sounding result. A second Lafayette no. 99-61566 switch will permit switching both capacitors in each of two channels; suggested capacitor values for the three positions are 10 µF, 1 µF, and 0.2 µF.

The selection of the IC op amp is up to you. The 741 can be used, up to a maximum gain of 20 dB. I chose the LM301A, which has lately become nearly as inexpensive as the 741; it permits gains to 30 dB and is slightly quieter. So the final, full-fledged preamp for the phantom 814 is shown in Fig. 5. The input jacks may be either Switchcraft/Cannon connectors or three-circuit phone jacks (Radio Shack no. 274-312), depending on how you choose to wire your mike cables. The total parts cost for the preamp is about $20, and the power supply is a pair of 9-volt batteries.

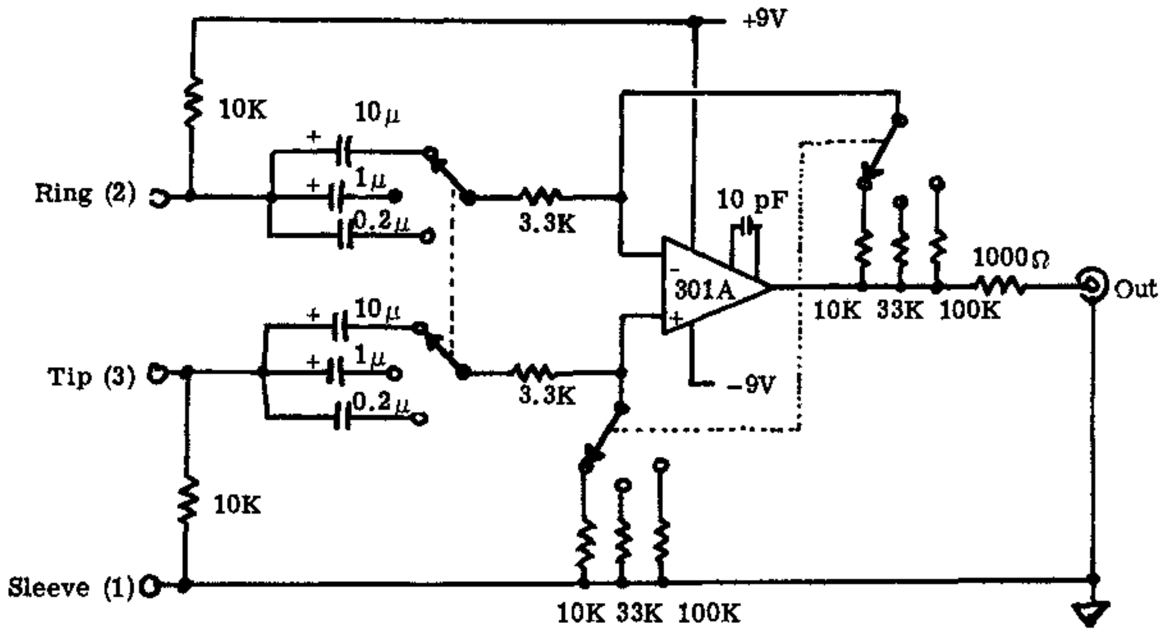
+9V
10K
10µ
Ring (2)
1µ
0.2µ
3.3K
10 pF
301A
-9V
1000Ω
Out
10K 33K 100K
Tip (3)
10µ
1µ
0.2µ
3.3K
10K
Sleeve (1)
10K 33K 100K

Figure 5

a

Using the BAS Oscillator

Peter W. Mitchell

Now that the BAS has produced a low-distortion sine-wave oscillator, what can one do with it? We will review about a dozen suggestions here, and if anyone comes up with additional uses, please pass them on.

But first a warning. In any test that involves playing an oscillator tone through speakers, be careful. An oscillator is a tweeter's worst enemy. In general, woofers can handle continuous tones at almost as high a level as musical peaks. But midrange drivers and tweeters, regardless of how much peak power they can handle in music, usually can take only a watt or two on a continuous, sustained basis. So when playing a high-frequency oscillator tone, keep it soft and keep it brief. Find out the woofer-to-tweeter or woofer-to-midrange crossover frequency for your speakers and "red-line" your oscillator at that frequency to remind yourself never to play tones loud at any frequency higher than the woofer crossover frequency. If you burn out a midrange driver or tweeter, it will cost you more than the price of the oscillator.

Tape Recorder Setup

The most obvious use for the oscillator is to do the setup adjustments on a tape recorder to mate it to a desired brand of tape. Complete instructions for that will not be given here. You should get the service manual for your recorder, available by mail from the factory or the importer for $2 to $5. It will contain detailed instructions, and you would want it in any case to identify the locations of the various internal control adjustments, since they usually are not clearly labeled. When ordering a service manual, state the model number and serial number of your machine in order to get the correct manual with the appropriate revisions. If you find the service manual too cursory, or if you want to plunge ahead without it, instructions for tape-recorder setup were included in the October 1973 BAS Speaker, and J. Gordon Holt published a series of articles on the subject in the Stereophile, particularly oriented toward open-reel machines.

Measuring Frequency Response

In principle, measuring the frequency response of an audio component is the simplest exercise you can do with an oscillator. Plug the oscillator into the line inputs of the device, feed the output of the device to a good ac voltmeter whose own response is known to be flat, and examine the variation in output level as the oscillator frequency is varied. However there are practical cautions which must be observed.

1) No inexpensive oscillator produces a signal whose level is absolutely constant at all frequencies. A variation of ±1 dB or so over the range is common, and this is adequate for general work—such as slowly sweeping the oscillator through the audio range while listening for

Copyright © 1975 Peter W. Mitchell

distortion or peaks and valleys in the response of a loudspeaker or headphone. But for critical measurements you must monitor the oscillator's output and readjust its level each time you change the frequency. This can mean either the use of two meters (one connected to the oscillator and the other connected to the output of the device being tested) or the use of a switch to connect a single meter alternately to the oscillator and to the output of the tested component.

2) The oscillator signal must be kept within the intended dynamic range of the test component at all frequencies. If you feed a 1-volt level into a phono input (you should not), the peculiar output that results may surprise you. And if you try to measure the frequency response of a typical tape recorder at 0 VU, don't expect to see a flat high end. In order to avoid tape saturation due to recording pre-emphasis, recorder tests should be conducted at -10 VU or lower at 7½ ips, at -15 VU to -20 at 3¾ ips, and at -20 to -25 VU with cassettes.

Dolby Record Calibration

In Dolby-equipped recorders which lack a built-in calibration tone oscillator, an external oscillator can be used. Clean the recorder's heads. Plug the oscillator into the high-level inputs and record a 400-Hz tone at 0 VU. The recorded tone should play back within ±1 dB of 0 VU; if it does not, the record calibration trimpot for that channel should be adjusted. You may need a service manual to identify the location of the pot inside the machine. Before doing this test be sure that the recorder's bias switch is correctly set for the type of tape you are using. There probably will be a pair of record calibration pots in the machine for each position of the tape bias selector switch.

Cassette Recorder Biasing

In open-reel recorders the bias must be adjusted for minimum distortion; then the equalization is trimmed for flat response in record/playback. But in cassette decks, thanks to the slow tape speed, slight adjustments of bias within the low-distortion range will alter the high-frequency response substantially, so you can mate the recorder to various tapes just by trimming the bias, without touching the equalization. A cassette deck is likely to have two or three pairs of internal bias trimpots, a set for each position of the front panel bias switch.

Once adjusted for one brand of chromium dioxide tape, the CrO_2 bias should be good for all CrO_2 tapes, because they are all quite similar. But the desired bias for iron oxide tape varies dramatically from brand to brand and from type to type within a brand. Using the oscillator, you can set the "regular" or "low-noise" bias of your machine to match whatever iron oxide tape you like. For example, in the "regular" position of its tape selector, the Advent 201 is factory biased for Scotch iron oxide cassettes, and Sony cassettes sound screechy and sibilant. But I find Sony regular LN (not UHF) cassettes a better buy, and by adjusting the "regular" bias of the 201 for Sony tape, I can make recordings on cheap Sony cassettes which sound identical to those made on CrO_2 except for a few dB of hiss and modulation noise. Of course, if your machine has a three-position bias switch, or separate bias and play-equalization switches, you should first find the switch combination which yields the flattest response with your desired tape, in order to minimize the readjustment of bias required to fine-tune the recorder for that tape.

Two cautionary notes. (1) Before rebiasing, switch off the Dolby and check the record/play response to be sure that an unsatisfactory frequency response isn't due to Dolby mistracking. For the same reason, leave the Dolby off throughout the rebiasing procedure if you do decide to alter the bias. (2) In the following procedures, if the 10-kHz tone plays back very low, it may mean either that the tape requires less bias for flat response or that the head is worn. If you suspect head wear, have it checked before changing the bias. Reducing the bias to restore highs lost due to head wear will also raise the distortion, increase the incidence of dropouts, and put a peaky and sibilant top end in your recordings which will be very obvious when you eventually do get the worn head replaced. That said, the procedure for rebiasing a cassette recorder will depend on the metering system available, as follows.

With the Advent 201 and any other deck having high-frequency pre-emphasis in its VU meters, the built-in meter can be used. First clean the heads and do the record calibration adjustment at 400 Hz carefully; any error here will affect the accuracy of biasing. Then set the oscillator to 10 kHz and record the signal at an easy-to-read level such as -3 VU. This will involve a large reduction in oscillator signal level, because of the recorder's high-frequency recording pre-emphasis. (In early 201's with low serial numbers the meter did not exhibit the full pre-emphasis, so use a lower level such as -7 VU.) If you have recorded at an indicated level of -3 VU, the tone should play back at between -1 and -4 VU. If not, adjust the bias as necessary for each channel and record again until the 10-kHz tone plays back at about the same level it was recorded at. Subjectively the recorder tends to sound best if the bias is set to make the 10-kHz tone play back 1 or 2 dB higher than recorded. Be careful always to adjust the correct bias pot for the channel being tested and for the tape type you are working with. As insurance it might be wise, before beginning, to note down the orientation of the slot in each trimpot; then if you accidentally alter the wrong pot, you can restore it to approximately the correct setting.

With the Nakamichi 500 and other recorders whose VU meters have a 40-dB useful range, the built-in meters can be used. The procedure is as above, except that the 10-kHz tone is recorded at about -20 or -25 VU.

With recorders having conventional meters, a 10-kHz signal of sufficient strength to be clearly indicated on the VU meters would saturate the tape because of the recording pre-emphasis. So an external meter is required, either an ac VTVM or a conventional VOM aided by a 20-dB preamp. (See "IC Op Amps," September 1974 <u>Speaker</u>.) Connect the meter to the recorder's line output jack for the left channel. Check the record calibration at 0 VU using a 400-Hz tone, reduce the oscillator signal by about 25 dB, change the frequency to 10 kHz, and adjust the meter sensitivity or the recorder's output level control to get a convenient reading on the external meter. Record the 10-kHz signal and play it back, noting the meter reading. If necessary, adjust the bias and record again until you obtain a reading in playback which is within ±1 dB (or 10% in voltage) of the reading obtained while recording. Then repeat the entire procedure for the right channel. As noted above, it is preferable to make the 10-kHz tone play back slightly high.

In every case, after adjusting the bias with the aid of the 10-kHz tone, recheck the record calibration using a 400-Hz tone. A substantial change in bias will affect the record calibration, so you may have to readjust the record calibration and then go back to fine-tune the bias. When finished, of course, the Dolby can be turned back on for normal recording.

<u>Measure Interchannel Phase Shift</u>

In order to record and play matrixed four-channel material (e.g., SQ) without altering its directional properties, the two channels of the tape recorder must remain accurately in phase over most of the audio range.. Consistent phasing is also important if recordings are to be made in mono on both tracks and mixed together on playback. Phasing also affects the stability of stage-center voices or instruments in stereo recordings.

The phasing is easily checked using an oscilloscope. Connect the left channel output of the recorder to the scope's vertical input and the right channel to the horizontal input. Record and play back a moderately low frequency such as 400 Hz at a moderately low level such as -20 VU, and adjust the scope controls to obtain a 45-degree line. Record and play back progressively higher frequencies. Ideally a 45-degree line will continue to be obtained at all frequencies, but as the 1972 BAS tape recorder clinic showed, most tape recorders go somewhat out of phase at high frequencies. So the 45-degree line will widen to an ellipse, a circle (indicating 90 degrees of phase shift), or will even flip over toward a 45-degree line tilted in the opposite direction (180 degrees out of phase).

Lacking an oscilloscope, the interchannel phase shift can be checked audibly by using the null switch described in the April 1973 Speaker. A 400-Hz or similar tone, recorded at identical level on both channels and played back, should drop to near-inaudibility when the null switch is activated. Repeat the test at successively higher frequencies; the less the reduction in volume when the null switch is activated, the worse is the interchannel phase shift (assuming that you have indeed recorded the signal at the same level on both channels). If there is no change in sound level when the null switch is activated, the phase shift is 90 degrees; and if the sound is stronger in the null position than in the regular stereo position, the phase shift is approaching 180 degrees.

Make a Test Tape

One of the principal sources of anxiety with a tape recorder is the fact that its heads can become worn and misaligned. To guard against this worry, make a test tape while the recorder is still new (or right after having it checked by a factory service technician), then put the tape away and play it once a month to check on the condition of the heads. When the heads do go bad you'll be able to have them fixed or replaced immediately; you won't someday find yourself in the sad position of having made many precious recordings with bad heads. What do you put on your test tape? As many kinds of tests as you can think of. Obvious possibilities include a full set of frequency response test tones and an interchannel phase-shift test (useful for checking head azimuth). At the very least record a 400-Hz reference tone and a 10-kHz test tone at a level of -20 VU or so on both channels. Play it back immediately and note the level of the 10-kHz tone relative to the 400-Hz tone; if they are not at the same level, note the difference. Then in the future when you use the tape, any drop in the 10-kHz level relative to the 400-Hz level will indicate head problems. In the Advent 201 this test is especially convenient since the tones can be read on its equalized meter. I recorded the two test frequencies at an indicated -5 VU on both channels, and the playback can be checked at any time without requiring external meters.

Such a test tape is also useful for checking intermachine compatibility. If you are concerned about whether music tapes made on one recorder will play back properly on another machine (given the possible problems of head alignment, Dolby calibration, nonstandard equalization, etc.), simply record a test tape, note how it plays back on your machine, then play it on the other machine and compare the playback results. Usually one reference tone (400 Hz) and one high-frequency tone (10 kHz) are sufficient to determine compatibility.

Test for Flutter

The ear is very sensitive to flutter in continuous tones having a frequency of about 3 kHz—more sensitive than it is to flutter in most music. So record a 3-kHz tone at a moderate level on the tape, and if the recorded tone sounds almost as pure and steady in playback as the tone coming directly from the oscillator, you can be confident that the recorder's wow and flutter are negligible. If the recorded 3-kHz tone wavers audibly or sounds fuzzy, you might want to have your machine serviced. A recorder's flutter increases as the machine ages, so you ought to do this test every few months.

Test Dolby Tracking

There are several internal adjustments in a Dolby circuit which control the behavior of the Dolby. If they go out of adjustment, the Dolby will cause level-dependent frequency-response aberrations in recordings. In order to test the Dolby tracking, the performance of the record/playback process must first be assured. Switch off the Dolby. Attach an ac voltmeter to the line output jack. Check the record calibration to be sure that the recorder plays back a 400-Hz tone at the same level it was recorded at (within ±1 dB). Then reduce the oscillator output level to about -20 VU, start recording, and adjust the voltmeter sensitivity or the recorder's output level control to obtain a convenient reading on the meter. Play back the tone; if the record calibration adjustment was made correctly, the playback reading on the external meter will be within 1 dB (±10% in voltage) of the reading obtained while recording. Change the oscillator frequency to 1 kHz (still at a recording level of about –20 VU), record, note the reading on the external meter while recording, play back the tone, and compare the playback reading to the level noted while recording. Repeat this procedure with a 10-kHz signal.

If the 400-Hz tone plays back accurately but either the 1-kHz or 10-kHz tone fails to play back within ±1 dB of the recorded level, then the machine's bias or equalization is incorrect for the tape used. The resulting frequency response errors will be magnified by the action of the Dolby. So in order to make good recordings with a Dolby, you must first have a recorder which records and plays back flat without the Dolby, i.e., a recorder which is correctly mated to the tape. A correctly adjusted recorder with good heads should have no difficulty recording and playing back those three frequencies accurately. If difficulty is encountered, make sure that the heads are clean and demagnetized and that the tape is a. good one without creases, cupping, or a damaged edge.

If the recorder is flat, then the Dolby tracking can be checked. Switch the Dolby on and redo the complete procedure described above; successively record 400-Hz, 2-kHz, and 10-kHz tones at a level of about –20 VU, and compare in each case the external meter reading in playback with the reading obtained while recording. The use of the Dolby will have no effect at 400 Hz. If the Dolby tracking is correct, the 2-kHz and 10-kHz tones will play back within 2 dB of the level they were recorded at. I.e., the use of the Dolby will not destroy the flat record/playback response which the machine had without the Dolby. If it does, the "law" and "gain" internal adjustments in the Dolby need fixing.

Incidentally, the three frequencies used here were not selected arbitrarily. They relate to the common ways in which recorders and Dolbys become maladjusted. If a recorder responds uniformly to these three frequencies, the entire frequency response curve is likely to be OK and the Dolby tracking is certain to be OK.

Measure the Impedance Curve of Your Loudspeaker

See the description by Joel Cohen in the June 1975 Speaker, as amended by Mark Davis in this issue.

Explore Your Standing Waves

Plug the oscillator into the high-level inputs of your amplifier, set it to a frequency between 30 and 150 Hz, and walk around your listening room to see how uneven the distribution of bass energy is. Try a different bass frequency, and the distribution probably will be different (though equally uneven). There may be places in the room where the bass energy at one frequency seems almost intense enough to curdle your brain, and other places where that frequency disappears almost entirely. This effect will be most pronounced at the frequencies of your room's dominant standing waves: 565/d, where d is the length or width in feet, and at multiples of those frequencies; thus in a 16-foot room, at 35 Hz, 70 Hz, 105 Hz, etc. (See "Listening Room Resonances," March 1974 Speaker.) For a related experiment, sit in your normal listening chair and slowly sweep the oscillator from 30 Hz to 150 Hz, listening for peaks and valleys in loudness. There isn't much you can do about standing waves, but it is stunningly educational to learn how uneven is the bass response you've been living with. By exploring with the oscillator, you might find a place for your listening chair where the bass is less uneven, or locations for the speakers where they don't stimulate the standing waves as severely.

Find Your Woofer's Bottom

The ability of your woofers to reproduce the bottom octaves accurately depends on two characteristics: frequency response and distortion. The frequency response depends not only on the woofer but also on the listening room, specifically on its standing waves and on the rigidity of its walls, floor, and ceiling, because these boundary surfaces tend to absorb low-frequency energy and pass it on to neighboring rooms. To evaluate the subjective low-end response of your systems, set the oscillator frequency to 150 Hz, set the amplifier tone controls flat and the listening volume to a fairly high but comfortable level, and slowly sweep the oscillator down in frequency. There will be a frequency below which the sound drops off very rapidly toward inaudibility, and even with distinguished speakers this point usually is around 50 Hz in typical rooms. (See the March 1974 Speaker for discussion of the problem and the design of a compensating circuit.)

To restore the bottom octave to audibility, bass boost is required via tone controls (if you are lucky enough to have a well-designed bass boost curve in your amp) or via an outboard equalizer. But the success of this approach depends on the distortion of your woofers remaining low at high drive levels. To check, switch in the bass boost and again sweep the oscillator down from 150 Hz or so. Naturally the quality of the tone should remain clean and its subjective pitch should continue to decline with the oscillator frequency. But it may happen that at some point you will hear a higher pitch due to "doubling," meaning that the woofer is producing sound at thrice the signal input frequency. In that case you may want to experiment with the amount of deep-bass boost to find the compromise setting which lowers the subjective frequency limit of your system as far as possible without incurring doubling. Ideally it will be possible to make the system subjectively flat to 30 Hz or so without audible distortion.

Appendix

For the benefit of new members who did not see the original article on the BAS oscillator in the July 1974 Speaker, we reproduce its schematic here as Fig. 1. Actually this is an improved version; three touchy trimpot adjustments have been replaced by a single 39-ohm resistor, simplifying the assembly and improving the reliability of the unit. Recent measurements on the improved circuit show that its harmonic distortion is typically about 0.05% at an output level of 2 volts rms.

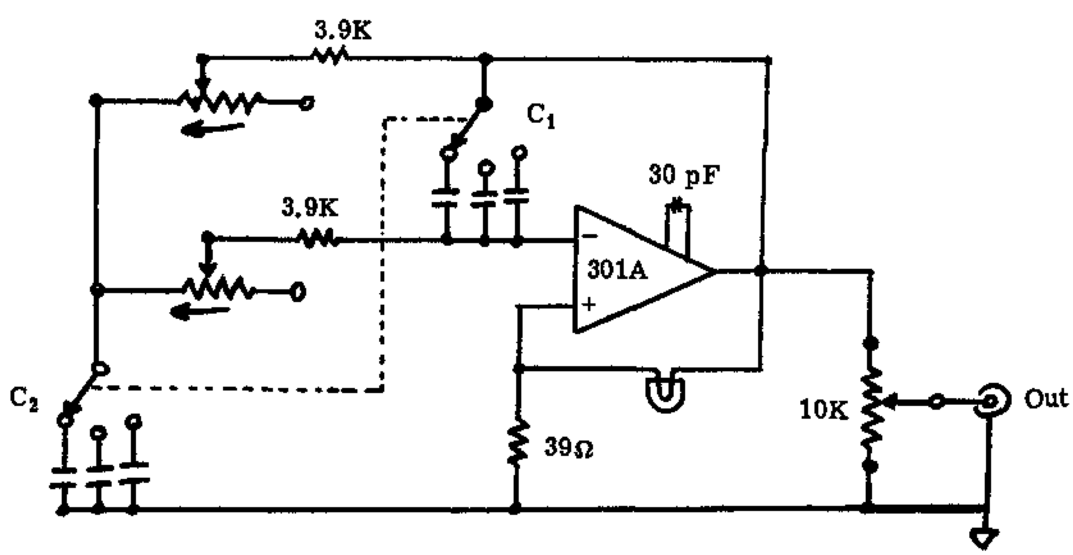

Fig. 1. Oscillator schematic. Since the dual pot turned out to have a maximum resistance of 40K on the average, the switched capacitors are as follows: C_1 0.056, 0.0056, and 560 pF; C_2 0.56, 0.056, and 0.0056 µF. The lamp is Radio Shack no. 272-1141.