THE B.A.S. SPEAKER

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In This Issue

You may be fully satisfied with your Thermo Electron 814, but according to Peter Mitchell's lights, it can be improved- He found some frequency response irregularities and cured them with a simple electronic equalizer circuit. A discussion of the response problem—a mild one—and its solution, together with plans for constructing the equalizer circuit form one of this month's publications-

Also, this month, we have a comparison of two very high quality FM tuners- The Pioneer TX-9100 seems to be approaching the "standard" category here, and Al Foster compares its performance with that of the McIntosh MR-78. The results aren't as clearcut as the price tags would indicate.

This month's third publication deals with a refinement of the pink-noise generator described by Rene Jaeger in the January Speaker, one that will make such a device more practical for most of us. Alan Southwick notes some applications for the noise generator you might not have suspected.

The newsletter section of this month's Speaker also is full of interesting material: more on TIM distortion, on tonearm damping, video recorders available on the cheap, the re-emergence of CBS's test-record series, plus regular features such as the (very interesting) report of last month's meeting and literature reviews.

• Navy surplus TS-239 general-purpose oscilloscope. Not direct coupled, but has response to 5 MHz. Many features. $28.00 Charles Pike, 862-4712, evenings and weekends.

A Spoken Speaker

Some visually handicapped members have indicated interest in a "talking-book-style" BAS Speaker. To fulfill this need, Alan Southwick has agreed to read the Speaker onto cassette for a $5 fee. He may be reached at P.O- Box 114, Billerica, Mass. 01821.

Experiments With Digital Time Delay

There being more than one way to skin a cat, and there being no digital time delay unit (as such) available, several BAS members got together to listen to a transient even recorder (TER) last month. Using a set of digital shift registers to store 10-bit-long words of music (that's not a contradiction, just engineeringese), the $3000 Physical Sciences TER made a dandy stereo digital delay line.

There were flaws: the unit's resolution was a bit poor and its output was noisy and contained clock-error noise, some aliasing, whistles, and squeeks. But addition of a dbx 157 2 :1 compressor at the input and output of the device turned it into a fairly high-quality delay line.

The results were exciting. In a system consisting of AR-LST-1's in front and LST-2's in the rear, the effect was judged superior to that of SQ discs and broadcasts (as something like full channel separation was maintained), and perhaps even superior to the discrete four-channel Boston Symphony Orchestra live broadcasts of two years ago.

"It was possible to `feel' the bass transients traveling down the hall," says BAS-member Ira Leonard. "The hall was recreated in a very realistic fashion, not exaggerated, but compared with two-channel, or matrixed or derived four-channel sound, plucked bass viol, bass drum, and other low-frequency transients were greatly enhanced in impact."

This seems to give the lie to the view that only the midrange and high end are needed for satisfactory quad reproduction. It also parallels Rene Jaeger's opinion that it is the accurate reproduction of the lowest frequencies that makes for the most realistic recreation of space. (See the meeting report in this issue.—Ed.)

"Cutting out the delay was a stunning experience in itself," says Leonard. "It was as if the sound field—which had formerly filled the room—collapsed into the flatness of the front wall. The sense of space vanished."

The experimenters found that the gain of the front channels relative to the rear channels was perhaps less critical than delay. "We found that we could play the rear channels at within 3 dB of the fronts with utterly no sense of sound originating from the back of the room," says Leonard. On the other hand, "delays of 50 milliseconds tended to accent the occasional note or create an echo effect, making the ear uncertain as to which was the 'front,' so to speak. At delays of 100 milliseconds, the ear mistook the rear signal for front channel information—the back of the room turned into the front. A very disconcerting effect."

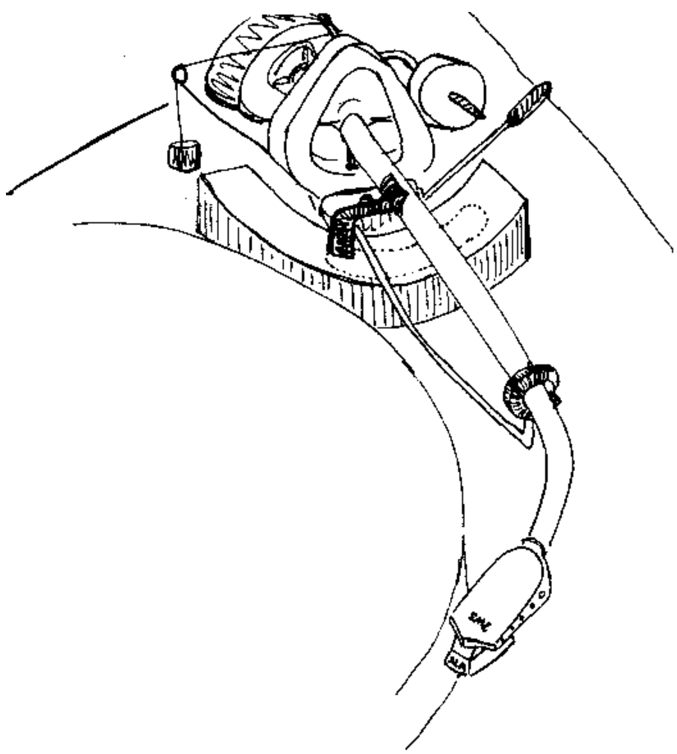
Dry recordings with little artificial reverb were found to sound best with digital delay, and those selections with low-frequency transients seemed to profit most from its enhancement of realism. Finally, noise was judged to be a prime distraction and destructive of the four-channel effect.

The experiments later were repeated in another room using KLH-9 electrostatic loudspeakers. The results were less dramatic than with the AR dynamics. The bipolar radiating pattern of the model 9's may have had something to do with these results, but in general, according to Al Foster, "It was possible to prefer either four-channel discrete tape or quad derived via delay, although (perhaps because of the mike placement used in Symphony Hall, the source of the four-channel tape) the delay added a 'sense of hall ambience.' Delay tended to 'enlarge' the sound, making it less useful, perhaps, on solo and jazz, than on large orchestral recordings."

All in all, digital delay seems to be well worth waiting for- It seems to promise far more realism and freedom from distortion than any of the quad disc formats, and was judged far superior to "ambience recovery" a la Hafler. Other BAS members participating in the experiments—the latest in a continuing series—included Mark Davis, Rene Jaeger, and Bill Wolk.—Jim Brinton

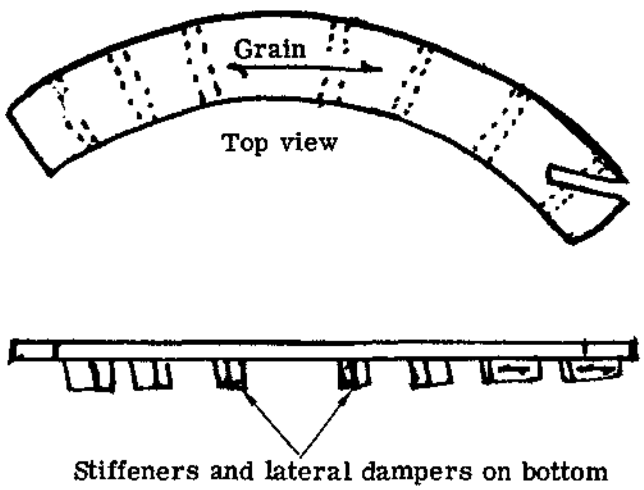
Wooden Damping

To damp my ADC-XLM/SME combination, I am using a trough made of plastic-coated balsa wood formed in an arc with its center at the pivot of the SME. This trough also acts as an arm stop to prevent the pickup from swinging beyond the center of the disc. The piston assembly is of 1/16-inch balsa. Rubber cement bonds the damper to the arm and additionally waterproofs the balsa. The piston is curved to conform to the trough.

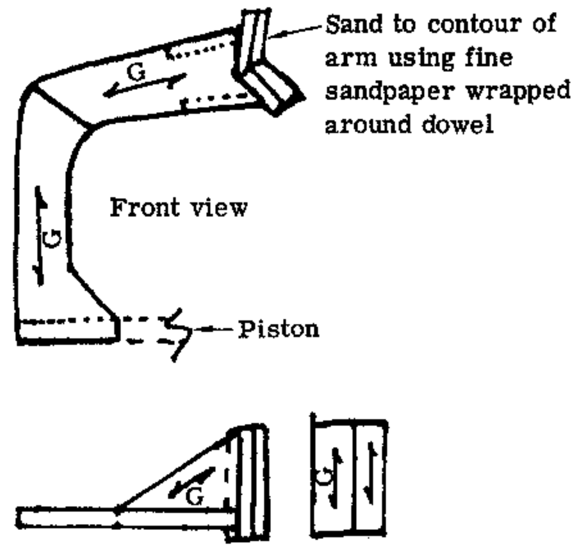


Since the piston materials are ultra light, they do not compromise the design of the low-mass SME. Also, I have compensated for the mass of the damping assembly by "gluing" the XLM to the fixed shell (without finger lift) using GE RTV silicone rubber. To compensate for buoyancy , the arm is balanced with fluid in the trough. Currently the fluid is not STP, but Corn Huskers Lotion.

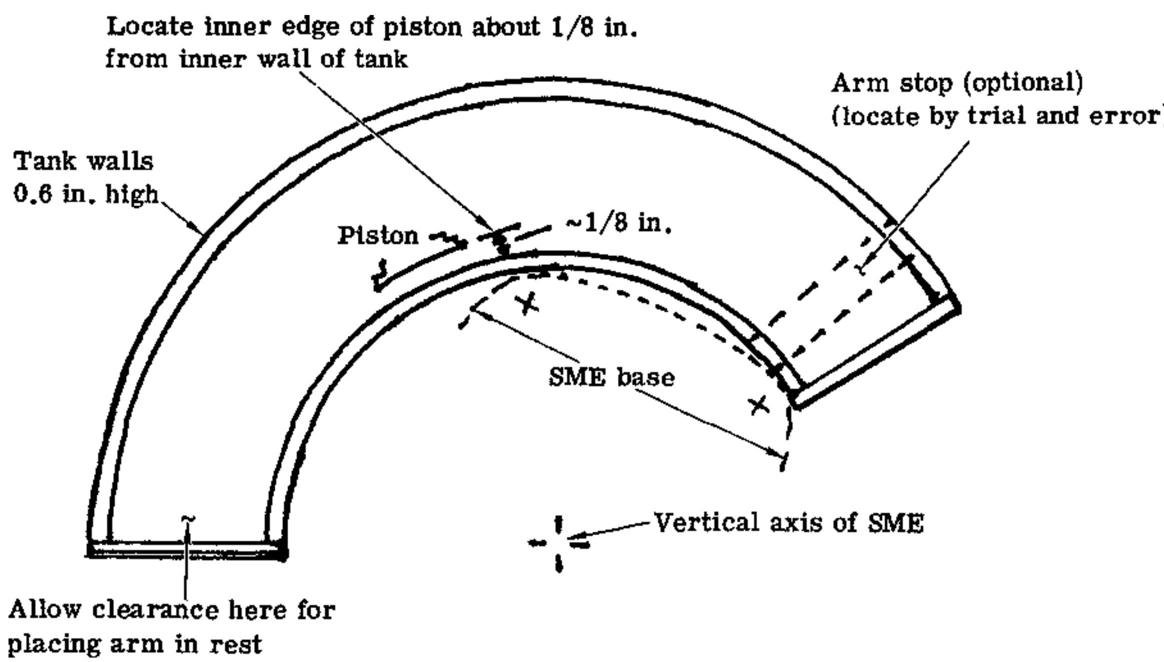
So far as balsa's rigidity is concerned, I feel that the stiffness of the piston assembly is nearly infinite relative to the stylus compliance, which should make it sufficient. Note too, that balsa has internal damping, unlike metal, and its strength-to-weight ratio is higher than that of almost any structural metal. Proper grain orientation should give all the stiffness needed. I do not try for super damping just enough to control the arm over warps or to prevent skipping caused by floor vibration or shock.



PISTON



PISTON ARM (rotated 180° from tank view)



TANK

Full scale

NOTE: This piston assembly is for use with glycerin. For Corn Huskers Lotion, make piston smaller. Piston material is 1/16-in, balsa. Note balsa grain direction. Bond balsa to balsa with airplane glue. Bond piston to arm with rubber cement. Coat piston assembly and tank with G.C. tape liquid (or equivalent) to seal balsa (also gives neat all-black finish).

The rubber cement joint is not rigid, but in use it seems stiff enough. The advantages are low mass and easy removal without marking the arm. One could use rubber-to-metal cement or epoxy, but I wouldn't recommend it.

I question using STP or any other petroleum-based fluid, since the aromatics given off may attack the elastomer suspension of your cartridge. These same fumes might also attack your records, with both problems aggravated by a dust cover if you use one.

I have been trying fluids such as glycerin and heavy mineral oil, neither of which attack much of anything. I have not tried pure silicone fluid as yet, but it would seem that the fluid to beat—so far as easily available fluids are concerned—is Corn Huskers Lotion, of all things. This hand lotion is extremely viscous, much more so than STP, and can easily be diluted with glycerin. These two characteristics make it a prime candidate for damping applications, as its higher viscosity would make possible smaller damping assemblies, while its miscibility makes it easy to achieve the precise viscosity needed.—Tom Mashey (Connecticut) (See additional note, page 17.)

Mashey's approach accomplishes several useful things. The trough is light and small enough (when used with the very viscous Corn Huskers Lotion) to be installed on the SME's mounting plate—a boon in installations where the trough (but not the arm) would otherwise be subject to vibration from external sources as on the AR turntable. The offset arm attaching the piston to the arm allows the SME's armrest to be retained. Finally, the design of the piston is such that damping *is* applied vertically, through the center of the pivot-to-stylus line. This is superior to the offset method developed here originally, although the effect probably is small-—Jim Brinton

The 814 Column

When Peter Mitchell published his article on the Thermo Electron 814 microphone in the February Speaker, we thought it was a pretty "compleat" treatment. It wasn't, as the March Speaker's "A Case for the 814," by Alan Southwick, showed, and as this month's Mitchell piece on "Improving the 814" underlines.

Most of this outpouring is the result of satisfaction with the TE 814 and the interest the unit has triggered as it, better-mouse-trap-style, beat a path into people's recording systems.

Kits and Assembly. To help those interested in getting this bargain performer, Al Southwick will supply a kit of parts including machined microphone cases for $15 (see his March article for details). For an additional $10 each, he will supply assembled cases for your capsules- These units would be phantom powered. Send prepaid orders to P-O. Box 114, Billerica, Mass. 01821.

Peter Mitchell also hopes to offer assembled battery-equipped 814 mikes installed in 1-inch aluminum tubing. He may be reached via Box 7.

Peter cautions capsule buyers about the possibility of damage to the unit when soldering its pins; it is much safer to use a socket as described last month by Al Southwick.

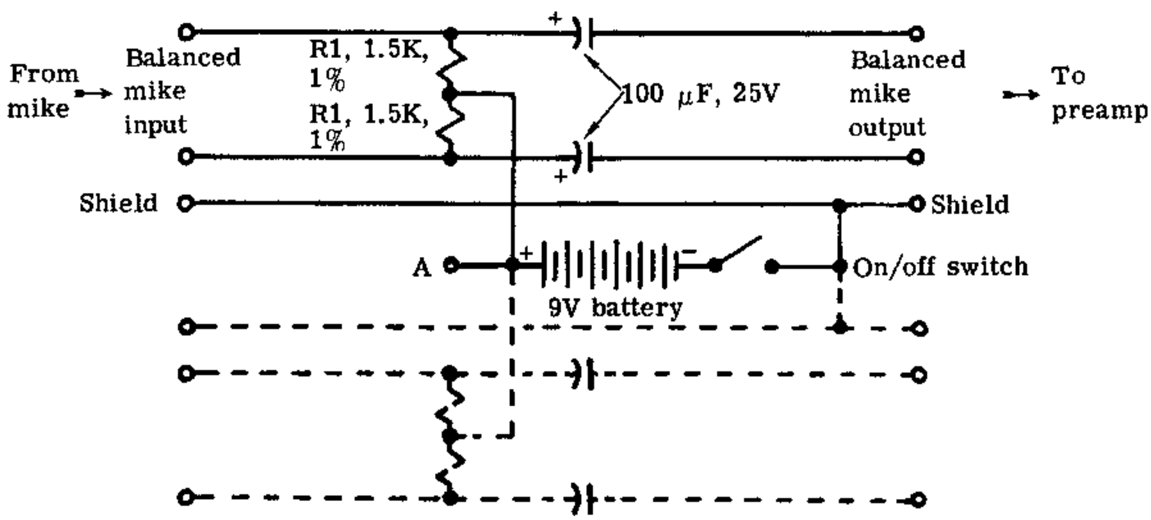
814 Erratum. The upper graph in Figure 5 of Alan Southwick's "A Case for the 814" is mislabeled. Instead of plus and minus 5 dB, the labeling should be plus and minus 10 dB.

Battery (Phantom) Power for the 814

Both the 814 and 814C will operate satisfactorily with a 9-volt supply, and the Zener diode used within the case described last month will protect the capsules from any overvoltage. Batteries can be series connected for higher voltage operation, particularly with the 814C, but the R_1 resistors shown in Fig. 1 of last month's article must be changed to conform with the formula

$$R_1 = \frac{\text{Supply voltage}}{2 \times 0.003 \text{ amp}}$$

Several mikes can be powered from one battery, or battery package, using the basic circuit shown in the figure.



The solid lines indicate the simple one-mike power supply; dashed lines indicate connections for multi-mike operation. Note that the 1% resistor pairs are duplicated, and that there are now 100-microfarad capacitors in the signal lines to prevent dc from entering your preamp's input circuitry. Alkaline or mercury batteries are recommended—especially for multiple mike operation because of higher current drain. Battery voltage can be measured (under load) between the shield side of the mike line and test point A.—Alan Southwick

EE 814 Versus BAS 814

Being human, it is impossible to resist drawing unflattering comparisons between the 814 microphone as realized by the BAS—notably Peter Mitchell and Alan Southwick—and its Electronic Enterprises incarnation (see review in The Audio Amateur, 3/74). Admittedly, the BAS is a nonprofit group, but even buying the capsules without quantity leverage, we are able to get them to you for $27 each; EE charges $42.00 for the capsule, and we assume that the company is buying enough of them to get some small price cut. The kit of parts sold for $23.50 by EE as the 814K includes a low-voltage penlight battery (soldered into the circuit, yet) mounting tube, and carrying case; the BAS's version includes provision for phantom powering (or high-voltage, and thus low-distortion, battery operation), balanced-line output, radio-frequency-interference protection, and looks first class at a $15 total price.

EE's total is $65.50; the BAS's is $42.00, for a difference of $23.50. For $10 more, Al Southwick will assemble the BAS version for you—and you are still $13.50 ahead. And about half-a-light year ahead in terms of low-distortion, low-RFI, high-overload performance.

This isn't an anti-capitalist tirade; if we were trying to make money, we might have to sell these units for something approaching EE's price. This is just to point out that you can get a better performer for less money as a member of the BAS.—Jim Brinton

CBS Test Records Again Available

CBS is again offering its test record series, although for some reason they're not making it exactly obvious. Now available are:

- STR-100. 10 Hz to 20 kHz frequency sweeps, which should yield a constant-voltage output from a "perfect" (and non-RIAA equalized) phono cartridge. $10.
- STR-101- Noise bands for tone control calibration, speaker phasing, tracking tests, etc. $6.98
- STR-130. 40 Hz to 20 kHz frequency sweeps, which should give a constant output voltage from a "perfect" phono cartridge with RIAA equalized preamp- $15-00

STR-110, -111, -120, and -140 probably will be available sometime in May. To purchase, add $1 for shipping and send check to: Columbia Special Products, 51 West 52nd Street, New York, New York 10019. Attention: Craig Ramsell. Ramsell's phone is (212) 765-4321, ext. 6106.—Dan Shanefield (New Jersey)

Recommended Demo Disc

Acoustic Research, Inc., Demonstration Record ENY-AR-1, " The Sound of Musical Instruments," is my personal favorite for giving the feel of the concert hall in a home listening room. It sounds especially good with Magnaplanar speakers, somewhat less so with electrostatics such as Quads or B&W-70's (although I had no opportunity to equalize the latter two systems as I did the Magnaplanars).

On Magnaplanars, this record's realism beats such superstar discs as Sheffield III, Nonesuch H-71291 Percussion, Armstrong High School Choir 1971, Ambiphon demo tape, etc. The disc has a rather distant sound—maybe about row N—even when equalized to produce a presence peak, but it provides a live-performance feeling especially with violins and percussion. The musical selections are mostly classical, with performance varying from OK to excellent.

Available for $5 postpaid, from AR, Inc., 10 American Drive, Norwood, Mass. 02062. — Dan Shanefield

Videophile Note

One Charles Knickerbocker (Knickerbocker Enterprises, Windcliffe Drive, Ballwin, Mo.) has evidently acquired the finished and in-process inventory of Cartrivision color video tape recorders. He is selling them without knobs, front panel, or cabinet (they normally were installed in console TV sets) but fully aligned and running for only $150 each. These are the same units being sold "as is" by Cramer-Olsen at $300 (removed from defunct Packard-Bell TV sets).

I recently received mine, and it works superbly. Of course, these units aren't up to the standard of modern cassette video recorders selling for $1200 to $1500, nor is the Cartrivision tape cartridge interchangeable with what's used today. Cartrivision uses a "skip field" system, recording every third frame and playing it back three times. Fast-action scenes may thus appear a little jerky.

Spare head sets and electronics also are available as are service manuals. I am checking on tape availability.

I have inquired about group purchases, and I suggest that anyone interested contact me as soon as possible. If you Would like more information, please call me at (617) 890-1727 (days) or (617) 734-5070 (nights). —Joel Cohen

Another Cartridge Heard From

My old Goldring cartridge having gotten bent, I ordered a new Stanton 681EEE. While waiting for it to arrive, I borrowed an Empire 2000E/III and got the shock of my life when trying it. It tracked the Shure TTR-102 test record perfectly and was flat to within a dB from 30 to 15,000 Hz. I have never heard sweeter, less "resonant" highs or cleaner, deeper bass on any cartridge. There is no sizzle to a soprano's top notes; instruments are absolutely clean and as realistic as anything I have heard live. . . unbelievable. I have tried everything available here except the Stanton 681EEE and B&O MMC-6000, and it is better than any other so far.—Nate Garfinkle (California)

TIM Revisited.. .

. . . And not for the last time either. Bob Carver, President of Phase Linear, like you gets his copy of the Speaker each month. He naturally saw the article on TIM distortion in the February Audio and the Speaker's note on the same topic in its February issue. He has this to say about TIM:

"An audio power amplifier has two basic limits. The first is the limit that is associated with the amount of input signal. If this limit is exceeded, the familiar overload condition known as clipping results. The second is the limit that is associated with the speed [i.e., frequency or rise time] of the input signal. If this limit is exceeded, a less familiar overload condition exists that historically has been called 'slew-rate jamming.' Recently, this condition has acquired the title of TIM. In a well-designed amplifier, both limits are reached abruptly, although not necessarily simultaneously. For example, prior to ordinary clipping, distortion is very low and rises abruptly after the onset of clipping. Similarly, as the slew-rate limit of the amplifier is approached, distortion remains very low until the slew rate limit is exceeded, and then distortion (TIM) rises abruptly and rapidly.

"If an amplifier is put on a lab bench and hooked up to a signal generator and an oscilloscope, it may be made to overload by applying an input signal that is either too large or too fast or any combination of both.

"If an amplifier is hooked up to a high-fidelity system, it follows that in order to prevent distortion, the amplifier must not be overloaded by the musical input signal. The input signal must be small enough and slow enough such that the two limits are not exceeded.

"Now for the $64,000 question. Can the musical signal source from a tuner or cartridge or tape recorder exceed the slew-rate limit of a conventional well-designed (heavy lag compensation) amplifier? The answer is NO WAY! (Provided no defective equipment.) This is not a matter of opinion; it is a matter of fact.

"The reason for this is that modern high-fidelity design always sets the slew-rate limit high enough such that slew-rate limiting (the cause of TIM) will only occur long after the onset of ordinary amplitude overload (clipping). Generally speaking, in order to reach the slewing limit of a modern transistor amplifier, it must be driven approximately 20 dB beyond clipping on musical material. Even the abrupt, steep rising wavefront of two wooden blocks smacked together will not approach the slew-rate limit of a conventional well-designed amplifier. Even if the amplifier is well into clipping. And certainly not if the amplifier is operating below clipping.

"My advice is, if you own a properly operating amplifier manufactured during the last 15 years, don't worry about TIM. For all musical matters in this real world, it doesn't exist.—Robert Carver (Lynnwood, Wash.)

Rabco Frustrations

For two years now a Rabco ST-4 has been my principal record player. I am convinced that the straight-line tracking makes for clean sound at minimal tracking weights. I like the ability to make quick cartridge changes by having extra complete arm assemblies. I also like the ST-4's almost total immunity to floor vibrations and feedback (of the players I know about, only the AR is its equal in this respect).

What I don't like (aside from the difficulty of accurately setting tracking weights) is the cheesy turntable that comes with the ST-4. When I bought mine, I couldn't find one sample that didn't have a wobbly platter and a sloppy spindle bearing. I was on the point of trying to install an AR table on the ST-4 chassis, but came to the conclusion that the deeper skirt of the AR platter would interfere with the belt that drives the ST-4 arm's tracking rod.

I'd be interested to hear from any BAS members who might have tried to mate the ST-4 arm with a better turntable. Could the tracking rod be driven by a separate motor, or would that introduce hum and/or rumble problems?

Perhaps this concern will become moot if the newer model to be introduced by Harman-Kardon retains the advantages of the ST-4 arm and has a better turntable.—Tom Shedd (Illinois)

Unpopular Recommendations From *Popular Science*

The February issue of Popular Science, a normally respectable magazine, contained reviews of 20 under-$100 loudspeakers. William J. Hawkins, unknown here, compiled the report and found the Dynaco A-25 ("honky") and AR-7 ("lacking in brilliance") coming in among the worst according to PS's listening panel's "subjective tests." Listed among the "best" were the Kenwood KL-55 and Superscope S-28. Literally incredible.

Perhaps the listening room's acoustical tile wall coverings were just too different from the furniture and carpeting that characterize most home listening rooms for the panel accurately to evaluate speakers for home use. Also, as photographed, the speaker auditioning setup looked a lot like that found in a hi-fi salon, and the difference in sound between such an environment and the home is legendary.—Dan Shanefield

Vestigal Arm

We understand that The Absolute Sound has received lengthy correspondence on the Vestigal arm. Editor Harry Pearson has solicited letters from diverse viewpoints. Try to catch the forth-coming issue—as well as the recent Stereophile.—Jim Brinton

In the Literature

Journal of the Audio Engineering Society, Jan.-Feb. 1975

- NQRC Measurements of Subjective Aspects of Quadraphonic Sound Reproduction—Part I: Study of subjective sound localization.
- Electroacoustic Transducers with Piezoelectric High-Polymer Films: Pioneer's piezoelectric transducers. (p. 21)
- Development of Compound for Quadradiscs: RCA's study of same, with some good photos of relative groove wear with Shibata and conical styli. (p. 27)

Journal of the Audio Engineering Society, Dec. 1974

- High-Fidelity Sound-System Equalization: Interesting discussion of a method of locating speakers within a room so as to provide smoothest response through minimal excitation of room resonances. A one-third-octave pink noise source and a good mike are used to measure the speaker/room response. (pp. 795-799)
- On Aural Phase Detection: Quantitative determination of the ear's sensitivity to phase shift: indicates that phase shifts of as little as 10 degrees are audible as changes in timbre. (pp. 783-788)
- Long-Term Durability of Pickup Diamonds and Records: Report with photomicrographs showing that use of the Lencoclean system (which applies a liquid film to the grooves of a disc as it is played) greatly reduces wear both of stylus tip and record groove. (pp. 800-806)

Audio Times, March 15, 1975

- News items: Sanyo is buying half-ownership of Fisher. KLH's new "Research X" line will be strictly fair-traded and marketed separately from KLH's classic line of speakers. JVC is planning audio clinics in stores to test not your equipment but your ears—including your frequency response, your SPL level preference, your wow-and-flutter threshold, etc.

Barclay-Crocker Reel News Nos. 15 and 16

- Discussion of Ambiphon tapes, with 96 dB S/N, running 25 minutes total with solo piano music in discrete four channel.

db, March 1975

- Versatile, Low-Level Crossover Networks. (p. 22)
- Solid-State Switching for Audio. (p. 26)

Electronic Design, March 15, 1975

- PLL IC's Taking on Dedicated Jobs as Temperature and Voltage Ills Recede: Phase-locked loop feature includes several units used in FM. (p. 38)

Electronic Engineering Times, March 10, 1975

- $150 Preamp Competes with $700 Units: Review of the All-Test Devices ATD-25 phono preamplifiers. "The ATD-25 is bettered, at this time, only by the $700 Audio Research and $900 Levinson preamplifiers," but there was a question about the possibility of TIM. (p. 37)

Electronic Engineering Times, March 24, 1975

- FTC to Propose Audio Device Regulations; Report on FTC plans to ask Congress to expand its authority to cover interference-rejection requirements in entertainment electronics devices.

Electronics, March 6, 1975

- Zenith's Serious About Audio—Just Ask Barry Kipnis: Short interview with Zenith's chief engineer. (p. 14)
- One-Chip FM Demodulator Needs No Alignment: Design using NE-563 IC. (p. 94)

Electronics, March 20, 1975

- Antilog Function Generator Keeps VCO Output Linear: Presents a seven-semiconductor circuit which generates a 20- to 20-kHz sawtooth, log-linear sweep, from a -5 to +5 control voltage input. Could be useful for a sweep generator, although the sawtooth output is not the most desirable. (p. 115)

- Silent Timer Warns of Tape Run-Out: Digital timer circuit designed to be added to cassette units to warn after 44 minutes elapsed time.

Electronics, April 3, 1975

- News Briefs: Announces that Collins Radio is the first manufacturer to specify the IM distortion of their stereo FM transmitter, which is given as "0.5%" for stereo — and they seem proud of this figure.
- Video-Disk Battle Goes Public. Announces that Philips and RCA have demonstrated their systems, and Teledec is selling theirs; we wonder how they sound.

Elementary Electronics, March-April 1975

- Class-D Pulse Power: Construction project; not state of the art.

Hi-Fi News and Record Review, U.K., March 1975

- The Disc: Present State of the Quadraphonic Parties: Part of a series. Current status of QS, SQ, CD-4, and UD-4 as described by their developers. (pp. 71-81)
- Equipping an Amateur High-Fidelity Workshop (Part II of V): Building a low-distortion sine/square wave oscillator, somewhat like an elaborate version of the BAS oscillator. (pp. 63-67)
- TV/FM: Making a UHF-to-FM RF converter for UHF-TV sound. (pp. 91-93)
- A survey of future improvements in discs.

IEEE Transactions on Consumer Electronics, Feb. 1975

- A Monolithic Four-Channel/Stereophonic Decoder for FM: GE reports on its two-IC breadboarded decoder for quad FM. Separation is high at 30 dB between all channels, but so is distortion at 1%. (p. 95)
- An IC Sound Channel for Television Receivers: Not the first. (p- 74)

Popular Electronics, April 1975

- A Short History of Four Channel: More interesting than the title sounds; emphasizes reverberation approaches and the BB&N system. (p. 17)

QST, March 1975

- RFI—A New Look at an Old Problem: Discussion of the legislative efforts to remove RFI from equipment. Also discusses an RFI information packet, available from an amateur radio task force on RFI, which is a letdown. A list of TV manufacturers who supply TVI filters is included, and one wonders if we couldn't compile a list of cooperative and uncooperative audio equipment manufacturers ?

Radio-Electronics, March 1975

- This Tuner Costs $2500: Len Feldman tours the guts of the Sequerra Model One. Text offset by charming hand-drawn simulated scope presentations. (p. 30)
- One-Sided Noise Reduction System: Feldman takes on the Burwen DNF-1201 part by part; last two paragraphs of 2½-page article state that auditioning took place during AES convention, during hurly-burly of which, Feldman could detect "no loss of high (sic) whatsoever." (p. 37)
- The Battle of the Tapes: New developments in cassette and open-reel decks compared. (p. 40)
- FM Tuner Roundup: Survey with comparative spec tables. (p. 44)

The Stereophile, Spring (1), 1975

- Reports on the Jecklin-Float headphones, Paoli 60M amplifier, and the Transcriptors Vestigal tone arm in which J. Gordon Holt says some very blunt things about Transcriptors' ads, paralleling Lipshitz' letter in the March Speaker. A vehement reply from the manufacturer displays not only his complete lack of tact, but also his stupidity or ignorance in re mechanical systems.
- If you need a cookbook for making component choices (or the blessing of an authority figure—Ed.), the list of Class A through D items for all parts of a system (given as recommended components) will wilt your checkbook.

Tektronix Cookbook of Standard Audio Tests

- Widely advertised, this free brochure is a sales pitch for Tektronix's audio testing scope/generator/spectrum analyzer. It contains information well worth your postcard request from Tektronix, Inc., P.O. Box 500-S, Beaverton, Oregon 97005.

March BAS Meeting

Business and Open Discussion

More than 50 BAS members and guests met at BU's George Sherman Union for the March meeting. Al Foster announced that he would not be distributing Advent CrO_2 tape but that a delivery was expected soon. Sadly, however, this tape will no longer be available at the old discount price of $2.00 per C-90 or $23.50 for a case of 12. Foster distributed the Shure TTR-103 test record for those who ordered copies.

Those who ordered the Thermo Electron mike capsules picked them up from Peter Mitchell, and new orders were accepted. Al Southwick demonstrated his adaptation of a Cannon/Switchcraft connector as a case for the TE mike. He is willing to organize a group purchase of the connectors and arrange for the necessary machining (see "The 814 Column"—Ed-)-

Heath 1O-18 oscilloscope kits are still available from Lawrence Kaufman at $60 each.

All of those interested in ordering Scotch 117 tape in bulk pack at about $6 for 3600 feet should get in touch with Jim Richardson. The 117 has no Posi-Track backing. Richardson also will be ordering reels and boxes.

Meeting Feature: Why the Audiophile Cannot Get Through the Maze

Rene Jaeger, Chief Engineer at dbx, Inc., has been considering the problems of the audiophile from both a professional and a personal point of view for years now. He has made substantial progress in penetrating the smoke screens, avoiding the diversions, identifying the false trails, and debunking the popular hoopla that make up the maze the audiophile must disentangle in his quest for the perfect sonic experience.

For most audiophiles the starting point is recorded sound, either discs or tapes. But it soon becomes apparent that the quality of generally available software is a severe limiting factor in achieving a satisfying musical experience. Except for some rare examples, an extensive amount of signal processing has taken place by the time the audio signal is recorded, making it extremely difficult to retrieve a close facsimile of the original performance. In dealing with this situation, the audiophile must seek out the best recordings available, be they his own or those few commercial examples of superior sound, e.g., Sheffield, Fulton, Marc Levinson, and certain European discs. Not even all of these recordings are good in all respects.

Cartridges. Rene believes the cartridge is the component most in need of improvement. Indeed, the lack of good cartridges is what has driven many audiophiles to tape even though discs are potentially superior to tape. Capable of greater (ungimmicked) dynamic range, and better signal-to-noise ratio, disc performance now is limited largely by the cartridge (the capabilities of disc cutters now exceed cartridge performance by a fair margin). Sheffield discs, for example, routinely exhibit a greater than 70-dB dynamic range.

What are the limiting factors in current cartridges? Rene feels the important characteristics for cartridges may be the ones not specified, e.g., distortion, phase response, and impulse crosstalk. With the low values of distortion present in today's electronics and speakers, cartridge distortion can predominate in many situations. Magnetic cartridges are particularly distortion-prone because of their usually single-ended magnetic transducer structure, which is nonlinear at large stylus excursions (B&O and Empire make exceptions). At 12-cm/sec velocity, harmonic distortion can be 3% or greater and is worse for vertical displacement than for horizontal (making vertical arm damping critical) (and warps an irritation—Ed.).

Rene's investigations show that impulse or transient perception forms an important part of the psychoacoustic appreciation of stereo space. The phasing and separation of transient information between channels must be preserved to fully recreate the spatial aspects of the original sound. Since much of this transient information is in the upper frequency range, the phasing and separation of channels in cartridges in this region is important, but seldom specified by manufacturers or measured by reviewers. Because cartridge mistracking is more likely to occur on transients, it is not always possible to infer the degree of impulse crosstalk from the usual separation versus steady-state frequency curves.

Rene has examined the impulse crosstalk of several cartridges by playing a disc with square wave modulations on one channel and looking at the output of the other channel. For the QDC-1, the sine wave crosstalk was -30 dB but impulse crosstalk was only -15 dB. In Rene's opinion, the QDC-1 has problems with stereo imaging, which could be accounted for by these figures- His sample maintains good right and left channel signals but has little center fill and virtually no depth information. On the other hand, the ADC-26, Rene's favorite cartridge, maintains -20 dB crosstalk for both sine wave and square wave signals, and exhibits good center and depth imaging. He indicated that ADC designs cartridges for best impulse crosstalk.

Rene believes that maintaining separation in a cartridge is almost entirely a mechanical problem related not only to geometry but also to stylus and shank resonances and how well they are controlled. These resonances, although occurring at 30 to 35 kHz, can also dump energy into the electrical response peak formed by the resonance of the cartridge inductance with the cable and preamp capacitance, causing annoying noise output at the upper end of the audio spectrum. To avoid this, a good cartridge should not have a peak in the 20-kHz region. Its square wave response should have a slightly rounded leading edge with no overshoot or ringing. Both the ADC-26 and AT-11 show this favorable characteristic, according to Rene's measurements.

Preamps. With the present state of the art, Rene believes there is no reason why any preamp should not have a signal-to-noise ratio at least 80 dB below 1 volt output for high-level inputs and 70 dB below 1 millivolt input for phono inputs. Actually this specification would be more useful for comparison purposes if it were given as a "noise figure" which would not depend on the gain of the unit or the signal level.

Most cartridges are specified for operation into 47 Kohms with recommended capacitance values sometimes given too. Rene speculates that these may not be the best values and that preamps should have adjustable phono input impedances to optimize cartridge response.

Amplifiers. With all of the factors that must be taken into account (e.g., distortion level, room size, speaker efficiency, source material, etc.), Rene was not willing to make any pronouncements on amplifier power requirements. He did say, though, that amplifiers of 50 watts and greater should have a signal-to-noise ratio of at least 100 dB at rated output.

Possibly the most common problem encountered with amplifiers is their reaction to the unusual loads presented by some speakers. The hard clipping characteristic of solid-state amplifiers combined with the inductive loading of some speakers can generate voltage spikes with a high harmonic content. Rene feels something could be done to soften this clipping characteristic and improve the sound at peaks. This might take the form of a diode limiting circuit at the amplifier input which would begin a gradual signal limiting action as the amplifier power supply nears collapse.

TIM. There has been much discussion lately about transient intermodulation distortion, or TIM, in amplifiers. This reportedly occurs most often in amplifiers employing large amounts of negative feedback (40 dB or more). When the input signal changes or slews more rapidly than the amplifier's feedback network can follow, an overshoot might occur that could saturate the amplifier's input stage. This may momentarily distort or suppress concurrent low-level signals, hence the term TIM (see Audio, February 1975, p- 34).

To get a better understanding of the effects of TIM and the conditions under which it can occur, Rene made some measurements on a 741 op-amp connected in a unity-gain buffer configuration. This should be the most likely condition for TIM, since 100% feedback is used and the 741 is notoriously slow (about 0.5 to 0.8 volts per microsecond slew rate). By feeding in a square wave and subtracting the output from the input, differences resulting from amplifier distortion were readily apparent. With a 3-volt peak-to-peak 10-kHz square wave having a 1-microsecond rise time, Rene noted a 5-microsecond, 1-volt spike on the edge of the output square wave. This correlated with the expected results derived from delay time considerations. Rene then added a 1-volt-rms, 10-kHz sine-wave signal to the square wave and looked for distortion products with a wave analyzer. In the audio spectrum, spurious signals were 65 dB down, while at 200 kHz, they were down 40 dB. Under these conditions (which exceed any likely to be encountered in an audio system), Rene found that the input stages of the 741 had not overloaded- TIM, then, is probably not the reason that the 741 sounds slightly inferior to higher-frequency op-amps in audio circuits. He concluded that preamps would be relatively free of audible TIM- It may, however, be significant in other components.

Power amplifiers may have large bandwidths for low-level signals but their power bandwidth usually extends less than an octave beyond the audible spectrum. High-frequency signals about 20 kHz generated by cartridge mistracking may exceed the slew rate of the amplifier and precipitate TIM. Rene recommends that a bandpass filter be used after the preamp to eliminate this high-frequency garbage (and subsonics also) which may be unnecessarily taxing the power amplifier.

Distortion. Pursuing the distortion investigation further, Rene set about to determine this own distortion sensitivity under various listening conditions. He constructed a circuit with which varying amounts of distortion could be dialed into the audio signal. The distortion was generated using a multiplication technique which preserved the signal impulse response. Since this is somewhat different than the distortion mechanisms of preamps and amplifiers (e.g., slew-rate limitations, charge storage problems), the conclusions cannot be applied directly to all equipment.

With the knobs uncalibrated, Rene and others attempted to adjust the box controls for minimum audible distortion in music heard over various speakers at the SPL range of optimum ear sensitivity (80- to 90-dB SPL). A range of settings gave acceptable sound, but subsequent measurements indicated that distortion usually became noticeable at settings giving 3% second and 1% third harmonic distortion. In a more critical test, ESP-9 headphones were used to listen to a 1-kHz sine wave, the second and third harmonics of which fall in the most sensitive region of the audible spectrum. In A-B comparisons, Rene could hear 1% second harmonic (40 dB below the fundamental) and 0.3% third harmonic (50 dB below the fundamental) distortion at about 90-dB SPL. Similar results were obtained with electrostatic and dynamic speakers. Sensitivity to the higher harmonics is greater because they are further from the fundamental and not as well masked by it. Above 90-dB SPL the natural distortion of the ear begins to dominate. At 100-dB SPL, the second harmonic became just audible at 3% for a sine wave signal.

It is not really valid to compare distortion figures between different components, Rene feels. IM distortion in an amplifier, for instance, may sound different than the same value of IM in a tape recorder because the mechanism for producing it in the two units is different. Tape IM distortion can be due to IM between the tape noise and the signal and will sound different than the IM distortion between two signals in an amplifier.

Loudspeakers - According to Rene, one of the most important characteristics of a loudspeaker is its impulse response. It is the attack transient of an instrument that gives it its characteristic sound and anything that alters this attack will change that sound. To demonstrate this, Rene maintains one can hear the result of a 2- to 3-dB change in amplitude of the first cycle of a tone burst. Impulse response measurements on speakers have not been available to the audiophile until recently when Audio began a new speaker testing and reviewing procedure. As yet not enough speakers representing each of the various categories of speaker design (e.g., dynamic cone, electrostatic, transmission line, etc.) have been tested to be able to draw broad conclusions relating impulse response to the characteristic sound quality of a design type. One preliminary conclusion Rene has drawn, however, is that most speakers employing enclosures suffer from multiple output pulses from a single input pulse. Usually this coloration can be attributed to the speaker box or cabinet either resonating or reflecting the sound internally.

Intimately related to impulse response is the phase characteristic of a speaker. Rene indicated phase problems are usually built into multiple driver speakers because of the separation of the virtual acoustic centers of the drivers. Even electrostatics that use low-, mid-, and high-frequency panels will not be entirely free of this type of phase problem when listening off axis.

In a listening room, most speakers will exhibit a frequency response containing many closely spaced peaks and dips of ±5 dB and more. Yet, according to Rene, these do not seem to be as important in characterizing a speaker's sound as the overall slope or trend in response. The bright characteristic of a speaker whose response rises gradually and is up 2 to 3 dB at 15 kHz tends to be more noticeable than local response deviations when compared with a flat speaker. Characteristics such as "forward", "dry", and "reticent" relate to these overall slope deviations. Rene's own speakers (Dayton-Wright electrostatics) have a measured 2-dB peak in the lower mid-range at about 800 Hz, which is readily detected by ear, indicating the ears' sensitivity to these small but broad frequency response variations.

Room Acoustics and Low-Frequency Response . One important aspect of recreating the effect of the original performance is reproducing a sense of the size of the hall where the performance took place. Rene explained that the sense of space around you is created by the natural or ambient sound fields which exist in that space. Out of doors, there are echoes from surrounding structures or hills (or perhaps no echoes on an open plain). In a concert hall the predominant sound field is produced by the low-frequency resonances of the room which are excited by outside noises such as traffic and wind. Measurement of the energy spectrum of a room like Symphony Hall would show maximum energy in the low-frequency resonances around 5 to 7 Hz, rolling off at 3 to 6 dB per octave with a minimum at about 10 kHz. It is these subsonic resonances that color your impressions of that room and cause you to identify it as a large room. A room with no energy below 50 Hz would be identified as small, like a closet. These subsonic "sounds" are actually perceived subliminally, since it would require well over 100 dB SPL at 5 to 10 Hz for them to be "heard."

When music is played this low-frequency energy is masked and would add nothing to the sound of an instrument such as a violin. However, the sense of the large or small room tends to be preserved in the reverberation time characteristic of each. The sound or spatial sense of certain instruments such as tympani can be affected when room resonances are excited at the frequency of drum head vibration or by the frequency of the impulses striking the head, which may be 3 to 10 Hz.

With this background, Rene suggested that reproduction of low frequencies may be more important in creating a sense of space than many may realize. But how can you accurately reproduce in a listening room the low-frequency information recorded in the concert hall if the room has strong low-frequency resonances of its own? Rene described some experiments he had performed in an attempt to reduce the effects of these room resonances by compensating for or canceling them using an auxiliary pair of speakers. Two Bozak woofers were mounted in windows at the rear of the room to form infinite-baffle speakers with a fundamental resonance of 30 Hz. The frequency response of the room was then measured with an SPL meter at the listening position, using first the front speakers and then the rear speakers as a source. Using the SPL at 100 Hz as a reference level, these curves were plotted and the dips and peaks were found to agree reasonably well in amplitude and frequency. This meant that it might be possible to feed the rear speakers an out-of-phase or time-delayed version of the front channel signals to cancel the effects of room resonances on the sound from the front channels.

Using only phase manipulation, Rene found that the rear speakers needed to be driven at nearly the same amplitude as the fronts and that the effect on the sound was to produce a rather ambiguous and unsatisfactory sonic image. He concluded that the scheme might be workable if <u>both</u> phase shifting and time delay could be used.

Although room equalization (or voicing) is another approach to suppressing room acoustics, Rene considered this solution to be lacking on two counts. It is possible to achieve satisfactory equalization for only one listening position, and when accomplished, it is often at the expense of amplifier power (to bring up those 12-dB dips). On the other hand, the time-delay technique should work for a fairly large listening area and would have a power requirement similar to that for front channels-

<u>Multichannel Reproduction</u>. If it is not practical to alter the low-frequency characteristics of our listening rooms, how are we to recreate the sense of space needed to reproduce the live experience from the recording? According to Rene, this may be possible by using more channels to give us information which suppresses the characteristics of the listening room. But how many channels are needed to do a credible job? After hearing an extraordinarily realistic demonstration of ambience recreation by the Ambiphon Company of New York, Rene is convinced that four channels should be enough if they are completely independent of each other (i.e., near zero cross-talk) and maintain a stable phase relationship.

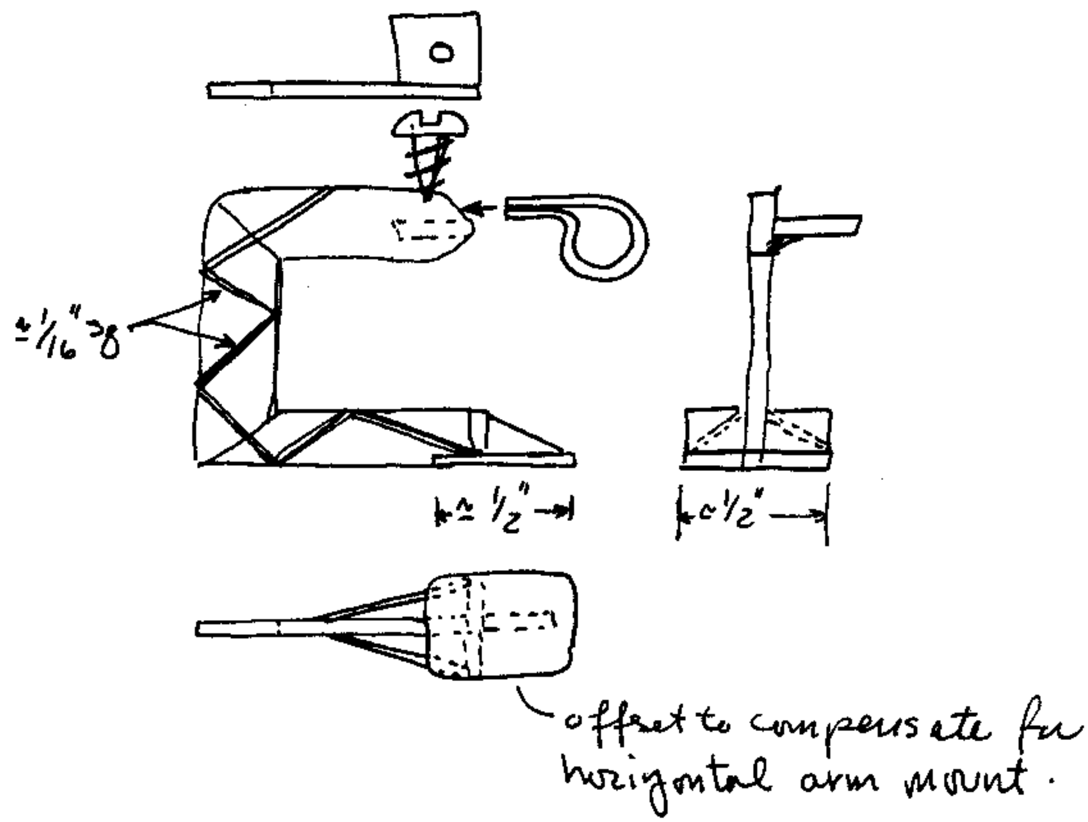
Ambiphon, whose slogan is "the sound of space," has produced some extremely realistic tapes of piano and band music using specially modified tape equipment. Rene had the opportunity to hear these tapes and reported that the listening environment was completely transcended. The characteristics of the recording hall, a large RCA studio, were readily apparent and bass transients could be felt "rolling by" and reflecting off the rear wall of the studio. Among the equipment used to produce these tapes, the key item was a state-of-the-art tape recorder having more than 80-dB dynamic range, a more than 90-dB S/N, 10-Hz to 35-kHz frequency response, and ±5-degree phase linearity. Realism was further enhanced by the absence of tape domain (inter)modulation noise (no fuzz around the piano notes). The recording setup employed four omnidirectional mikes set on the corners of a 15- by 20-foot rectangle with the performers off the 15-foot side. Playback was through eight AR-LST-1's set in the same configuration as the mikes, two per channel.

In tying it all together, Rene stated that the maze exists for the audiophile because he is lacking the measurements—those correlated with the psychoacoustic perception of sound—that would allow him to make reliable decisions about components based on their specifications.—John Schlafer.

Damping Update

Just before going to press, we received a note from Tom Mashey with some new information about Corn Huskers Lotion. It seems that some of the ingredients of the lotion evaporate and leave a rather sticky mess in the trough. To prevent this, Mashey suggests floating a thin (1/16-inch) film of mineral oil on top of the Corn Huskers. Also there is something in the lotion that attacks the airplane glue used to bond the balsa. Coat the balsa piston assembly with fiberglass resin, polyurethane varnish, etc., to protect the glue joints.

Mashey has changed the piston configuration as shown below for use with Corn Huskers Lotion. He points out that the 1/2- by 1/2-inch paddle may result in overdamping, but he thinks that with the SME/XLM combination this may be a good thing (but not for the SME/Shure Type III combination). Also, the damper seems to provide adequate antiskating as indicated by the Stereo Review test record tracking tests.



Improving the Thermo Electron 814 Microphone

Peter W- Mitchell

Since writing the introductory article on the Thermo Electron 814 electret microphone capsule (February Speaker), I have made many more recordings with these mikes in various acoustical environments, with a wide range of vocal and instrumental soloists and ensembles. In listening to the recordings on several different stereo systems, including mine with AR-LST's, it has become evident that all of the recordings exhibit a mild but consistent deficiency not due to the tape recorder. Fortunately the imperfection is easy to correct, and doing so effectively upgrades the 814 from a very good mike to a superb one.

In the earlier article I noted the suggestion (originally made by David Griesinger, one of the area's most respected recordists) that the 814 has a slight "absence" peak (i.e., a mild dip in response) in the upper midrange where many mikes have a "presence" peak. I now think he is right. The 814's midrange is impressively smooth, but perhaps a little too smooth. It is a deceptive quality; normally a transducer with an upper-midrange dip will sound dull, but the 814's lively high-frequency response prevents any impression of dullness.

Extensive listening elicits the impression that the 814's string sound has a slightly "feathery" quality due to the combination of strong extreme-top response with a muted mid-treble. And solo vocal lines, especially soprano, have a pleasing sound but seem slightly remote and unfocused with the diction not quite clear. Of course these remarks are not a damning criticism of the 814, which at its price is still an excellent bargain.

But I find that this imperfection is decisively eliminated by a 3-dB boost of the 3500-Hz control on the old Metrotec FE-1 equalizer. This substantially improves the subjective accuracy of the timbres of both voices and instruments. Vocal lines come into crystal-clear focus and the "rosin" is restored to string sound. Of course this news isn't much help if you don't have that equalizer, and in any case it is a pain in the neck to have to post-equalize a tape every time you want to play it or dupe it. Ideally, corrective equalization should be applied at the time of recording, so as to produce the best possible tape. (Ed. Note: Applying this equalization will also improve signal-to-noise ratio by the amount of the boost in the ear's region of greatest sensitivity.)

You wouldn't want to have to carry a bulky multiband equalizer around to every recording session, but since the equalization required to upgrade the 814's sound is rather simple, I have put together a compact, battery-powered, fixed-boost equalizer for the purpose. First I found by experiment the setting of the Metrotec equalizer's controls that produces the best sound from recordings made with the 814's; then I measured the Metrotec's actual response, and designed the new equalizer to duplicate that response.

Before describing the 814 equalizer, a detour for a question: Is the 814's apparent mild "absence" peak real? Thermo Electron's published curves and the typical 814 curves run in TE's test chamber (March Speaker) show no evidence of an upper-midrange valley. Is there something wrong with TE's test setup? Or is there some psychoacoustic factor in two-channel recording and playback that requires some "presence" boost for subjective realism? I don't know. But in audio, if something sounds right, it is right, and the addition of this equalization does improve the already good sound of the 814.

The 814 equalizer circuit is based on one of a collection of op-amp circuits published by Edward Gately in the June 1970 issue of Audio. It employs the audiophile's friend, the type 741 op-amp IC (see the September 1974 Speaker). The basic schematic for a fixed-boost equalizer is shown in Fig. 1.

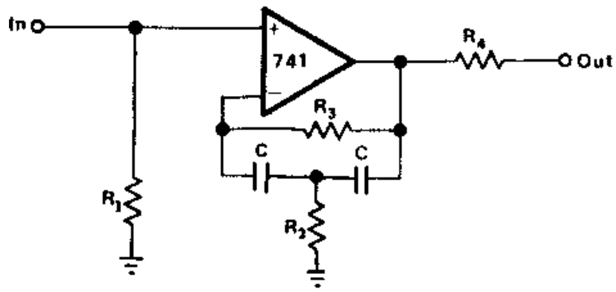


Fig. 1. Basic circuit

In this circuit, R_1 sets the input impedance of the circuit. Note that, unlike many op-amp circuits, the input signal goes to the noninverting input of the IC, whose own impedance is in the megohm range. The feedback network is a "bridged-T" filter. Its frequency of peak boost is set by R_2 and the two equal capacitors: $f = 160{,}000/R_2C$, where C is in microfarads. R_3 bridges the filter and thus controls the amount of boost. R_4 sets the output impedance.

The 741 isn't quiet enough to be placed ahead of the mike preamp, so the circuit will be connected to the line inputs of the tape recorder or noise-reduction encoder and will be driven from the mike preamp or mixer. (Of course the equalizer may also be used at the output of the recorder for listening to any tapes made without its benefit.) A load impedance of 10K is suitable with modern mike preamps and mixers, so R_1 = 10K. The circuit should have approximately unity gain, so R_2 should also be about 10K. With R_2 = 10K, a boost frequency of 3500 Hz requires capacitor values of 0.005 microfarad (5000 picofarad); since the desired boost curve is a broad and shallow one, the exact peak frequency is not critical, and any capacitors in the range from 0.004 to 0.006 microfarad would be usable. By experiment I found that the value of R_3 which gives the desired peak boost of 3 dB is 8.2K. The value of R_4 can be any moderately low resistance; I chose 1000 ohms. If you will never short the two channels together to make mono, you may eliminate R_4 entirely.

The 741 operates from matched positive and negative power supplies in the 8- to 18-volt range. The current drain is 6 mA (3 mA per channel), so ordinary 9-volt transistor radio batteries will last 15 to 20 hours, i.e., a couple of months for the occasional recording that most of us do. For full-time recording, an ac supply would be preferable, such as the one described in Fig. 25 of "IC Op Amps. . ." (September 1974 Speaker).

The finished 814 equalizer has the following typical performance specs: gain = 1.0; nominal output, 1 volt rms; maximum output, 5 volts rms; S/N, 100 dB below clipping (85 dB re 1 volt); THD at 1 volt, 0.05%. The measured frequency response is shown in Fig. 2.

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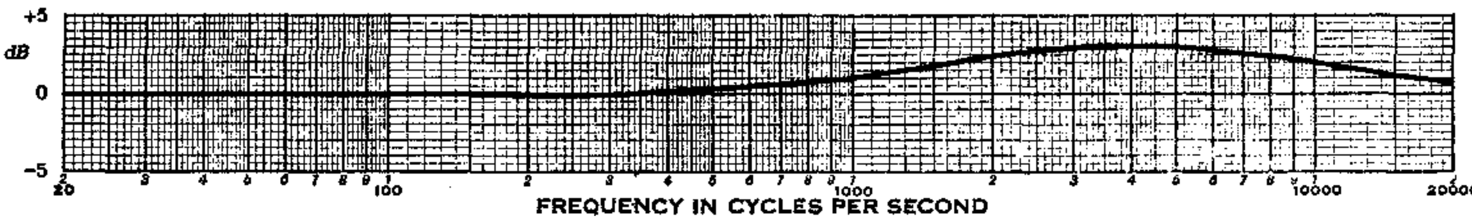

Fig. 2. Equalizer frequency response

Construction Hints. Type-741 op-amps come in several forms. The most convenient form for this circuit is the Signetics 5558V, a dual 741 providing both stereo channels in one 8-pin miniDIP package. It is available from Lafayette Radio for $1.30, Radio Shack for $1.50, and from discounters for half of that. With this IC the entire stereo equalizer can be constructed on a small Radio Shack printed-circuit board (called a "socket adapter"). The board comes etched and drilled to accept an 8-, 14-, or 16-pin IC socket, and each IC connection leads to a square copper-clad pad which you can drill (using a no. 60 bit) for installing wires, resistors, and capacitors. A recommended parts layout and wiring diagram is shown in Fig. 3 (as seen from the upper or nonclad side of the board). The IC pin numbers are labeled in the appropriate pads.

Note: the uppermost pair of predrilled IC holes are not used for the IC socket; pin 1 (indicated by a dot on the socket and a dimple on the IC) is installed in hole number 2. If you can find an 8-pin IC socket, install it as shown in Fig. 3; otherwise use a 14-pin socket and install the socket in the lower 14 of the 16 available holes. The IC then would be installed in the socket's upper end.

If most of the parts are brought from Radio Shack, the total cost of the 814 equalizer is about $12. If a group purchase is desired, I will accept prepaid orders. For an additional $1, I will supply the PC board fully predrilled for all of the components. And for an additional $7 (i.e., $20 total), I will supply the complete circuit assembled on the board and tested; you then drill the box, mount the input/output sockets, test jacks, and switch, and install the circuit board and batteries.

Suggested Parts List

	Radio Shack	
5558V IC	276-038	$ 1.50
PC board	276-024	2/1.00
DPDT switch	275-1546	1.90
4 RCA phono jacks	274-346	1.50
2 9V battery clips	270-325	5/1.00
2 50-μF, 16-WV electrolytic capacitors	272-1004	1.20

Plus an 8- or 14-pin IC socket, a box chassis, battery test jacks, batteries, 4 0.005-μF capacitors, and 8 resistors (4 10K, 1/2-watt, 10%, 2 8.2K, 2 1000 ohm)

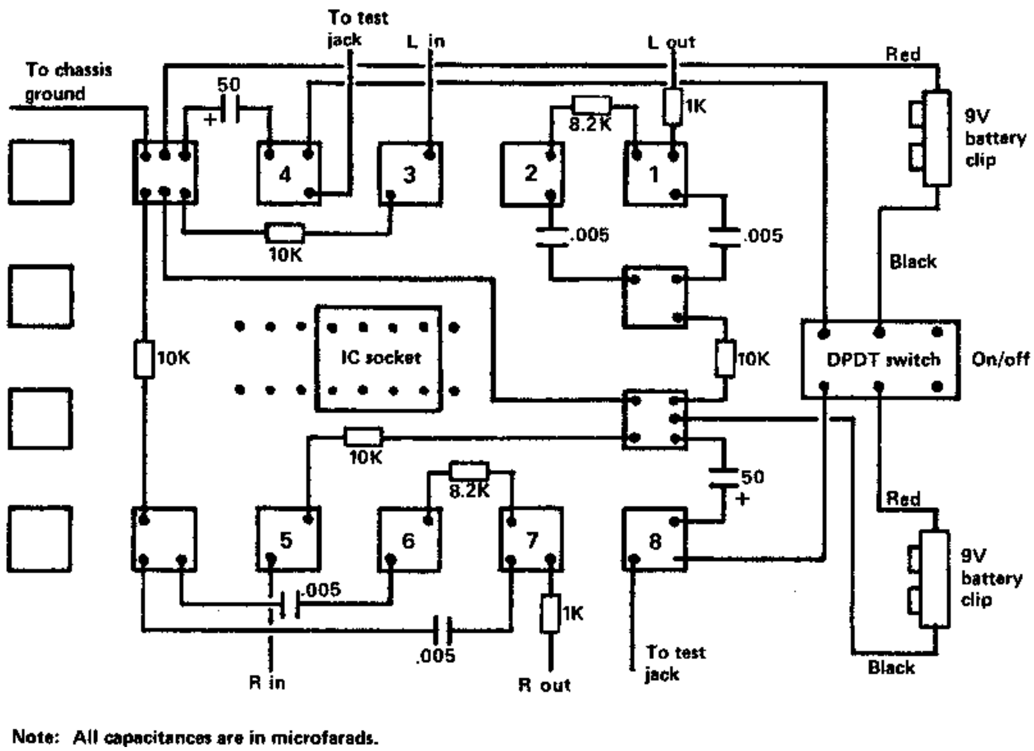

Fig. 3. Parts layout and wiring diagram

A BAS Test Report

Tuners: Pioneer TX-9100 Versus McIntosh MR-78

Alvin Foster

Since the published specifications of the McIntosh and Pioneer tuners are so similar, I doubted that an A-B comparison would show much difference. To a degree this was true, but some of the results were confusing.

Features

The two tuners have in common: multiplex noise filter; fixed- and variable-volume outputs; separate tuning meters for center-of-channel and for signal strength; a variable sensitivity muting system; linear tuning dials; oscilloscope outputs; provision for 75- and 300-ohm antenna inputs; and automatic switching to mono when signal strength is low.

The Pioneer ($350) has separate volume controls for AM and FM and an impulse noise-suppression circuit. Its headphone amplifier is capable of driving low-impedance phones.

The McIntosh ($800) has a three-position selectivity control, a switch that converts the signal strength meter into a multipath indicator, a function limiting reception to stereo broadcasts, a high/low gain antenna-input jack, and a two-level panel light dimmer. The scope outputs are easier to use than the Pioneer's for center channel tuning and multipath detection, and additionally, the scope outputs provide a visual indication of incoming signal strength.

The tuners were tied into a splitter, connected by 300-ohm shielded twin lead to a roof-mounted Finco FM-4G antenna.

How They Performed

Summarizing how these two tuners performed is very difficult; the results were ambiguous. For example, the Pioneer in general was more sensitive and quieter on weak stations while the McIntosh, with its selectivity control (which narrows the bandpass and which increased distortion) was more often a better performer when it came to picking out closely adjacent, distant stations, e.g., WVBF, 105.7 MHz, and WHCN, 105.9 MHz. But in other instances the results could go either way.

For example, WOTM and WWON both broadcast at 106.3 MHz, have 3,000 watts, and are about equidistant from the test site. This would seem to be an excellent test of capture ratio; the better tuner should fix on the stronger of the two stations and ignore signals more than 1 or 2 dB down. However, the tuners did not pick up the same station. The Pioneer fixed on one station and the McIntosh locked in on the other. Neither tuner was capable of receiving the other station, and when the antenna was rotated, each tuner picked up its respective station more clearly. Two theories have been offered for these results: perhaps the tuners were "talking" to each other, i.e., local-oscillator signals from one (or either) of the tuners might have been leaking via the splitter into the other tuner causing the RF stages to misbehave. The other theory was that both tuners are so close in performance that mixed results are all one can expect.

Results alternated even when it came to comparing quite obvious features such as signal-to-noise ratio. Unless dealing with particularly weak or closely adjacent distant stations, the results could favor either tuner. However, the Pioneer had less hiss on the greater number of stations.

One way to test for multipath rejection is to fix on a station and then rotate the antenna- The tuner that pulls in the station best throughout 360 degrees will often have the better multi-path rejection ability. The results of this test also were mixed: sometimes the McIntosh was superior, but more often the Pioneer tended to hold the station longer. This test for multipath rejection could easily be confused with one for sensitivity. To minimize this possibility, we chose strong stations that suffer from severe multipath only when the antenna is not aimed properly.

To test for SCA rejection, the tuners were tuned to WCRB at 102.5 MHz. Both tuners performed equally well and exhibited no deterioration of signal quality.

The muting on each tuner was impressive. Neither tuner exhibited any noise between stations, and when a station was tuned in, it would suddenly appear; otherwise there was dead silence. The McIntosh tuned more smoothly (mechanically) than did the Pioneer, but both tuning dials were precise in indicating frequency. The switch on the McIntosh which converted the signal strength meter into a multipath indicator seemed of no value since poor multipath rejection was audible long before the meter reacted.

The signal strength meters on both tuners were only marginally useful in aiming the antenna. They either indicated maximum signal strength or an inadequate signal level. Unfortunately no manufacturer seems to want to design a signal strength meter that reflects the typical levels fed to a tuner.

How They Sound

For listening tests, BAS member John Emerson (owner of the McIntosh) and I chose a "Victorized" live Boston Symphony Orchestra broadcast. ("Victorized" refers to the process, popularized in Boston by Victor Campos, in which the signal running between Symphony Hall and the transmitter is Dolby-A encoded and the FM transmitter's compressor and limiter are bypassed.) Both tuners seemed equally quiet. However, while the orchestra was not playing, the McIntosh produced a whistle, audible at extremely high volume levels. An AC-VTVM connected to the audio outputs of the tuner indicated that the whistle was above the background noise of Symphony Hall. A scope connected to the audio outputs showed that the signal was being modulated by the 19-kHz stereo pilot. None of this occurred with the Pioneer. To keep things in perspective, the whistle was audible only when the background was very quiet and the volume control on the amplifier was turned beyond its normal setting.

We next tuned to several stations while comparing the two tuners to see if there was a difference in audible distortion. We first matched levels and then used blind A-B testing. Each person was asked to indicate only the tuner he thought sounded best or the one he thought had the least distortion. Strong multipath-free stations were used so that the effects of background hiss could be eliminated.

The Pioneer was selected well above chance as having less audible distortion. On most stations, however, there was no discernible difference. In fact, to keep things confusing, the differences were most noticeable when the music being played contained cymbal crashes or other extreme high-frequency information.

Two explanations were offered for this inconsistency. First, perhaps a minor frequency-response dip or rise accentuated high-frequency information in one of the tuners. In order to eliminate the possibility that a minor frequency response variation could explain the difference

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between the tuners, the preamplifier's high filter was switched in, rolling off information above 9 kHz. The Pioneer was still selected as having less distortion. Such a frequency-response variation between the two tuners should not have allowed them to sound identical on most stations.

The second explanation speculated that perhaps the 19-kHz stereo pilot was intermodulating with the high-frequency content of the music and producing a slight audible signal degradation in the McIntosh. This seems a bit more likely.

In summary, the tuners were audibly identical most of the time, as one would expect from their specifications, but on a few occasions the Pioneer was cleaner and more transparent than the McIntosh. However, our mixed results indicate to us that all the important parameters for measuring tuner quality do not appear on the manufacturer's specification sheet.

White and Pink Noise Revisited

Rene Jaeger and Alan Southwick

Noise Source

For those of us with FM tuners, there is an easier way to derive white noise than by using the sensitive, high-gain circuit described in the article, "White or Pink: Adding a Little Noise to Yom Life" (January Speaker). The multipath or horizontal-oscilloscope output on most modern tuners provides an excellent source of white noise when tuned to interstation hiss. If no such output is available, a "re-emphasis of the pre-emphasis" can undo the 75-microsecond rolloff (reshaping the "red" noise from the audio output). Either way the signal then is fed through a "pinking filter,' yielding pink noise. We can do this by modifying a portion of the January circuit.

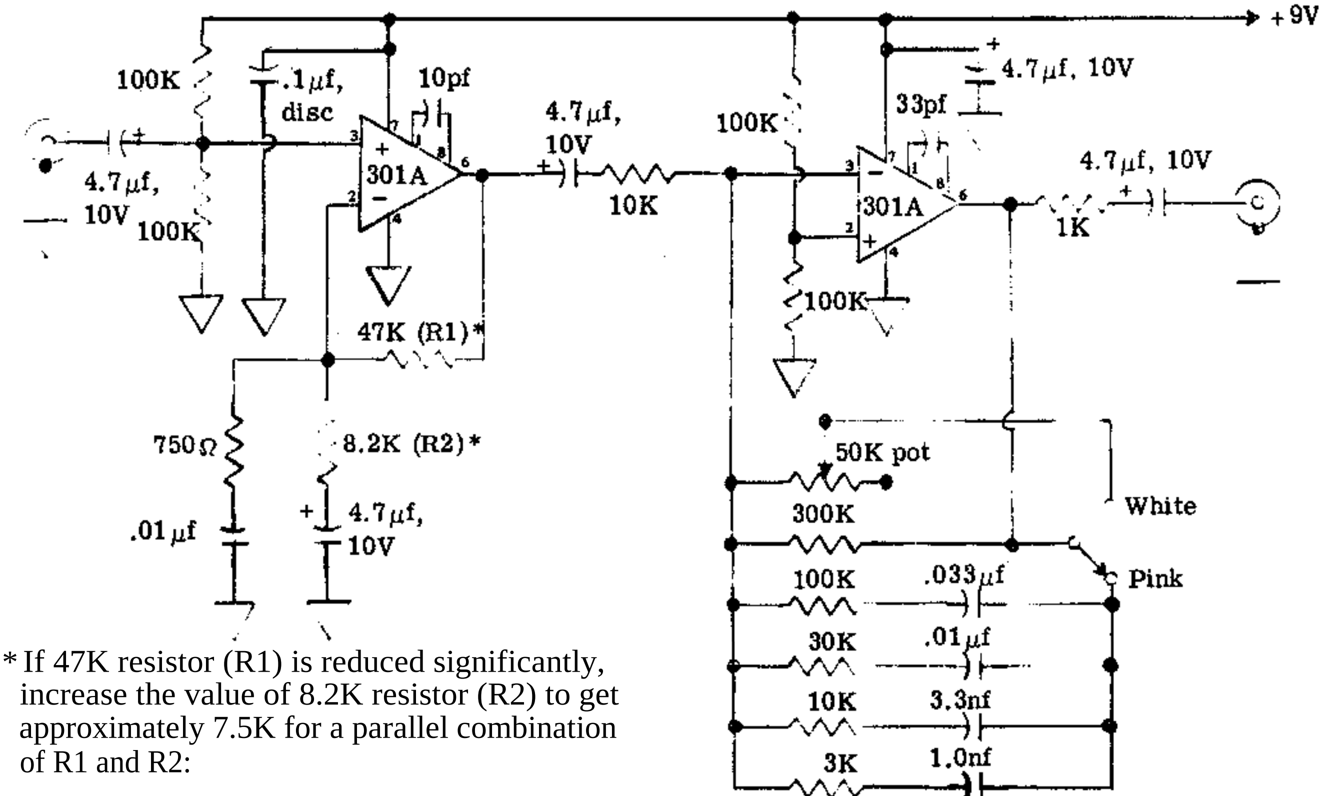
In the earlier circuit design the generator section is somewhat difficult to construct, owing to the large voltage gain required to raise the microvolt noise of the resistor to a useful level, and is not therefore recommended for inexperienced constructors. Even the experienced may find it difficult to keep the stray capacitance from shunting the noise source and rolling off its high-frequency output.

We can avoid the difficult generator portion of the circuit by utilizing another source of noise—as noted, your FM tuner. The audio output of most stereo tuners (in the mono mode) is white noise, flat from about 50 Hz to 2 kHz and rolled off at 6 dB per octave from there to about 15 kHz. Above 15 kHz, multiplex filters may sharply attenuate noise. (If you have an old mono tuner, this will not apply.) Low-frequency response seems to be deficient in many tuners, but some may be good down to 20 Hz.

To get pink noise from this output we require a network whose response rolls off at 3 dB per octave from 50 Hz to 2 kHz and rises 3 dB per octave from 2 kHz to 15 kHz. This gives an output that rolls off at 3 dB per octave across the audio spectrum.

If we need white noise, then we first must undo the 75-microsecond de-emphasis of the tuner and then feed the broadband pinking filter (as in the previous article). To do this we need a network whose response rises 6 dB per octave above 2 kHz.

Figure 1 is a schematic of the white-noise shaping network plus pink-noise filter to be fed from an FM tuner output. The tuner must be set to a gap in the broadcast spectrum (usually the ends of the band are best) and the antenna disconnected, preferably. If your tuner has exceptionally high output, the value of the 47K resistor, R1, should be reduced to avoid overloading the filter. Generally, the pink noise should sound as loud as the noise directly from the tuner; adjust the 50K white-noise output pot for equal loudness of white and pink noise.



* If 47K resistor (R1) is reduced significantly, increase the value of 8.2K resistor (R2) to get approximately 7.5K for a parallel combination of R1 and R2:

R1	R2	Relative Output
33K	10K	- 3 dB
22K	12K	- 6 dB
15K	15K	- 10 dB

Fig. 1- Schematic of white-noise shaping network and pink-noise filter

Additional Applications

In addition to some suggestions mentioned in the earlier article, here are two specific things to do with a white/pink-noise generator. To start with, check out your tape recorder by feeding some pink noise to it at -20 VU (to allow headroom for peaks) and compare the recorded noise to the original signal- A properly adjusted machine should sound identical to the source.

Secondly, armed with a sound pressure level (SPL) meter, a pink-noise generator, and an equalizer of some sort (from a simple three-band one to an elaborate 1/3-octave equalizer), one can effectively equalize a listening room for reasonably flat acoustic response.

Initially, feed pink noise at a comfortable listening level through your speakers and note the SPL reading on the SPL meter at your listening position. Now switch in your equalizer with all bands set to minimum level and then raise each band's level individually, jotting down the level indication that produces an SPL reading 3 dB lower than the direct pink-noise feed. Reduce the original band's level back to minimum and repeat the procedure for each band on both channels if possible. Once all the individual band settings are noted, reset all the controls to their respective recorded settings and compare this newly arrived at, equalized level with the straight pink-noise SPL reading. Raise or lower all the controls simultaneously by the same amount until the equalized output is equal to the straight pink-noise feed. This procedure will effectively "equalize" the listening position where the SPL meter is located. For a broader equalized zone, move the meter to several predetermined locations for each band adjustment and average the equalizer settings for the -3-dB SPL reading.

The accuracy of this procedure is limited by the errors in your SPL meter's response. For example, if you are using the Radio Shack meter, use your ears instead of the meter to equalize above 9 kHz and below 100 Hz.

Generally, the ends of the spectrum require the greatest boost, but don't be upset if you can't get both "tails" to exactly equalize to the proper SPL. For small rooms this could require about 20 dB boost at 30 Hz (a 100-fold increase in power at this frequency). Beware of excessive bass and treble input to your speakers; if the manufacturer recommends fuses, by all means use them even if you have a low-power amplifier. At first, the extreme high frequencies (from 5 kHz up) may sound excessively bright but remember that the microphones used were not placed in the tenth row but probably shoved right into the performer's instrument. If listening in your equalized room proves disconcerting, the adjustments can of course be tailored to suit your personal preferences.