THE B.A.S. SPEAKER

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In This Issue

This issue is chock full of audio goodies. Peter Mitchell has pulled together the data from the BAS tuner clinics and has evaluated the results, which are surprising indeed in some cases. It appears that really good tuners are available for under $200. And after deciding what tuner you want, you can turn to Steve Seto's antenna survey and figure out what you should hook it up to.

For those of you who have bought or built outboard phono preamps, Charles Pike offers an approach to adding gain and switching controls. And Dan Shanefield offers some pointers on how to properly equalize your room.

The newsletter contains all the usuals: two meeting summaries (one on the hottest new item in audio—time delay—the other on what's cooking at AR these days), a few more equipment notes, and some odds and ends of current interest.

Finally, the Society is going through a number of changes, not the least of which are in Speaker personnel and production facilities. We ask that you bear with us (if necessary) as we shake down the new operation.

Membership Renewals

Because the amount of dues for next year cannot be established before the September meeting, we ask that you do not send in renewal checks at this time. Dues information and a new application form will appear in the October Speaker. All 1975-76 members will receive the October issue.

Membership dues are $12 per year (October 1 to September 30) or portion thereof. Dues include a one-year subscription to the BAS Speaker. (Note that almost the full amount of dues is allocated to production of the Speaker. The local activities of the BAS are strictly self-supporting.) For further information and application form, write to: The Boston Audio Society, P.O. Box 7, Kenmore Square Station, Boston, Mass. 02215.

For Sale

- Magnecord 1028-24, mint condition, 101/2-inch NAB reels, 2 track, 7 ½ and 15 ips. Less than 700 hours total use. $895. Tom Tyson, 707 Florham Drive, Highpoint, N.C. 27260.
- SAE Mark XXVII equalizer, mint condition. $400. P. Vendegna, (617) 964-1020.
- McIntosh MR71 FM stereo tuner, newly aligned, good condition, with case. $500 new. Best offer. Martin classical guitar model CR-7, nylon strings, excellent condition. $250 or best offer. Al Southwick, P.O. Box 601, Penacook, N.H. 03301, (617) 663-8340.
- Pioneer 1020L open-reel tape recorder, $450; Harman-Kardon Citation 16 power amp, $530; SAE Mark IVDM power amp, $395; Crown IC150 preamp, $245; Bose 4401 preamp, $295; Sony TC280 (used once), $210. Shipping included. Dick Denise, (617) 358-7949.
- Phase Linear 1000 autocorrelator noise-reduction unit, $265. Teac A1200U tape deck, three motors, three heads, best offer. Dick Glidewell, (617) 342-8468.
- Supex SD-900E moving coil and Levinson JC-1. Cartridge has about 6 months use, diamond is OK; pre-preamp has TAS modification for flatter midrange. $110, Steve Seto, (213) 970-4791 days.
- Audio Research SP-3a-1 with new Amperex tubes, $500; KMAL mercury contact tonearm with Lyrelift and Linn-Sondek mounting board (if you want it), $100 takes all; Stax SRD-5 energizer box, outlets for Stax phones (SR-3's, SR-5's, SR-40's, and SRX's), $25; Tandberg TCD-310, as is (needs some work on end-stop mechanism, but heads and audio are fine), $250. Lewis Dalven, (617) 273-1105 days.

Wanted

- Pioneer TX-9100 tuner; Kenwood Audio Display Scope; Teac AN-180, AN-300, or Sony NR-335 Dolby unit. Dick Glidewell, (617) 342-8468.
- Dynaco Mk. IV/PAM-1 combo in working condition. Lewis Dalven, (617) 273-1105 days.

At Last, the Ultimate Speaker

I have found a speaker driver that might live up to the fantastic Tawdry Turtle amplifier described by Bob Graham (April 1976). Building 19 advertises a 2-inch speaker, two for a dollar, which "gives deep and powerful, extended treble." They deserve the Atlantis Sound award for restraint in advertising. It brings to mind a recent Building 19 sale of what appeared to be JBL sculpted foam speaker grills beneath a sign proclaiming "FOAM FURNACE FILTERS."

— Jim Nichol (Massachusetts)

In Search of the Ionovac

The article, "Improving the High-End Sound of Your Loudspeaker System," by Robert Graham (Jan. 1976), has provoked me to the point of purchasing an Ionovac tweeter. I have located a source for the Ionofane 601 ionic tweeters but think that the price ($353 per pair) is rather steep. Shortly after rereading Graham's article, I became resigned to the price and I'm willing to pay it for two units in top condition. I would like to know if the Ionofane 601 and Ionovac are the same units, and does Mr. Graham have a source for used Ionovacs?

— Roger Kisner (Virginia)

Bob Graham tells us that the Ionovac and Ionofane are the same and that reconditioned used ones ma be available for $175 from Gil Custom House, Inc., 8813 W. 95th St., Palos Hills, Ill. 60465. — Ed.

Information Wanted on Electronic Crossovers

Has anyone had experience with the Audio Amateur electronic crossover or with amps to drive a pair of FMI 80/RTR ESR-6 sets with a common M&K woofer and Stereophile modifications to the RTR's? I am interested in single amp, bi-amp, or tri-amp and would prefer a solution that is not too expensive. (I believe a Dyna 400 would be a decent single amp, but I already have a Stereo 70, which is supposed to be excellent with the RTR's.) — Bob Sellman (New Jersey)

Teac TLC

In the continuing search for good service, I can recommend A&M Service Center (137 S. La Brea in Los Angeles) for servicing Teac cassette decks. I took my 450 to them for general maintenance and re-biasing. Despite the fact that my deck was at the ragged edge of its warranty, they performed several warranty repairs, checked the deck for spec compliance, and charged me less than $40 to completely re-bias the deck and service it.

Because this was the first time I had dealt with them, I had the work checked by an impartial third party, a dealer who does not handle Teac, but who is very good with tape decks. He reported that the work had been done competently. Considering their reasonable fee and their good work, I feel that A&M is worth recommending. — Steve Seto (California)

The End of an Era

RCA has closed its tube manufacturing facility; its contents are being sold at an industrial auction. If you want an absolutely reliable source of 12AX7 tubes for your Marantz 7c, this is the only chance you'll ever have of going into the business for yourself.

Professional dbx/Dolby Replacement Cards

The dbx Corporation has announced a plug-in card replacement for the professional Dolby-A noise-reduction system. This will allow studios with Dolby-A systems to have dbx noise reduction available to its customers at reasonable cost, only $250 per channel. Hopefully, this will speed the spread of quiet disks to us hiss-hating audiophiles.

A Closeout Sale on AKG Mikes

Norlin Music, Inc., of Chicago is liquidating certain AKG mikes made for Chicago Music Industries, which Norlin bought out. These carry the AKG name, packaging, and full factory warranty. They are being sold at approximately one-fourth of their original retail prices, which makes them an unusual bargain indeed. Both condenser and dynamic types are availabe in both high- and low-impedance versions. Unit prices range from $46 to $64. Member Dick Lewis has purchased and used some of these and has offered to advise other interested members and to coordinate a possible BAS group purchase. A variety of Moog synthesizers is also being moved out at prices from $250 to $750. Call Lewis at (617) 899-3250 for more information.

Discounts on dbx

If you are interested in buying dbx noise-reduction equipment, Illinois Audio (12 E. Delaware Place, Chicago, Ill. 60611), a mail-order discount house, recently quoted the following prices to me: dbx 119, $133; dbx 124, $256; dbx 157, $476. They also told me that they might be willing to give a further discount of as much as 2% for an order of six or more units, so perhaps some Boston area members could save even a little more money. I am a strong believer in purchasing by mail (the factory does back up the warranty) and have had no real problems, except for occasional delays, in purchases made over a period of about 12 years. Although I have not personally purchased from Illinois Audio yet, I have been told by other people that they are a good, reliable firm. They also sell the Soundcraftsmen PE 2217 preamp for about $330!

— Bob Sellman (New Jersey)

Mail-Order Miseries and What To Do About Them

I have had very poor service from two mail-order businesses. The first, E&E Audio, 2816 Church Ave., Brooklyn, New York 11226, sells speaker drivers through a classified ad in Audio. After waiting one month, I wrote inquiring about my order. I received no reply. In succeeding months I wrote several times to E&E Audio as well as to Audio magazine. After 31/2 months, I finally received one speaker with a refund for another, which as back ordered. The second business is Delta Electronics of Amesbury, Massachusetts. A similar situation occurred over a five-month period, with a refund eventually forthcoming.

As a result of these experiences, I have become very adept at prying refunds loose from obstinate mail-order firms. First, forget the new guidelines issued by the Consumer Protection Division of the Federal Trade Commission, which require a mail order to be filled within 30 days or notification by card, etc. Writing the FTC will only get you a form letter saying they are very busy and can't bother with mail-order problems amounting to less than a few thousand dollars. If you are going to be had, be had for lots of money.

A better solution is to write the Mail Order Hot Line, 6 East 43rd St., New York, New York 10017. This service is a Better Business Bureau of mail order and will send a nasty letter to the offending business. This is also the place to write to get off of junk mailing lists or (perversely) to get on a junk mailing list. Another useful tool is a letter to the magazine carrying the ad, as well *as* filing mail fraud charges with the post office. — Jim Nichol (Massachusetts)

Another View of Delta Electronics

Over the last year I have had considerable dealings with Delta Electronics, but never by mail. I've spent many Saturday mornings browsing in their warehouse and have discovered some remarkable bargains. They are the best source of power transformers, heat sinks, and relays that I've found anywhere near Boston.

The owner, Melvin Weiss, and the salespeople have always been friendly and extremely helpful. Occasionally I've had to return items because they were not as specified in the catalog (something one has to expect from surplus houses), but refunds or replacements were readily forthcoming.

Having done business with Delta and most of the other surplus houses in the Boston area, I strongly recommend the personal approach rather than mail order. But if you must order by mail, attach a note to the effect that you prefer a partial shipment with refund rather than waiting for out-of-stock items to be back ordered. — Bob Borden (Massachusetts)

Win STD-10 Strain-Gauge Cartridge

The Win SDT-10 strain-gauge cartridge is a lightweight pickup (less than 2.5 grams) using semiconductor strain gauges to modulate current from the power source. The output of this unit is very high (stated to be 0.8 V rms at 3.54 cm/sec) and RIAA equalized. It is connected (via its power source) to any high-level input of the preamp or directly to the power amp if the power amp has level controls. No controls are provided on the power source. It is an amplitude sensing device with moving elements said to be of very low mass. For further description of the operating theory and specifications along with some critical evaluations see Sound Advice, Vol. 1, Nos. 1 and 2, and Audiogram, Vol. 1, Nos. 1, 2, and 3. The cartridge and power source cost $225.

Several of us here in California were in the market for a new cartridge. We finally succeeded in contacting Dr. Sao Win of Win Laboratories. He offered to send one of his latest versions (April 1976) to try out and compare with other cartridges. This one, he said, has much flatter frequency response than earlier models and no tracking problems using a stylus force of 1.6 grams. We were advised that our Formula 4 arm was among those considered ideal to mate with the Win. Advice was also extended to bypass any preamp for best results.

The Win/Formula 4/Thorens TD-125 was auditioned as a unit in four audiophile systems. Most of the listening was done bypassing preamps altogether. Here are a few findings: High-level massed violins sometimes become flat, not necessarily harsh or strident, but there was a meshing of separate sounds into one big whistle-like sound (Shostakovitch, Symphony No. 8, EMI ASD 2917). There was a minimal sense of air, space, and openness about the sound—as if a Windex-cleaned window had been closed. The Shostakovitch is merely an example; we listened to many more works. The overall impression of classical music was that a peculiar dullness had settled on the orchestra. On the other hand, single violins and double basses were highly musical (Vivaldi, Seasons, Argo ZRG 654, and Rossini, Sonata No. 2, MHS 1166).

The redoubtable trinity—perspective, image stability, and instrumental focus—was exemplary and new musical aspects of old favorites were consistently revealed. There was, however, a slight forward shift in location of solo violin and trumpet as notes ascended the scale (Bach, Milstein, DGG 2563 415/17, and The Sound of Musical Instruments, Acoustic Research Demo Record, Vol. 1). One new favorite (Pigs Eye Jass, Fidelity First, Vol. 2, Insight Records) was a stunner when played with the Win. Nothing seemed lacking or amiss. There were moments when the thrill of live sound in the room would grip the psyche. Indeed, the Win would have won had this been the only record heard.

Bells and cymbals, at times, mistracked or mistraced (John Renbourn, Sir John Alot, Reprise 6344; M&K Sound, The Bottom End, Vol. 1; and the Shure Trackability Test Record). If the recording level is low enough not to cause breakup, hard transients are crisp and clean with what seems to be the right amount of ring and shimmer. Sibilants were not exaggerated or spiffy, but on occasion would leave the singer and spatter harshly into the room (Joan Baez, Farewell, Angelina, Vanguard VDS 79200). Singing voices, male and female, were not always entirely natural in tonal quality. Surprisingly, a couple of our torture discs passed the tracking/tracing test (Miles Davis, Sketches of Spain, Columbia CS 8271, last few minutes side 1, and Moussorgsky, Pictures, Organ, MHS 1472, last few minutes of side 2).

Bass definition and solidity are winners with all types of records and music, especially if the rising low end output can be flattened somewhat with an equalizer. Inner clarity and detail were maintained to a high degree on thickly textured material (Rachmaninov, Symphony No. 2, EMI ASD 2889). Every record, except those with exaggerated stereo separation, presented a broad stereo stage with optimum center fill. There was no right/left clustering of instruments.

We were baffled by a hum that persisted even after we had tried all the usual hum-reducing tricks. It turned out that the cartridge is sensitive to light. The hum completely disappeared when the turntable light was switched off. This phenomenon can be verified by passing a flashlight beam rapidly back and forth across the cartridge—loud thumps issue from the speakers with volume well up. For further verification, leave the turntable light on and shade the cartridge with a piece of cardboard--the hum vanishes. Other extraneous noises such as surface noise were less obtrusive than with the ADC XLM II, which is itself a quiet cartridge.

If the foregoing seems inconclusive, it is because the Win has such tremendous potential and so many positive attributes that it is hard to accept its imperfections. If only those violins were sweet and airy, the singing voice more convincing, and the tracking problems cured!

— Dow Williams (California)

[Ed. Note: It is widely understood here that the Win cartridge is based on the 1960's Euphonics design. Apparently Dr. Win has bought rights to Euphonics' patents—which are also said to be near expiration—and made relatively minor changes in the design, not that it was unsuccessful to begin with. Further, there seems to be a problem in the marketing of the cartridge, an agreement to sell through ESS having fallen through. Finally, Win is said to be having trouble with unit-to-unit repeatibility. — J.B.B]

Sound Guard/Discwasher Compatibility

Because several BAS members are investigating Sound Guard, I decided to determine if there is any interaction between Sound Guard and the Discwasher brush and fluid. I called the Washington, D.C. office of Ball Brothers and talked to Mr. Jack Deitrich of the technical staff. Deitrich told me that they had tested Sound Guard with the Discwasher and had found no deterioration of the Sound Guard coating as a result of Discwasher use; as well, they had found no deleterious effects on the Discwasher. Though it was not stated, I gained the impression that Discwasher was not the only record cleaning device so tested and that the results were all similar.

In response to my comment that record manufacturers might use Sound Guard on new discs, Deitrich indicated that they had approached manufacturers with the idea, and they were not particularly interested: their concern was the sale of records, not prolonging the life of records already sold.

Ball Brothers' findings indicate that recoating is not necessary as often *as* specified in the directions; allowing for a light application and building in a safety factor, they do recommend recoating after approximately 25 playings.

After talking with Deitrich, I spoke with [BAS member] Dr. Bruce Maier of Discwasher. Dr. Maier was not only aware of Sound Guard but of another product as well—Disc Protect, made by Canyon Products in Boulder, Colorado. Maier said there is nothing special, proprietary, or magical about the ingredients or action of the teflon derivatives used for friction reduction. He does, though, object to the mode of Sound Guard's application.

He feels that it is extremely complex and inexact to hold the record (either in one hand or on a surface), spray it with a pump spray bottle 9 to 12 times per side, and <u>burnish</u> it with some sort of pad. This is particularly difficult when one has been trained not to touch the surface of a disc. When you spray the fluid in this manner, you not only have considerable waste due to the 9 to 12 splotches of fluid that form, but you must burnish thoroughly to spread the chemical <u>evenly</u> across the surface; for unless the chemical is spread evenly and thinly, it can <u>raise</u> the noise level in areas of improper application. Maier explained that in these cases, the chemical sits on the disc surface as small globules (he's seen such under electron microscopy with the Sound Guard product), and if not properly smoothed, the globules act like small dust particles themselves. His experiments with Sound Guard find enough fluid in the bottle to treat both sides of approximately 20 discs (as opposed to the 25 Sound Guard advertises). He also noted that Discwasher and Sound Guard exhibit no interaction.

Now the important news ! Dr. Maier is readying production of a product he sees as a replacement for both Sound Guard and Disc Protect. It will be a case, 12 ½ x 12½ x 1 inch, with a handle and four apertures, and an atomizer bottle inside for storage. The procedure is to first thoroughly clean the disc to be treated, then open the case and remove the bottle stored within, insert the disc, and reclose the case. The bottle nozzle is applied to each of the four apertures in turn, spraying a finely atomized (Maier compared the Sound Guard "basketball" droplets to his vapor's "ping-pong balls"), metered amount of chemical on both sides of the disc in an even coating. The enclosed system is designed to reduce waste; the metering of the fluid ensures reproduction of technique from use-to-use; the smaller particles in the vapor due to better atomization facilitates a more uniform, thinner coating on the disc. Dr. Maier has visually tested the uniformity of coating by tagging the chemical with a dye and viewing the disc after spraying. He will check for uniformity of coating under electron microscopy and thereby determine whether burnishing is necessary. He also noted that this potential need will be further explored by listening tests—can we apply a microdeposit evenly enough on a surface so that we do not hear the irregularities due to the imperfections in the smoothness of the applied chemical?

Dr. Maier plans to market his product for under $20. It will treat both sides of over 100 discs.

— David Weinberg (Maryland)

More on Sound Guard's Safety

The glowing reviews of Ball Brothers' Sound Guard in Radio-Electronics (March 1976, pp. 41-43, 98-99) and Audio (April 1976, p. 62) prompt me to write to warn members of the BAS that all may not be quite as rosy as these reports would make out. Specifically, what started me thinking about this matter was the statement that Sound Guard coats the vinyl surface to a thickness of 5 millionths of an inch, as if this is a negligible factor. This is, however, not the case, as I shall show.

Let us suppose, fairly conservatively, that at 1 kHz the peak velocity that can be recorded on a disc is 35 cm/sec (corresponding to a sine wave amplitude of 5.6×10^{-3} cm). Further, assuming the signal-to-noise ratio of the disc to be 60 dB at 1 kHz (a frequently achieved figure), we can calculate the amplitude of the still-audible 1-kHz signal when cut at a level 60 dB below the above peak level, i.e., at 3.5×10^{-2} cm/sec peak velocity. (To relate peak velocity vp and amplitude A at frequency f on a sine wave, we use the formula vp = 27rfA, where f is in Hz, A in cm, and vp in cm/sec.) We find that a 1-kHz signal is still audible on the disc at a recorded amplitude of 5.6×10^{-6} cm, i.e., 2.2 millionths of an inch. Now this is already less than half the thickness of the Sound Guard coating. But let us continue.

Consider now the situation at 10 kHz. Because of the RIAA pre-emphasis of 13.7 dB at this frequency, a signal recorded 60 dB below our assumed level at 1 kHz will at 10 kHz be down only 46.3 dB, corresponding to a peak velocity of 0.17 cm/sec, or an amplitude of only 1.1 millionths of an inch. Note that this signal is still audible (ignoring Fletcher-Munson effects), even though these amplitudes are much less than the wavelength of light.

My point is that a coating of 5 millionths of an inch is not ignorable as regards unimpaired record reproduction. One would expect, on the basis of these calculations, that fine mid- and high-frequency recorded detail might indeed be seriously impaired by the Sound Guard coating, whereas loud signals would escape sonic degradation because of their large amplitude. In this connection one should note that neither of the cited reviews measures the effects of Sound Guard on anything but relatively large-amplitude signals. (The 30-kHz carrier signal used on CD-4 discs is, for these purposes, a large amplitude signal. Being recorded at a peak velocity of 3.5 cm/sec, this signal has an amplitude of 7.3 millionths of an inch. This may explain why Len Feldman did not measure any change in 30-kHz signal output upon treating the record; compared with the peak-to-peak signal amplitude of 14.6 millionths of an inch, the 5 or so millionths of an inch coating of Sound Guard would not have had a marked effect.) I can certainly see a reduction of surface noise—the fine dirt would be imbedded in the Sound Guard coating.

To substantiate my claims, Toronto BAS member Chris Gupta, a friend, and myself conducted a brief experiment comparing treated and untreated discs, and found that we could indeed hear an impairment of the fine inner detail (not a blatant change, but a subtle one, which can only be for the worse) on the treated records. So I would warn members to conduct experiments of their own to satisfy themselves of the value of the product before following the advice implicit in Len Feldman's last sentence: "I still have about 300 more records that need to be 'sprayed'. . ."

By the way, the records we used were all well buffed. It does occur to me that since the effect of Sound Guard appears to be subtle, and if my explanation is correct, then it will be most apparent with a stylus of very small tracing radius (e.g., a Shibata-type), and may be much less audible with an ordinary elliptical or spherical stylus. Our tests were performed using a Sonus Blue Label cartridge, which has a very small (0.0001 inch) tracing radius.

— Stanley P. Lipshitz (Ontario)

Dr. Lipshitz sent a copy *of his letter* to Ball Brothers for comment. The following are some extracts from their letter *of response,* written by Virgil Friebel, Ph.D.:

"The initial report on the thickness of the coating was worded 'less than 0.000005 inch thick' because that was the limit of our measuring capability at that time. . . We have now developed new optical techniques using birefringence patterns and we have now established that the coating is in the range of 1/2 millionths to 0.000001 inch thick, even after repeated coatings on the same record.

"In regard to your detection of a subtle change on the fine inner details after treating records with Sound Guard, I would be interested in hearing if you have done any more evaluation in this regard. One thing that we have /earned *has* been that people have a tendency to buff less than is necessary in many cases. We realize this *is* a normal tendency for people who take excellent care of their records, but good firm buffing *is* no problem because of the protection provided by the Sound Guard coating.

"I would be particularly interested if you or any of your fellow BAS members have had a chance to run comparative wear tests such as one record being repeatedly played with one side coated with Sound Guard and the other one in the normal uncoated condition. I believe you would find this test very interesting."

The Futterman H-3A Stereo Amplifier

The Futterman H-3A operates class A to approximately 30 watts/channel, then in class AB to its rated power (60 watts rms/channel into 8 ohms, 100 watts rms/channel into 16 ohms). The price is $350, and a 100/150-watt mono version is available at $500/pair.

Is the Futterman H-3A the "perfect" amplifier? No, it has one weakness that may be important to those with single-amplifier systems. The bass below about 80 Hz tends to be slightly soft compared to, say, the Ampzilla. If this is not an overwhelming shortcoming to you, then read on.

My loudspeakers are ARC Tympani MA's with Ampzilla-driven subwoofers. "Aha I ," I *can* hear the audiophile purists say, "The Maggies are agressive and unmusical sounding. Therefore are you (the author) a worthy judge of amplifier performance?" Well, I must admit I had always felt that same way about the Maggies, maybe because dealers always use ARC power amplifiers to drive them. Until I heard the Futterman-Maggie combination, I simply did not like the sense of agressiveness that I heard in the Maggies. To my ears the only real difference between the ARC-driven and Futterman-driven Maggies is that lack of agressiveness with the Futterman-Maggie combination. The depth and detailing seem about the same.

With all due respect to Mr. James Bongiorno, Ampzilla simply cannot compete with the Futterman in the mid-range and treble. Ampzilla just does not have the depth and lacks the sweet detailing of the Futterman.

The Futterman also sounds good on double KLH-9's but perhaps here two Futterman amps in parallel would really make more sense since power goes up with load and one cannot defeat (easily) the internal crossover of the KLH-9. However, the KLH-9's don't seem to produce quite as glaring a difference between the Futterman and other amplifiers as do the Maggies.

There is, of course, one drawback—perhaps a major one. You must periodically readjust the bias and balance potentiometers in both channels to compensate for aging of the tubes. And sooner or later those tubes will go kaput. The adjustments are easy, but I don't know how easy it will be to buy new tubes.

I am planning to order another Futterman; I want to eliminate the passive crossover in the Maggies, and two Futterman amplifiers cost slightly more than half as much as one ARC Dual 76A.

Forgive me Julius—your three-month, word-of-mouth backlog may mushroom beyond belief. For further information contact: Mr. Julius Futterman, Futterman Electronic Labs, 200 West 72nd Street, New York, N.Y.

— Tom Mashey (Connecticut)

Janszen Electrostatics

In the January 1976 issue I encountered Robert Graham's piece on the Janszen 130 and the Ionovac tweeters. What nostalgia! I bought my first Janszen 130/AR-1W in 1956. This beautifully mated combination was part of my first component system bought from Irving M. Fried, who was then in retail audio at Lectronics of City Line Center in Philadelphia. He personally delivered it to me and set it up in my New York apartment. The previous fall I had heard Arthur Janszen demonstrate the speaker combination at the New York audio fair. In those days of loaded horn and bass reflex enclosures (the Karlson enclosure was a big seller also) the electrostatic/acoustic-suspension combination sounded to my ears like the real thing. Old timers will recall that Julian Hirsch adopted the speaker system as his reference when he was publishing the Audio League Report. I bought my second Janszen/AR second-hand around 1962 when I went stereo. Inflation hasn't really been all that harsh during the intervening years. I paid $165 for the new Janszen 130 and $120 for the used one, compared to a present price of $200.

Now the odd part of all this is that the Janszen/AR's are still my principal speakers. I have a couple of KLH 7's as rear ambience speakers with Dynaquad supplementing the Janszen/AR's in the main listening room. But the latter have remained my mainstay for all these 20 years.

Why haven't I moved to something more "modern"? Graham's description of their sound tells the story better than I can. I concur thoroughly in his evaluation—except in one respect. I do not agree that the 130's lack lateral dispersion. Vertical dispersion, yes, but not lateral. Because of their nearly semicircular array of radiating elements, they propagate—to my ears—a wide sound front with no beaming. I have them mounted above the woofers in corners 14 feet apart and about 15 to 20 feet from the various listening positions in the room. They are turned so that the output of one of the four radiating elements bounces off the adjacent side wall. I can move around from one listening position to the other and the instruments of the orchestra stay in the same place. I can stand up or sit down and, far enough back, the 130's weak vertical dispersion is not apparent. It becomes evident only if I stand right on top of them.

As I say, these units have played on and on, the 130's plugged into the ac outlet (for bias supply) at all times, except *when* I moved to Connecticut. But finally, after 20 years, the older 130 got problems; it started making warbles, chirps, and occasionally a few zips. After some phone calls I found out, *as* Graham says, that Electronic Industries in Minneapolis now make and service the line; so I called them and sent the unit back for a checkup. It turned out that all it needed was a new capacitor.

I have not heard the newer 132, 134, 138, et al., nor the newer version of the 130. In talking to EI, I learned that they are really tweeters with restricted output below 1500 or 1200 Hz. The old 130 is both a tweeter and mid-range unit, with a low-pass filter at 500 Hz. Without any cross-over at all it mates beautifully with the AR-1W, whose rolled-off output is useful up to 1000 or 1200 Hz.

It seems to me the emphasis of the audio publications on the latest gear overlooks the bulk of the equipment that most audiophiles still use. Probably everybody has some "vintage" units. He hasn't upgraded them because he is not convinced the financial outlay is worth the difference in sound. And maybe it isn't. Many vintage units are competitive with the latest when tweaked in some way: Equalization, speaker placement, tonearm damping, "mod" treatments of electronic gear. What we do *not have* (outside The BAS Speaker) is comparative evaluations that include (1) the new, (2) the "standards" of former years, and (3) units with various types of modifications or enhancements.

— William S. Vincent (Connecticut)

At the Threshold

For those who are still looking for the perfect amplifier and who require (or prefer) a lot of power, I commend the Threshold 800A basic amplifier to your attention. At 200 watts per channel and $2,100, *it's neither small* nor cheap, but it *is* one of the smoothest, most detailed amps I have ever heard. A class A configuration, the Threshold uses active biasing to reduce wasted power to a minimum; at the end of a 45-minute listening session the unit was only slightly warm to the touch.

In terms of sound quality, the 800A combines the solid bass of good transistorized designs with an exceptionally smooth high end, fully the equal of good tubed designs. Added to that is an unusually nice front panel design with over-under channel meters, and it even has an LED to tell you when it's turned off. In total, the Threshold 800A may be very close to the state of the art; I suggest you listen to it and decide for yourself.

— Steve Seto (California)

Rectilinear Midrange

The new midrange driver mentioned by Carlos Bauza *as* now being used in the Rectilinear HI *is* the Philips AD5061/M8, which Rectilinear is selling for $11.75. This unit is a high-compliance version of the earlier AD5060/M8 and may be had for $5.95 plus postage and $0.30 handling fee from McGee Radio, 1901 McGee Street, Kansas City, Missouri 64108. The 5061/M8 is less efficient than the old 5060/M8 and sounds worse. Anyone who has an old Rectilinear HI with the stiff, low-compliance midrange should hang on to the original driver. Those who have the newer Rectilinear HI with the high-compliance midrange might consider replacing the midrange driver with the *Phillips* AD5060/SQ8, which is comparable with the old unit and is available from McGee for $11.95 plus handling and postage.

— Jim Nichol (Massachusetts)

Formula 4 Modification

The pins that restrict the fore and aft travel of the sliding cursor weight are press fit and can easily be extracted without removing the arm from the turntable. Grasp the pin with needle-nose pliers, and work it side to side while exerting gentle downward force. Slit the heat-shrink tubing that contacts the lifting shoe and remove it. Now slide the cursor weight all the way back toward the arm pivot. Rebalance the arm with a good stylus force gauge, leaving it tracking slightly light. Apply the final tracking force by sliding the cursor weight forward, but not more than 1/4 inch from the rear-most position so that it will not interfere with the lifting shoe. These alterations do not affect the operation of the arm, cost nothing, make balancing easier, and reduce effective mass.

— Dow Williams (California)

Seriously Folks, the Last Word on the 814

I recently joined the BAS and became quite enthused about the 814 microphone capsule, but then became very disappointed when I read that it was no longer available. In desperation, I tried to find a replacement and have discovered the following:

Thermo Electron sold patent rights and some production facilities to an English subsidiary of Knowles Electronics, called, I believe, Knowles Electronics, Ltd. They are expected to make only the 5336, 5356, and 5351 mike capsules, but I don't know if they will be distributed in the U.S. The 814 capsules were virtually hand built. They will not be built by KE, Ltd.

The capsule KE makes for Group 128 is or was for their contact mike. KE does not supply them with capsules for their other mikes. Mr. Ken Sikora of KE sent me data sheets on their mike capsules, but they really make capsules for hearing aids and voice transmitter mikes, so only one capsule has a response even close to audio requirements, and it appears to peak around 15 kHz and to be unsuitable for high-quality audio. So much for that source of mike capsules.

But fortunately I didn't give up. I was told by other sources that Group 128 may be out of the high-quality audio mike business, making the search for the supplier of their high-quality mike capsule fruitless. However, I then talked to the president of an audio company known for making high-quality, very musical sounding equipment, and he told me that he is perfecting a mike capsule for a very high-quality mike that should sell for around $300. (His company will make the mikes.) The mike is presently in the stage where they are perfecting the vowel sound of it! He projected a price of around $50/capsule and said he will notify me when they are available (expected by the end of the year). Although I do not feel I should reveal the name of the company right now, I can confidently state that the capsule should be equal to or better than Sony condensers such as the C37 and C500. I will, of course, be talking to Peter Mitchell about these capsules when they become available so he can get one or two for experimenting. If all works well, we may end up with a mike that costs about $20 to $40 more than the 814, but is substantially better. Keep your fingers crossed!

— Bob Sellman (New Jersey)

Tape Copying Service

Since I do some live recording myself, albeit infrequently, and have expanded my system to supply copies of my live recordings to the performers, I thought that some members might have use for such a tape copying service. Because this is only a hobby, I am willing to try making copies of recordings for BAS members for small amounts of money that hopefully will cover my expenses. If there is enough interest in copying, I may eventually add a dbx 124 to my system to provide more copying capability.

At present, I can make quarter-track reels (7.5 ips), cassettes, and 8-track cartridges of very good quality. (Even the cartridges sound half decent!) I Dolby-B process all cassettes and cartridges and can use Dolby-B or dbx 119 (normally at 1.4 compression factor) for reels. All copies are made at playing speed. Masters can be 7.5-ips or 3.75-ips reels, dbx 119 or Dolby-B processed if possible, and on 10½-inch reels if necessary. Unfortunately, I cannot at present handle half-track, 15 ips, or dbx 120 or 150 series encoded recordings.

I will duplicate recordings for BAS members under the following conditions:

1. I ask for the following amounts in advance to help defray the costs of tape and equipment: set up, $6/master; copies, $3/cassette and $4/reel or cartridge. These figures include my supplying low-noise tape and are for two-sided recordings of up to 44 minutes/side (88 minutes total). Deduct 50 cents/copy for recordings on one side of tape only (44 minutes maximum). I can get up to 55 minutes per side on reels using 1-mil tape and use 120-minute cassettes and cartridges if necessary, but please contact me for details. You pay all shipping and insurance costs.

2. Specify how many copies you want in each format and the type of noise reduction for reels (Dolby-B will be used if none is specified). Since I can make one reel, one cassette, and one cartridge simultaneously, I prefer to have a mixture of types.

3. Masters should be quarter track stereo, 7.5 ips (preferred) or 3.75 ips. Specify total length of each side of the recording in minutes, noise reduction used, and, if possible, the maximum level of the recording (decoded) referred to Dolby calibration level as 0 dB. This level will help me make better recordings. If you have only a cassette master, I will make an unedited submaster from it for duplicating for $6.

4. You must do all labeling of the tapes. I do ask, however, that you let me know the details of the recording so I at least know what I'm listening to and copying.

5. If you want, I will hold the master you send me for a while, in case you want to have me make more copies.

6. Submitting a tape for copying implies that you have made any and all necessary arrangements so that there are no legal restrictions on copying the tape for you and the people for whom you want the copies.

7. If you must have an edited submaster made from your recording (applause cut out where possible, pauses cut down, etc.), I will make one by re-recording (yes, it does work quite well), but I must ask you for at least $15 because of the amount of time and work required, and you must accept my editing decisions. I will, of course, follow any preferences you indicate as much as possible. Your recording will be returned unharmed.

I have found that members of performing groups enjoy having recordings of their efforts. Also, it is personally satisfying to have them tell you how good the recording you made is. With luck, the copies might even make it easier to make more live recordings, at least of the same group.

Interested parties should write me at 14 Station Avenue, Haddon Heights, New Jersey 08035.

— Bob Sellman (New Jersey)

RFI: The Risks of Federal Legislation

In a recent issue Harry Zwicker concluded that the solution to the RFI problem is legislation now pending before Congress. For reasons too numerous to fully detail here, I disagree.

For openers: there is no RFI problem as such. What exists is a number of separate difficulties sharing common symptoms. At any given time far more users of high-fidelity equipment are <u>not</u> having this sort of trouble than are suffering from it. RF troubles can arise only when equipment is used in excessive ambient fields. They do not come from any sort of equipment failure, but from statistically abnormal conditions of operation.

For the unfortunate with the problem, the situation is extremely complex. The offending signal can originate from a nearby AM broadcast tower; from an illegal 200-watt linear amplifier on a neighbor's CB transceiver; from a close-by business band AM base station with a mistuned transmitter; or from any other barely conceivable, improbable, and nearly random cause. (Virtually every manufacturer I contacted described illegal CB power as the most troublesome source. One company reported incidents in which illegal transmitted .power had fried customers' units.)

The offending RF field can be coupled into the equipment at the phono cartridge, by phono preamp leads, on the AC power line, by direct pickup, on high-level audio lines, on speaker lines, by parasitic modulation of desired RF signals, or by one's subscription to the catastrophe of the month club. After the unwanted RF energy gets into the machine, its mode of conversion to audible disturbance depends both on its level and the specific circuitry used by a particular manufacturer. Often a quantitative difference in interfering signal can cause a substantial qualitative difference in audible disturbance.

This means that each case of RFI is a unique phenomenon, and no single solution can work everywhere. Especially when the only element common to all cases is the operation of equipment in excessive RF fields.

To help alleviate the problems faced by RFI sufferers, I have contacted four manufacturers not mentioned in Zwicker's article. All were very interested in helping customers with such difficulties. Phase Linear reported they have formed no RFI policy because they have had no complaints. All others stated that their policy is to contact the customer directly, and then base a solution on the information obtained from his answers to their questions. This enables them to determine what particular trouble he is having. Here are people to contact if you're having trouble:

Mr. Larry Winter
Phase Linear Service Department
20109 48th Avenue West
Lynnwood, Washington 98036
(206) 774-8848

For Technics:
Matsushita Electric Corp. of America
Panasonic Service Division, Eastern Region
Attention: Quality Assurance
50 Meadowland Parkway
Secaucus, New Jersey 07094
(201) 348-7461

Mr. Bill Gehl
Sherwood
4300 North California Avenue
Chicago, Illinois 60618
(312) 478-7300

Mr. Al Hyle
McIntosh
2 Chambers Street
Binghamton, New York 13903
(607) 712-3512

This illustrates a rather interesting property of the free market. It has a built-in regulator that, when properly used, is sufficient for the solution of such problems: the profit motive. One of the most financially rewarding investments a manufacturer can ever make is a happy customer. There is no other form of advertising as convincing as word of mouth. In fact, in many consumer-oriented businesses, higher managers in charge of daily corporate operations tend to be very concerned with end-user satisfaction. This is especially true where high-fidelity equipment is the only source of revenue.

Be sure to contact the right person. Companies are made up of individuals, and you need to find one who cares about you and has the resources to solve your problem. Generally someone fairly well up the corporate ladder is best. For something like this don't waste time yelling at your local dealer; talk to the manufacturer's national service manager. Other choices aren't as good.

[The BAS will note and print the names of people to contact and which policies to follow with which manufacturers. This will inform equipment owners of solutions available for RFI problems and simultaneously inform prospective purchasers of what sort of service they are buying along with their equipment. — Ed.]

If we turn to government to solve the problems, we not only deprive the manufacturers of the incentive discussed above, but also drive equipment prices up by making everybody buy what only a few need or want. The law is the law for everybody; and government can only require the same standards everywhere—from Levinson to Electrophonic. There are more appropriate approaches available, but not when one must treat all the divergent problems, situations, and standards as being essentially the same.

In addition, design constraints for RFI immunity and cartridge impedance interaction immunity are at least partially opposed. As Tom Holman said in the revised paper which he submitted to the Journal of Audio Engineering Society, "Any design for RF filtering at the input of high-fidelity amplifiers will need to be subjected to close scrutiny for not causing any audible difference while still rejecting the RF. Also, the legislation may be so restrictive that it prevents making preamplifiers which do not interact with cartridge source impedances."

Things may not get quite that bad, but such a law might still require designers to trade off whatever parameters Congress deems appropriate for increased RFI immunity. And even if everything else I've said here were to be proved untrue tomorrow, I would never support legislation that constrained our ability to purchase equipment designed the way we think best—and I rather expect the same courtesy in return.

— Dick Bowser (Nebraska)

But If You Disagree:

The Honorable Barry Goldwater has introduced radio-frequency interference (RFI) legislation into the Senate. The bill, S. 3033, is virtually identical to the RFI bill introduced into the House last year by Vanik. Goldwater's bill has been referred to the Senate Commerce Committee. Interested readers should forward their comments on the bill to: The Honorable John O. Pastore, Chairman, Communications Subcommittee, Senate Commerce Committee, United States Senate, Washington, D.C., 20510. Correspondence should indicate that a similar measure is awaiting hearings in the House of Representatives (H.R. 7052).

— H. Zwicker (Massachusetts)

Record Reviews

Since the earlier review of a disc on the Oryx label (Feb. 1976, p. 11), I have acquired two more recordings from this company. One is "Organ and Harpsichord, and Forte Piano and Guitar" (Oryx EXP 58 stereo). Side one is a recording of music by S. Giussani, by C. P. E. Bach, and by T. Giodani for organ (a chamber organ of 1680 played by Franz Haselbock) and harpsichord (one built c. 1782 and played by John Henry van der Meer). The music is in a light vein, exhibiting the sonic contrast between the harpsichord and the calliope-like organ. The performance and music are very enjoyable, as is the recording. The surfaces are quiet and the recording clean with good detail, which gives the instruments a sense of "thereness." My one complaint is that the two instruments do not occupy a natural sound space; they are placed at the extreme left and right, causing a hole in the middle. Side two is music by A. Diabelli and by J. Kuffner for forte piano (one built c. 1815 and played by Rita Maria Fleres) and guitar (a Staufer guitar played by Mario Sicca). The music reminds me of 19th century parlor music, for which I do not have an affinity. The recording quality is essentially identical to that of side one except it is a bit softer in sound (perhaps reflecting the character of the instruments more than of the recording techniques).

The second Oryx record is "Renaissance Music of the Imperial Court" performed by the Pohlert Renaissance Instrumental Ensemble (Oryx EXP 56 stereo). This is music taken from the "Liederbuch" published c. 1520. It is similar to other Renaissance music with which I am familiar, though at times it drags a bit (not enough melodic or harmonic interest to sustain slow meter). Also, the performances occasionally seem lackluster and mechanical. The recording quality is very good, though variable in terms of instruments. Sonically, the disc is detailed with good airiness and depth, but high frequencies seem a bit muted at times and percussion instruments are poorly defined and distant. Otherwise, the musicians are well recorded, especially the recorders and the tenor voice. The record surfaces are reasonably quiet, with noise evident only during quiet solo performances.

The three Oryx discs I have heard so far indicate that this English company produces records of very high recording quality. I should add, however, that the discs I have dealt with are not representative of their catalog. Material ranges from the Middle Ages to the 19th century, with special emphasis on the Baroque period and J. S. Bach.

Another excellent recording of renaissance music is "The Pleasures of the Royal Courts" by the Early Music Consort of London, directed by David Munrow (Nonesuch H-71326). This is music from five European courts, ranging from the 13th century court of the Trouveres to the early 16th century court of Spain. The material is of greater interest to me, with more life, than that of the above Oryx collection, and the music is performed with greater feeling. This disc would certainly be my musical choice between the two. The recording quality is very good, but again uneven. The instruments occupy a good stereo space with somewhat distant perspective. Great detail is elicited, revealing more of the characters of the various instruments than the smoother Oryx effort. But despite its fine definition and spatial characteristics, the recording seems less open than the Oryx product. Again, there is an unevenness in the recorded quality of the instruments. The worst recorded is the countertenor vocalist, who sounds subdued and strangely colored (reminding me of a person singing falsetto with a head cold). The best recorded are the percussion instruments, which have a strong sense of "thereness," followed by the recorders. The remaining instruments fall somewhere between the extremes but closer to the recorders. The above criticisms perhaps leave a more negative impression than they should. All of the above discs are, on the whole, a clear cut above the average disc in their sonics.

An exquisite product is A. Scarlatti's "I Madrigali," performed by the Monteverdi-Chor Hamburg under Jurgen JUrgens (Archiv Produktion Stereo 2533 300). Here one is offered the eight madrigals composed by Scarlatti. Despite the madrigal-like lyrics, the basic sound is that of Baroque choral music and thus a bit of a surprise if one is expecting more traditional madrigal forms. The choir is made up of young voices (as opposed to the mature and overly rich voices of the Metropolitan Madrigal Singers, which by comparison seem inappropriate to this kind of music) but is superbly trained and directed. They offer a beautiful, moving rendition of the rich, polyphonous rondo form of these compositions. The recording is also excellent. The best way to describe it is to say that if you know what the best Fulton choral records sound like (detailed and three dimensional), then you know what this disc sounds like. Archiv Produktion should be congratulated on such an outstanding effort.

The next recording is quite different, Stravinsky's stage work "L'Histoire du Soldat" performed by the Boston Symphony Chamber Players (Deutsche Grammophon 2530 609). John Gielgud is a good narrator and Ron Moody a properly sinister Devil, but Tom Courtenay's Soldier seems too dry and deadpan. The musical performance seems excellent, and overall the production works very satisfactorily (this is the only version of the work with which I am familiar, so I have no basis for comparison). The disc is quiet, with a forward, detailed, dynamic recording of the chamber ensemble. The characters are recorded in a closer perspective with excellent definition. Overall, the work is a pleasure.

In a similar vein, there is the recording of Stravinsky's "Chamber Music," also by the Boston Symphony Chamber Players (DGG 2530 551). Here are spirited performances of some delightful compositions. There is the brassy "Octet for Wind Instruments" (1923/52), the gentler "Pastorale for Violin and Quartet of Wind Instruments" (1934), the humorous "Ragtime for Eleven Instruments" (1918), the constantly evolving "Septet" (1953), and the energetic "Concertino for Twelve Instruments" (1952). This is the first time I have heard any of these works, and I really do not feel comfortable commenting on the merits of the performances. I will just state that musically this is my favorite disc of those I have disucssed. The recording is excellent, with depth, airiness, dynamic range, good detailing, good imaging, etc. Judging by the above two discs and other recent DGG releases, perhaps DGG is finally starting to restore their former high level of quality.

More on the PCM Disc

I would like to add another comment to the continuing dialog on the PCM recording of Telemann. After reading Mr. Lipshitz' comments in the July issue of the Speaker (p. 3), I listened to the disc again to see if he was correct about the nature of the background noise. Having read his suggestions, I could indeed interpret the background as studio noises and certain rumblings (especially on side one) as traffic noises. But I must admit that unless someone had suggested it I would have done no more than consider the noise tonally odd but still tape noise. Several audiophile friends have listened to the record on different systems and have thought they just heard tape hiss, but another friend, a recording engineer, quickly agreed that the nature of the noise is as Mr. Lipshitz has described, though accompanied by hiss (perhaps just surface noise). So I am willing to admit, given the peculiar characteristics of the noise, that Mr. Lipshitz' interpretation of the nature of the background sounds is probably correct, though I still do not agree with his conclusions.

About the only conclusion that can be drawn is that the recording was made in a poorly designed or badly constructed studio. The background sounds mask any tape noise, so all one can say is that the inherent noise level is not above that of background sound sources, which is really relatively high compared to that of the quietest discs available (such as some of the recordings on the Oryx, Ensayo, Archiv, Nonesuch, EMI, and other labels, as well as on Ambiphon/Sonar tapes). Given the level of these background noises, it is no great feat for a tape deck to record them. I have a Crown CX 844 (a good deck, but certainly not one of the best) and make recordings with a pair of Advent microphones (fairly good microphones—especially for the money—but certainly of relatively low quality compared to professional models). Unfortunately, the tape deck is in the same room in which I record, implanting in the backgrounds of all my tapes the whine of its transport cooling fan. The fan is about as far below the level of the music source as the background sounds in the PCM recording and is more clearly recorded. So, in answer to the question in the last sentence of Mr. Lipshitz' comments, if I can do it with my average setup (which has no noise-reduction system), then I should assume any professional deck and the best consumer models would do the same thing (my recording engineer friend's personal experience confirms this conclusion). The PCM deck offers nothing special here.

In reviewing the recording again, I have reevaluated my earlier positive comments on the merits of the recording (April 1976 Speaker, pp. 5-6). Recent experiments with multiple miking techniques to capture ambience have made me sensitive to overdoing this sort of thing. I now find the strong ambient/reverberant quality of the recording unpleasant. My other friends, on hearing the disc for the first time, commented on how they disliked this aspect of the recording. The recording engineer even went so far as to volunteer that it gave him a headache to listen to it. I should conclude now that the Teleman disc is (relative to the best discs available) a poorly engineered recording made under difficult circumstances. It certainly is not the quiet, state-of-the-art recording described by Mr. Garfinkle (December 1975 Speaker, p. 5).

Little can be concluded about the PCM deck itself. We will have to wait for releases of PCM recordings made under better circumstances to really evaluate this newest advance in tape deck technology.

— Collins Beagle (Virginia)

In the Literature

Audio, Aug. 1976

- Doppler Distortion in Loudspeakers: Imported from Britain, land of phase shift, comes "definitive" proof of the existence of this ever-recurring type of distortion. Next month or so we shall read the rebuttals, but for now it seems that small-cone systems really are inferior. (p. 42; note numbers at the side of the pages)
- A nice technical review of the McKay-Dymek AM-5 AM tuner and DA-5 antenna are followed by a Heyserized view of the Duntech Labs DL-15. The antenna is, by the way, available as a kit. (p. 54)

Audio, Sept. 1976

- An interesting book review on a history of audio. (p. 30)
- Tutorials in this issue include "Understanding S/N Ratios" (p. 32), which applies to tape decks in particular, and "Reading VU Meters" (p. 42)
- Equipment reviews include the Phase Linear 2000 preamplifier; the inexpensive Audioanalyst A-100X loudspeaker, which is found overly bright and somehow "thin" in the bass; and the Nakamichi 600 cassette deck with its distortion suppressor, fine bass response, good treble, excellent speed stability (if a bit off correct speed), and interesting mechanical layout. (pp. 56-68)

Audio Amateur, 2/76

- A good editorial discusses frequency balance and miking, loudness levels, quad, and more. (p. 1)
- A new build-it tonearm for the brave is a work of art, although the suggested length may raise some eyebrows. (p. 5)
- Review of the Heathkit power amplifier (AA-1640, 200 watts/channel) discusses in depth its electrical design and building the kit. (p. 10).
- Other items include "Audio Mixers, Part 2," more modifications for the PAS-3x, "Active Filters, Part 3," and an unusually interesting letters section, including missives from BAS members. (p. 26, p. 39)

Audio Engineering Society, Journal of the, June 1976

- **SQ** Dichophony—Quadraphonic Earphone Listening: Benjamin Bauer describes localization of SQ signals heard over binaural headphones. Conclusion: "The human hearing apparatus acts as a reasonably good quadraphonic decoder for earphone-applied SQ signals." (p. 387)

db, July 1976

- The 54th AES convention and exhibit are discussed. (p. 33)
- The Signal Path: H Sine Wave Oscillators: Various op-amp circuits are described by Walter Jung. (p. 38)
- Budget Sound, or What About the Garage?: Description and pictures of a garage studio. (p. 44)

db, Aug. 1976

- Theory and Practice: Loudspeaker impedances and crossover networks. (p. 10)
- Broadcast Sound: Discussion of noise—types of noise, how to find the sources, and how to eliminate them. (p. 16)
- Biamplification—Why and How: Don Davis, well-known author and lecturer, deals with some of the misconceptions regarding biamping. (p. 26)

Electronic Design, Aug. 2, 1976

- From the Consumer Electronics Show, Jim McDermott selected the Sequerra tuner, the ADC Accutrac turntable, and the Audio Pulse time delay unit as show stoppers. (p. 26)

Electronic Design, Aug. 16, 1976

- Measure Peak Signals Accurately: Describes the circuit for a fast, accurate, but still rather simple peak and hold circuit. (p. 90)

Electronics, June 24, 1976

- The Winning Ways of Video Games: Discusses the latest in TV games. (p. 89)
- MOS Moves Into Higher-Power Applications: Siliconix engineers describe new V-MOS devices. Includes circuit diagram for 80-watt push-pull stereo amplifier, claims 0.04% THD with only 22 dB of negative feedback. (p. 98)
- One-Op-Amp Oscillator Keeps Sine-Wave Amplitude Constant: Simple circuit uses 741 op amp and six passive components. (p. 107)

Electronics, July 8, 1976

- Coming: Programmable Video Games: More on TV games. (p. 67)

Electronics, July 22, 1976

- Pre-Emphasizer Speeds FM Tuner Measurements: Circuit compensates for tuner de-emphasis, (p. 122)

High Fidelity, Aug. 1976

- Edgar Villchur is featured in the Pathfinders series, part nine. (p. 34)
- The new Harman-Kardon HK-2000 cassette receives its first review in a major hi-fi magazine: fine performance for a medium price. (Note the new noise-weighting and THD measurement methods described on p. 44.) The Nakamichi 600 review shows a high end that runs off the page, IM is much lower than in the HK, owing to "IM suppressor" circuits. Poor speed accuracy is in agreement with a "Shop Talk" comment on Nakamichi's QC.
- Tests of Twenty-One C-90 Cassettes: Another good comparative review of the various modern tape emulsions, curiously omitting dropout data. (p. 48)
- Note in the Ampex ad, p. 71: "at standard factory bias setting"—even for the chrome-like emulsions?

High Fidelity, Sept. 1976

- A letter (p. 6) claims that both RCA and Angel have improved their disc quality, while DGG has gone downhill. Do members agree?
- Another Boston-area audio designer, Victor Brociner of Avid Corp., is spotlighted in the Pathfinders series. (p. 44)
- The ESS Heil-Driver headphones look like fine tweeters, while you must decide for yourself on the importance of an ultrasonic resonance in the Sonus Blue Label (Equipment Reviews, p. 50).
- The "new equipment for 1977" piece on p. 55 seems unusually informative this time.

IEEE Trans. Consumer Electronics, May 1976

- Signal Processing in the SQ Logic System: Discussion, performance curves, and a circuit for a full-logic SQ system. (p. 149)

Popular Electronics, Aug. 1976

- Stereo Scene: Discussion of tube versus transistor sound, mostly clipping and TIM. (p. 14)
- Buyer's Guide to Antenna Rotators. (p. 39)
- Test reports on the Onkyo TX-4500 receiver and the B&O Beogram 1900 turntable. (p. 66)

Popular Electronics, Sept. 1976

- Stereo Scene discussion of hi-fi in Japan is interesting, but not really informative. (p. 14)
- Special HiFt Section: Useless. A ten-band/side stereo equalizer is offered in "kit" form for $130, but the remainder of the section (discussions of Class D amps, phono preamplifier design, 4-channel tape, and choosing loudspeakers) is devoid of interest or insight. (p. 53)

Radio-Electronics, Aug. 1976

- Not at all related to hi-fi, but part one of a very useful two-part article called "Digital Clock Roundup" begins in this issue, discussing all of the clock kits on the market for quality, simplicity, features, and everything else one can never glean from the advertisements. (p. 33)
- Class G High Efficiency Hi-Fi Amplifiers: Discussion of the Hitachi circuit, which goes "Class D switching" one better. (p. 47)
- Equipment reviews include the Hitachi 3500 (3-head, $400) cassette deck and the B&O receiver, which is found to barely outplay a good transistor radio. (p. 53)

Radio-Electronics, Sept. 1976

- Clocks, Part II. (p. 45)
- A discussion of the AR 16-channel time-delay system is stocked with error and not worth reading. (p. 57)
- Tested: JVC S-300 receiver and Empire Z-2000 cartridge. (p. 66)

Recording Engineer-Producer, June 1976

- Understanding Magnetic Tape Performance Specifications: As is true of every tape article ever written, this does not tell all, but it is worth the trip. Note, however, that pages 34 and 38 are interchanged. (p. 30).
- Interview with the Columbia engineers responsible for the Grammy winning "Daphnis" and the Cleveland/Thomas "Carmina" are also worth reading. (50 mikes for Carmina?) (p. 43)

Stereo Review, Aug. 1976

- Roy Allison presents one of his best and most useful articles to date, in a career which has been loaded with absolute winners for clarity and content. There is absolutely no flavor of sales for Allison loudspeakers in this piece (unless the new bookshelf will have some very strange property). Even if you never buy SR, get a copy of this article. (p. 56)

Stereo Review, Sept. 1976

- The Q&A section by Larry Kline has good ones this month, particularly if you are not yet familiar with magnetic fluids (see August meeting summary below). (p. 22)
- Also in a usually ignored monthly column, Craig Stark discusses compromises in record/playback heads. (p. 30)
- Has anyone really bought the Sony TC-880-2 tape deck ($2500 list) reviewed on p. 35? The Tannoy Cheviot is reviewed on p. 46, and we wonder again if anyone finds these lines particularly outstanding?
- Preview of new equipment (p. 65) seems uninformative (as if it were written from advertising copy). Compare with the one in High Fidelity.

July BAS Meeting

Business Meeting

This month's meeting opened on a note of concern. Jim Brinton again pointed out that the BAS is in danger of collapse. Many of the people who have been the Society's backbone will not be available for the next membership year. We need more workers to keep up with the work generated by a large and rapidly expanding membership. The July Speaker contains a list of open positions and job descriptions. Look for something you can do, something you might even find yourself enjoying.

Peter Mitchell announced that WBUR is now using its new antenna and transmitter. He commented that the station's signal is now vertically as well as horizontally polarized, for better penetration, and that the phone lines between the studio and the transmitter are unusually quiet. WBUR also has a new stereo generator, with less noise and IM distortion, and, in the studios, Davis-Brinton phono preamps.

Another member announced a new product—the ADS 910 speaker system. Designed for DGG, it features two ten-inch woofers, a dome midrange, and a dome tweeter, relatively high efficiency, and provisions for bi- or tri-amplification. A consumer version will be available for about $500 per speaker. ADS will also make an electronic crossover module for the speaker.

Meeting Feature

Audio delay lines were the evening's main topic. Joel Cohen, president of Sound Concepts, Peter Mitchell, now a consultant for Audio Pulse, and Richard DeFreitas, designer of the Audio Pulse Model One, were the principal speakers. Mitchell opened with a discussion of the relevant psychoacoustics. According to Mitchell, the main difference between ordinary stereo reproduction and real music is that stereo produces no ambient field. Even when the ambience of a live performance is captured in a recording, the reproducing system distorts it by putting it all in front of the listener. A delay line can restore lost ambience and put it in the proper perspective.

Perception of ambience depends upon the precedence (or, after its discoverer, Haas) effect, which is the aural equivalent of persistence of vision. The brain has a 0.050-second short-term memory. It compares all fresh input with the contents of this memory. Provided new sounds arrive within 50 milliseconds and are not much different in character or much louder, they are interpreted as reflections and suppressed from consciousness. It is the brain's subconscious analysis of these sounds that tells us the characteristics of the surrounding space.

Mitchell, who admitted there might be some controversy over his choices and that his comments were biased somewhat toward the Audio Pulse view, identified six essential factors defining the character of the reverberant field. The first is the length of the early time delays. The interval between the direct sound and the first reflections provides an index (about one foot per millisecond) of the performer's distance from the nearest reflective surfaces, e.g., stage walls. Although close miking techniques tend to eliminate these complex reflective patterns from most commercial recordings, Mitchell feels that it's generally bad practice to tamper with the signal to the front speakers, as this results in distortion of information already in the recording. For extra-dry recordings, the Audio Pulse delay line does have provisions for mixing some delayed signals into the front channels. Additionally, it has taps on the rear panel that can be used for front delay speakers.

The second important factor is the lengths of the late time delays, which arrive from the sides and the rear of the hall. In large halls, these delays can be 20 to 100 milliseconds or more. (The Audio Pulse unit uses six discrete delays of up to 94 milliseconds, available in two mixes of four delays according to the position of a front panel switch. These delayed signals are filtered and recirculated with progressive attentuation in a number of selectable combinations to provide reverberant decay. The Sound Concepts unit uses a single continuously variable delay of up to 50 milliseconds in stereo or 100 milliseconds in mono, also filtered and recirculated. The amount of reverberation is set with another control.) Mitchell said that the delay channel speakers should go to the sides, because studies have shown lateral reflections to be the most important, and that rear and ceiling speakers are desirable, though not essential. Audio Pulse recommends that the delay speakers be placed to the sides near the ceiling. Their unit has additional taps for running longer delays through a set of rear speakers.

Another important concern is the spacing of the delays. They must be close together so as not to be heard as separate echoes and unequally spaced to prevent flutter echoes, which produce a twangy coloration. Audio Pulse deals with the problem by using a mixture of different delays. Sound Concepts relies upon listening-room reflections, which are adequate fillers within the SD-50's delay range.

A hall's absorption characteristics are also significant, especially in the first 20 milliseconds. The most natural results are achieved when the decay time of the reverberations introduced by the delay line match those of the hall in which the recording was made. Then the electronics effectively expand the recorded ambience. Consequently, the best recordings benefit most from time-delay enhancement.

The fifth important factor in Mitchell's list is the frequency content of the reverberant field. In a good hall, the high frequencies roll off rapidly and the bass rises below 100 Hz; the midrange between 200 and 2000 Hz remains largely unaffected. Typically, the treble rolloff begins in the **I-** to 2-kHz range, leaving virtually no energy above 8 kHz. Both the Audio Pulse and the Sound Concepts units attenuate the treble in the rear channels, the Sound Concepts somewhat more than the Audio Pulse. The rolloff increases with delay in the Sound Concepts unit, which cuts off above 10 kHz. The Audio Pulse uses a fixed filter and cuts off above 8 kHz. Because delay systems help smooth out room resonances (standing waves) through cancellation, they automatically provide some bass enhancement. In addition, the Audio Pulse has a switchable bass boost circuit tailored to typical hall characteristics. Mitchell emphasized that the midrange characteristics of the secondary speakers should be similar to those of the primary speakers, lest the illusion created by the Haas effect be impaired.

The final important characteristic is the ratio of direct to reflected sound. Correct setting of the rear channel level is critical. A nominal level about 6 dB below the front speakers is a good starting point.

Mitchell also pointed out that the reverberant field is highly noncoherent, without appreciable stereo separation. This is not to say that it's monophonic, as it would then be localized at rear center. It should be nonlocalizable. The Audio Pulse uses phase shifters and recirculation to achieve this effect; the Sound Concepts depends on recirculation between channels.

In reply to questions, he said that delay lines probably wouldn't be useful for suppressing room echoes in recording. On the other hand, he did say that delay systems tend to suppress the characteristics of the listening rooms in which they are used. The room remains important, however. If longer delays are required, Mitchell said, delay boxes can be connected in series.

Joel Cohen, the designer of the SD-50 and president of Sound Concepts, spoke next. He said, and Mitchell agreed, that in A-B comparisons the sonic differences between the two devices are quite small. In contrast to the Audio Pulse unit, which employs analog-to-digital conversion and digital shift registers, the Sound Concepts device clocks the signal through bucket-brigade IC's, which are simply analog shift registers. Cohen feels that apart from basic design, the main difference between the two units is one of emphasis. The Audio Pulse lavishes attention on late and multiple delays; the Sound Concepts concentrates on the initial delay, which Cohen thinks is the most important. He also said that he thinks 50 to 60 milliseconds of delay is about the most that's usable without ping-pong effects.

According to Cohen, most recordings have some ambience information in them, and the function of a delay device is to "peel it off the front wall." He uses a continuously variable delay to facilitate "tuning in" on the recording and on the room, to smooth out resonances. He did say, however, that even a simple 15-millisecond delay without recirculation or additional signal treatment helps a great deal, and both he and Mitchell expect a flood of such devices from established manufacturers. Both Sound Concepts and Audio Pulse, they said, are aiming for the state-of-the-art market.

The Sound Concepts unit features a continuously variable reverberation control, because records vary so much in the amount of reverb they already contain. With some, especially SQ discs, the user may want to use no reverberation at all.

Cohen said that short delays caused by reflections in the listening room fill in the gaps between electronically produced delays, so he doesn't worry much about using only one delay at a time. He also said that varying the delay helps compensate for variations in listening position. It's best, he feels, to sit closer to the rear speakers than to the front.

Despite the lower complexity of its basic design, the SD-50 costs about the same as the Audio Pulse ($600). The reason is that it uses a compander circuit to improve its signal-to-noise ratio. This is necessary to keep the rear speakers from being obtrusive during silences.

Cohen indicated that he will sell bucket-brigade IC's for the Popular Electronics kit or any other at cost (about $9) to BAS members.

Dick DeFreitas, the designer of the Audio Pulse Model One, spoke next. He briefly recounted the development history of the Model One, then turned to an explanation of its design. The heart of the unit is a new kind of analog-to-digital converter, called a delta modulator with memory. Pulse code modulation, the conventional digitizing technique, is too inefficient and expensive for use in a consumer delay line. It's inefficient because it uses coded pulses to represent the instantaneous change of the signal voltage. Consequently, delta modulation requires only about half as many bits of information as PCM for the same degree of accuracy. This, in turn, reduces by half the number of shift registers required. To prevent slew-rate-limiting, the circuit keeps track of recent variations and predicts on the basis of this memory how the signal slope will change between the last sample and the next. This improves the circuit's ability to follow fast, high-frequency waveforms. After the pulses have been delayed by passing through the shift registers, they are reassembled into a choppy replica of the original waveform, which is then smoothed into a near perfect copy by a low-pass filter.

Six discrete delays are generated by tapping off the shift register chain at various points. These delays are mixed to generate others of various amplitudes, which arrive at the rate of about one every 5 milliseconds, after recirculation, filtering, and phase shifting. The prototype had continuously variable delay, but it was found that about 20 switch-selectable delays provided essentially the same flexibility.

Unlike the Sound Concepts, the Audio Pulse uses no noise-reduction circuitry. The listener uses a level match control to achieve full dynamic range with low noise.

DeFreitas indicated that Audio Pulse would provide information to consumers for specific modifications but not schematics.

He and Cohen agreed that only reducing performance will lower the cost of delay lines. They also agreed that the ultimate subjective effect of a delay system depends upon a number of factors, including the design and adjustment of the delay line, the speakers used, their attitudes and placements, the listening room acoustics, etc. The best rooms, they said, are those of average size with good acoustics; the worst are large rooms.

After the presentation, members were given an opportunity to hear a comparative demonstration of the two units in Audio Pulse's sound room. — Michael Riggs (Massachusetts)

August BAS Meeting

The August meeting of the BAS was held at the Norwood facility of Teledyne Acoustic Research, housing both corporate research and development as well as speaker manufacturing. Prior to the featured tour of the plant, BAS members and guests were graciously treated to a buffet dinner.

After a brief business meeting, Frank Sax, AR Executive Vice President, explained AR's relationship to Teledyne, a technically based conglomerate whose directors are not afraid to support product development efforts requiring technological advances. Robert Berkovitz, Director of Research, summarized the research projects shown on the tour (time delay, impulse testing, automotive acoustics, and psychoacoustics) and introduced Tim Holl, Chief Acoustical Engineer.

Tim described AR's production facility and techniques for fabricating and testing speakers. He explained that individual drivers should be designed to behave well not only through the frequency range in which they will be used but also outside of this band. A speaker with a resonance at 2 kHz might be used with a 12 dB/octave crossover at 1 kHz to minimize excitation of the resonance by signal components at 2 kHz, but the non-linearities of the speaker could still allow the resonance to be excited by the second harmonic of a 1-kHz signal. Speaker misbehavior, such as cone breakup or edge reflection, is controlled by treatment of the cone paper with proprietary energy-absorbing compounds, and by the use of high damping cone roll supports. Similar treatment is applied to soft-dome tweeters, whose softness is important because it helps suppress breakup.

Development of a new speaker with a special plastic cone material having a high damping coefficient is now being pursued. One advantage of such a speaker is that the plastic's properties are much more consistent than those of the usual treated paper material, reducing quality control problems and perhaps manufacturing costs.

A recent development is the use of magnetic fluids (synthetic fluids that are attracted to a magnet) between the voice coil and the magnet. They are held permanently in the voice coil gap by the magnetic field. These special fluids were originally used at AR to suspend and center the voice coil in the gap of a new midrange driver whose design did not leave room for the normal mechanical spider suspension. Though this worked quite well, two additional advantages of using magnetic fluid were soon discovered: it helps damp cone overshoot and resonances and dramatically increases the speaker power handling capacity by providing an excellent thermal conduction path for drawing heat away from the voice coil to the cooler magnet structure.

Anechoic and Reverberant Speaker Testing

After the proper designs for the individual drivers are achieved, they must be integrated into a speaker system and measured. The measurements should be designed to correlate closely with the listening experience. Anechoic chamber measurements pose problems with microphone placement, especially in the vicinity of crossover, where radiation from two drivers can interfere and produce notches in the radiation pattern. Diffraction from the cabinet also produces confusing interference effects. These problems are minimized by averaging measurements from many positions around the speaker to obtain the frequency response characteristic of the total radiated energy. This is the function of the reverberant chamber.

When testing new speaker designs or performing quality control checks on their production models, AR uses both anechoic and reverberant test chambers for frequency response measurements. The anechoic chamber gives information about on-axis response from about 160 Hz up. Below 160 Hz, where the chamber is no longer anechoic, measurements are made out of doors with the speaker and microphone suspended about 25 feet above the ground on a specially prepared crane arm.

The reverberant chamber is a small, solidly built room in which none of the walls are parallel or perpendicular. It therefore has no characteristic resonant frequencies and tends to integrate the sound from all axes of the speaker to produce a measure of the total energy output at each frequency. Uniform energy output with respect to frequency is deemed a desirable trait for a well-balanced speaker system.

Other smaller anechoic chambers are located in the speaker production facilities for quality control inspection of individual drivers. Though these chambers are not large enough to be truly anechoic, they have been calibrated with drivers of known characteristics, whose responses in the chamber are used for comparison with production units. Smoothness of frequency response, cutoff frequency, and distortion are the normal test criteria. AR's harmonic distortion reject level on a typical midrange unit is 1%, but most speakers pass at the 112% to 3/4% level. Rejected units are scrapped; only the metalwork and magnets are salvaged. The speaker is rebuilt, and the metalwork is given one more chance. After a second failure, the metalwork is also scrapped. Some drivers are so difficult to build that the reject rate may exceed 25%. In one corner of the production floor stand the fruits of AR's stringent quality control efforts—over fifty barrels of rejected speakers.

Automotive Acoustics

The recent proliferation of audio systems for cars, which offer performance well beyond the 8-track-and-rear-deck-speaker hookup, indicates a market potential for really high quality auto audio. AR has been investigating the factors that affect sound reproduction in an automobile: speaker location, frequency response and how it is affected by open windows, noise level, and spectral energy distribution are some of the items that have been examined.

To determine a generalized frequency response for auto interiors, various commercial 6- x 9-inch car speakers were mounted in the rear deck apertures of a number of late model cars and their third-octave pink noise frequency responses were measured. At low frequencies the listener sits in the near field of the speakers, with the effect that variations in the volume of the interior have little effect on the response. This was demonstrated by having members watch the response on a real-time analyzer while opening and closing the doors. Opening and closing the trunk also has a negligible effect. The high-frequency response generally agrees with the speaker's anechoic high-frequency characteristics, indicating that most of the high-frequency energy is bounced off the rear window toward the passengers.

When the separate in-car responses of all of the speakers were averaged and the result plotted, the bumps and dips characteristic of each individual unit were smoothed out, except for a low-frequency peak at 250 Hz and a dip at 650 Hz. These were found to be a consequence of the rear deck location and characteristic of all cars. Their presence may justify designing compensating equalization into any future AR automotive speakers or sound systems.

Impulse Response

Would you like to find out all about your speaker's anechoic frequency response, polar-energy response, phase response, and energy-time response, all in your own living room without special chambers? All you need is a pulse generator, a microphone, and a minicomputer. AR has these and is busy gathering data on a number of their speakers in a small laboratory with no special acoustic treatment. Impulse testing, popularized by Richard Heyser in his speaker reviews for Audio, is a powerful technique for characterizing speakers through their response to an impulse of electrical energy. The impulse is a very short (10 µs) burst of energy delivered to the speaker from the amplifier, which should sound like the sharp crack of an electrical spark. All of the frequencies of the audible spectrum are contained in the impulse and sent to the speaker in that instant. The acoustic reproduction of the impulse by the speaker is picked up by the microphone, recorded by the computer, and analyzed to extract the speaker's amplitude and phase response for each of the frequency components. From this the computer can, for instance, derive the time of arrival of energy in various bands of the frequency spectrum and determine the phase relationships and diffraction properties of the individual drivers.

The power of impulse testing lies not only in its speed and convenience but also in the ability of the computer to manipulate the data for presentation in a variety of formats and to expedite the measurements. As an example of the latter, it is straightforward to subtract in the computer those imperfections in the measured response contributed by the amplifier and microphone. It is also possible to select for analysis only the first few milliseconds of sound arriving after the pulse is applied to the speaker, cutting off before room reflections arrive, to give the anechoic response. Or, by including later sounds, which will have first reflected from a room surface, speaker-room interactions can be analyzed in detail.

AR is working toward ways of applying this technique to speaker design as well as analysis. It should allow much faster turnaround on evaluation of new designs and perhaps provide insight into some relevant characteristics of speakers not discernible from steady-state testing. This should be the result of AR's plans to simulate with the computer the sound of various speaker systems and to investigate the psychoacoustic effects of modifications in speakers' phase or energy-time responses.

Psychoacoustics

With hi-fi advertisements boasting equipment distortion levels of 0.05% and less, many people would be surprised to learn that their threshold for perception of distortion in a complex waveform is 2% to 3%. This was demonstrated at AR with a standard psychoacoustic testing technique. The subject, listening to a signal, controls the level of distortion with a switch, causing it to rise or fall gradually at will. He is told to decrease the distortion until it is no longer audible to him and then to increase it until it can again be heard. This is repeated several times as a chart recorder, out of the subject's view, traces the rising and falling distortion level. The resulting graph, having a sawtooth shape, allows the tester to draw a line representing the subject's distortion threshold midway between the peaks and valleys. One of the audience, eager to demonstrate his aural acuity, scored right along with the majority, detecting 0.2% sine-wave distortion but falling back to a 2.5% distortion threshold on a sawtooth waveform, where the rich spectrum of harmonics tends to mask harmonic distortions.

AR is interested in the psychoacoustic aspects of sound reproduction as it affects speaker design and also from the standpoint of manufacturing tolerances and quality control. It is helpful, for example, to be able to set meaningful specifications for acceptable distortion levels based on the limits of human perception. Psychoacoustic analysis and testing procedures will also pay a part in evaluation of the computer-simulated phase and energy-response experiments and of the time-delay system.

Time Delay

AR's 16-channel time-delay system was assembled to determine what the minimum requirements are for acceptable, realistic synthesis of ambient information by time delay. The system consists of two front channel AR-11's driven by an AR amplifier and eight pairs of AR-7's driven by 10-watt modular amplifiers and distributed about the room. A digital signal-processing unit has been constructed that allows delays from zero to 256 ms to be separately programmed for each of the 16 channels. The system has been set up to simulate roughly the acoustics of Symphony Hall. Delays for each channel are based upon the calculated arrival times of wall reflections meant to be reproduced by that channel. High-frequency attenuation was introduced in each channel, *as* a function of that channel's delay, to account for air absorption.

Plans for this system include further psychoacoustic studies of time-delay ambience *synthesis* and attempts to boil the findings down to a "popularly priced" product.

Some audiophiles may have lost confidence in AR's motivation to advance the state of the art since it became a part of the Teledyne conglomerate. But, if the work presented on this tour is any evidence, those audiophiles should look to Norwood and keep the faith.

—John Schlafer (Massachusetts)

Report on the BAS Tuner Clinics

Peter W. Mitchell

Introduction

In many areas of the country an FM tuner is a trivial adjunct to an audiophile's system; its quality is unimportant because of the dreadful engineering practices at most FM stations. Severe compression of dynamic range, Volumax limiting (which rolls off the highs in loud passages), and high distortion are pandemic in FM broadcasting. But in Boston the quality of one's FM tuner is, for many audiophiles, more important than the quality of one's phono cartridge. The reason for this can be summarized in four letters: WGBH. Arguably the finest FM station in the country, WGBH broadcasts three programs which demand a state-of-the-art tuner.

1) Live Boston Symphony broadcasts from Symphony Hall with only mild Volumax limiting occur nearly every Saturday night. Better yet, on Friday afternoons and occasionally on Tuesday nights the BSO broadcasts are "Victorized." (The signal from the microphone console is Dolby-A encoded, sent to the transmitter via 15-kHz phone lines, and decoded at the transmitter, bypassing all limiters, compressors, and studio consoles. This technique was developed by Victor Campos and the WGBH staff for "Adventures in Sound.") The resulting sound has a degree of clarity, low distortion at high levels, and impact on transients that simply are not available on disc recordings.

2) Victor Campos' "Adventures in Sound" provides master tapes from Columbia, RCA, Nonesuch, DGG, etc., which in most cases are so much superior to the resulting discs as to be an embarrassment to the disc-pressing industry. Most of the tapes are Dolby-A encoded and are left in that form until the signal reaches the Dolby decoder at the transmitter. No Volumax limiting is used. With first-rate master tapes, such as the Bernstein/MET "Carmen" produced by Tom Mowrey for DGG, the resulting sound is indescribably wonderful. As for those tapes created by such insensitive producers as John McClure of Columbia, their flaws are mercilessly exposed.

3) Nat Johnson's "King of Instruments" program on Sunday afternoons is sometimes Victorized as well, providing a test for any woofer and ecstasy for any pipe-organ buff.

In addition to these programs, WGBH airs tapes of other local live concerts, and WHRB and WBUR also sporadically provide live-concert broadcasts with lifelike sound.

So the quality of one's tuner is a matter of critical concern. And unlike our amplifiers, for which we can trust McIntosh and Marantz to provide occasional test clinics, our tuners are not subject to competent free testing. Manufacturer's tuner clinics are rare, and when they do occur, they often concentrate on the less important parameters. Moreover, if an amplifier is well designed, it will automatically work well as soon as it is assembled, so we don't really need tests to reassure us that our amps are OK. But a tuner's performance depends equally on its design and on the correctness of various "alignment" adjustments within it; for this reason, impressive specifications and favorable magazine test reports provide no assurance that your tuner is any good.

Background

There has been a major revolution in FM tuner performance during the past 10 years—not a change in the theoretical capability of tuners, but an enormous change in the performance level that *is* routinely achieved and maintained. There are two reasons for this, and one is simply the elimination of most of the alignment adjustments. A dozen years ago a good stereo tuner had 20 alignment adjustments, all of which had to be set correctly at the factory or service shop. On a production line, alignment tends to be quick and sloppy, and few tuners actually met their advertised specs (and with a tube circuit the tuner would soon drift out of alignment anyway). So first-class tuner performance was very rare, which is why a tuner such as the Marantz 10D (whose performance is equalled or surpassed by some modestly priced tuners today) was considered a classic. Thanks to the development of crystal and ceramic IF filters, monolithic IF IC's, and phase-locked loop (PLL) multiplex decoders, a good tuner can be inexpensive to manufacture and can have as few as five critical alignment adjustments. The simplification of manufacturing enables good tuner performance to be achieved routinely rather than rarely.

The second reason for the progress in tuner performance has to do with test equipment. Both in circuit design and in production-line alignment of tuners, an engineer is limited by the quality of his test gear. The introduction of the Sound Technology Model 1000 FM stereo signal generator in the late 1960's underlies much of the improvement in tuners in recent years. The $1800 ST 1000A with its optional M3 low-distortion stereo modulator provides a better FM stereo signal than any broadcast station and is much freer of noise, distortion, and spurious signals than earlier FM signal generators. In addition, the ST 1000A is equipped with a unique "dual sweep" feature, which radically reduces the time and skill required to achieve optimum alignment of a tuner's circuits. *En* fact, the ST 1000A is such an outstanding servicing tool that its use should be considered mandatory for any service shop claiming to do high-quality repair and alignment of FM tuners and receivers. (Among local service facilities, two that do use the ST 1000A are Stereo Lab and Suffolk Audio.)

The possibility of a BAS tuner clinic arose when we learned that Advent's R&D department has an ST 1000A, used in the development of the Model 400 radio and the Model 300 receiver. Tom Holman, Advent's chief electronics engineer, granted permission for us to use the generator, and the BAS test clinics inevitably followed. Other test equipment used for the clinics included Alvin Foster's Heathkit IG-18 audio signal generator (with The Audio Amateur's first set of low-distortion modifications), a Hewlett-Packard 333A harmonic-distortion analyzer, a Heathkit oscilloscope, and an active A-weighting noise filter developed by Peter Mitchell. In addition to the standard A-weighting of noise frequencies to correspond to their relative audibility at low sound levels, the filter also attenuates signals above 15 kHz so that ultrasonic garbage will not dominate the signal-to-noise (S/N) ratio measurement.

We made no attempt to do a systematic survey of the performance of all the tuners on the market. We simply tested the tuners that members brought in. The inclusion of a large number of Acoustic Research, Pioneer, and Dyna tuners reflects the tendency of BAS members to buy products which have a reputation for high performance at moderate cost. In two of the three brands mentioned, our tests justified that reputation.

Each tuner test required 30 to 40 minutes of time, so the 70 tuner tests were spread out over nine Sunday afternoons. The members who performed all this work were Peter Mitchell, Joyce Brinton, Jim Brinton, and Alvin Foster. The final batch of tests was delayed because the signal generator broke; completion of the tests was made possible by the generosity of Dick Pierce at Suffolk Audio.

The Test Data

The essential test results are summarized in the accompanying table. The tests are described below.

IHF Sensitivity (Mono). This is not a very useful number, but we included it to see whether tuners actually meet their advertised specs. Ideally this test should be performed in a shielded environment to produce accurate results. In the various environments in which we did our testing, it is possible that interference affected some of our data. Two of the AR tuners were particularly susceptible to static at RF levels below 5 microvolts (µV), preventing a usable IHF sensitivity measurement. But since some of the tuners did meet their IHF specs in our ordinary test environments, we believe the numbers are generally valid.

We also tried to measure the IHF sensitivity in stereo. But most tuners automatically switched into mono at a level between 5 and 10 µV.

Sensitivity for Full S/N Ratio. This is one of our most important tests. The essential superiority of FM over AM as a broadcasting medium is FM's suppression of static and reception noise, resulting in the ability to reproduce music with its full dynamic range. But FM's noise suppression is not a simple on/off phenomenon: it varies drastically with signal level. A lifetime of advertised IHF sensitivity specs has conditioned many of us to assume that normal and proper FM signal strengths are only a few microvolts in level. But at that level even the best tuner exhibits a S/N ratio of only 50 dB in mono and 30 dB in stereo. To compound the folly, nearly every tuner manufacturer supplies with his product a dipole antenna and implies that a dipole is all that you need—that a roof antenna is a last resort, intended only for the fanatic and the fringe-area listener. This is a very deceptive practice. The bald fact, as revealed in our tests, is that in order to fully suppress reception hiss and produce its advertised S/N ratio, a good FM tuner requires a signal strength of at least 200 µV in mono and 1000 µV in stereo. A good FM tuner is capable of a dynamic range equal to that of a 15-ips Dolby-A master tape, but only when fed an adequate signal. If you are using an antenna that delivers only 100 µv from your favorite station, you are throwing away 10 to 20 dB of S/N in stereo.

For most of the tuners this test was performed by ear: an unmodulated signal was fed to the tuner and the signal strength was gradually reduced from 10,000 µV until an audible increase in the residual noise level was heard in the monitor speaker. In the tests conducted at Suffolk no monitor speaker connection was available, so the signal strength was reduced until the measured noise worsened by 2 dB. Evidently this is a less stringent test, as it produced somewhat lower sensitivity values than did the tests done by ear.

We calibrated the signal strength meters on some tuners and found them lacking. In most cases they register full scale for all signals greater than a few hundred microvolts, indicating that only signals which "pin" the meter will be fully quieted by the tuner. Such meters are useless for orienting antennas.

Signal-to-Noise Ratio. Most test labs measure S/N ratio by connecting the tuner output directly to a meter, with the result that the dominant noise that is measured is hum and ultrasonic subcarrier leakage. In order to produce S/N data which correlate with what the ear hears, the noise must first be filtered to include only the audio bandpass, and then A-weighted. This is how our S/N ratio measurements were made. This and all subsequent measurements were made at a signal strength of 10,000 µV, midway in the range of signals commonly captured by a good roof antenna. A first-rate tuner will exhibit a weighted S/N ratio of 70 dB or better in stereo, and some tuners exceeded 80 dB in mono.

Incidentally, when reading magazine test reports, one must be careful not to confuse "quieting" with S/N ratio. Despite its misleading name, "quieting" in an IHF test is primarily a measure of distortion, usually at a test frequency of 400 Hz. So when High Fidelity reports that a tuner has a maximum quieting of only 46 dB, this just means that the THD at 400 Hz is 0.5%. Regardless of the test-report writer's confusion, it implies nothing about how quiet the tuner is.

In the few BAS tuner tests performed at Suffolk, an A-weighting filter was used which lacked the ultrasonic rejection of our standard noise filter; consequently in some of those measurements the tuner's 19-kHz and 38-kHz pilot tones contaminated the S/N ratio data.

THD at 400 Hz. The true harmonic distortion of an FM tuner should not vary substantially with frequency, so we chose 400 Hz to describe the basic THD of the tuner. Unfortunately the THD of a tuner does vary substantially as the tuner's frequency is fine-tuned within a reception channel. One of the best-kept secrets of the tuner business is that in actual use tuners seldom achieve the low distortion that their makers advertise and that magazine tests confirm. The culprit *is* the standard IHF test for tuners, which specifies that the tester should ignore the tuner's own meters and watch a distortion analyzer while fine-tuning the tuner to find the tuning point which gives lowest distortion. Then measurements are taken. If, instead, test-report writers were instructed to simply tune the tuner according to its own meters (as the ordinary user must do!) substantially higher distortion figures would be published for many tuners. The user of an FM tuner, of course, has no way of knowing how to tune for minimum distortion; the center-tuning meter is his only guide.

In our tests we followed the IHF procedure and tuned for minimum THD, but we also showed each member where the optimum tuning point was on the meter of his particular tuner. Unfortunately in some tuners the tuning position for minimum distortion varies with signal strength, a hopeless situation. The "Notes" column in the table identifies those tuners whose minimum-distortion point coincided with the center-tuning (CT) indication on the tuner's meter. On the average we found that when center-tuned a typical tuner exhibited about twice its minimum THD. Fortunately some manufacturers are paying attention to the discrepancy between a tuner's nominal distortion and its actual distortion when tuned by the user. Dyna's Dynatune, Yamaha's tuning-knob-actuated AFC, Radio Shack's Auto-Magic, and Onkyo's quartz-lock are desirable approaches to ensuring that a tuner will be tuned to its minimum-distortion point. But they are desirable only if they are correctly aligned at the factory. Half of the Dynatuners we tested were not; in these the Dynatune circuit pulled the tuner away from its minimum-distortion point.

Of course in most tuners the distortion is higher in stereo operation than in mono. Moreover the stereo THD depends on where the signal is on the stereo "stage." To obtain a representative range we measured the stereo THD twice: once with left-channel-only modulation and once with L R modulation (as would be obtained with a center-stage soloist). This is one area where recent advances in tuner design (both in the multiplex decoder and in the phase-linearity of the IF strip) have paid off. Many older tuners produced THD figures which were admirably low in mono but an order of magnitude higher in stereo.

12-kHz THD (IM). For years the test-report writers at High Fidelity and Audio have been measuring "stereo THD" at 10 kHz, finding typical values of 17%, and telling us that this result really should be ignored because the harmonics of 10 kHz are at ultrasonic frequencies and so are inaudible. This is simply not true. Every normal FM tuner contains a sharp 15-kHz low-pass filter; consequently the harmonic distortion products of a 10-kHz tone (at 20 kHz, 30 kHz, etc.) are blocked by the filter, cannot escape from the tuner, and so cannot be measured. In other words it is electrically impossible for a normal FM tuner to produce any measurable harmonic distortion from any test tone higher than 7.5 kHz. The published stereo "THD" at 10 kHz therefore does not actually represent harmonic distortion, but is a measure of something *else.* That, as we on "Shop-talk" have been saying for several years and as the magazines have lately begun to realize, is IM (intermodulation) distortion, or "beat-note" distortion. This produces a distortion product whose frequency is the difference between the test tone and the tuner's 19-kHz stereo pilot tone. Thus a 10-kHz test tone will produce 9-kHz distortion, and when the music contains an overtone at 12 kHz, the tuner generates an IM product at 7 kHz.

In most tuners this IM distortion is plainly audible, for the following reasons: (1) The IM product usually occurs at a frequency that clashes harmonically with the music. (2) After the distortion occurs in the multiplex circuit, its relative level tends to be boosted by the tuner's de-emphasis curve (e.g., the de-emphasis reduces the 12-kHz level by 15.2 dB and the 7-kHz level by only 10.7 dB, thus raising the distortion percentage by 4.5 dB). (3) The sensitivity of the human ear tends to be better, and most loudspeakers have better power response, between 5 kHz and 10 kHz than between 10 kHz and 15 kHz. As a result, in a typical case, a distortion product at 7 kHz is actually subjectively louder than the 12-kHz fundamental fed to the tuner.

It might be supposed that this distortion would be a minor problem, because in most music the relative energy level of the high harmonics above 10 kHz is relatively low. But FM broadcasting employs a strong high-frequency pre-emphasis (+17 dB at 15 kHz), with the result that high modulation levels in broadcasting usually involve a substantial percentage of high-frequency energy. So we have found, when doing A-B comparisons of tuners on high-quality broadcasts, that the severity of the tuner's high-frequency stereo IM is one of the characteristics that separates the excellent tuners from the ordinary ones. With a good broadcast, violins, bells, and cymbals will sound fuzzy and edgy in an ordinary tuner, clear and sweet in an excellent tuner.

As with the 400-Hz THD measurements, we made three measurements of 12-kHz IM: mono, stereo left-channel only, and stereo center-channel (L + R). The mono number is of course meaningless; the value obtained reflects only the residual unweighted noise of the tuner. In the stereo data, older tuners commonly produced IM values of up to 20% or 30%. The newer designs employing phase-linear IF stages and the Motorola 1310P phase-locked loop or other advanced multiplex decoders achieve distortion values of 5% or less.

Frequency Response. It is not difficult to design an FM tuner to have ruler-flat frequency response, and there are only three plausible causes of frequency-response error:

1) A poor choice of coupling capacitors in the audio circuit may cause a bass rolloff. Some manufacturers deliberately roll off the deep bass in the mistaken belief that FM stations are not permitted to broadcast signals below 50 Hz. Good tuners are flat to 30 Hz.

2) A stereo tuner must contain a filter to remove the 19-kHz and 38-kHz pilot signals, in order to prevent *tweeter* burnout and interference in tape recordings. Ideally the response would be flat to 15 kHz and then drop to –60 dB or so at 19 kHz. Such a sharp filter circuit is expensive, so the usual procedure is to use a filter which starts dropping about 13 kHz and is down 3 dB or so at 15 kHz. The audible effect of this is slight, because it covers less than one third of an octave.

3) Because broadcasting involves a 75-microsecond (μS) pre-emphasis, the tuner must contain an identical 75 μS de-emphasis to restore flat response. The de-emphasis is a simple RC circuit. If cheap, broad-tolerance resistors and capacitors are used, then the de-emphasis is not exact. The result is a "shelf" in the tuner's frequency response, with the transition occurring at 2100 Hz. That is, incorrect de-emphasis will cause all frequencies above 2100 Hz to be elevated or depressed by the amount of the de-emphasis error. Because the de-emphasis error occurs equally over nearly four full octaves of the audio spectrum, even a slight error (e.g., 0.5 dB) is plainly audible as a brightening or dulling of the sound, much more audible than a simple roll-off of the extreme top end. Incidentally, our measurements of frequency response are accurate to about ±0.2 dB. (In several cases, frequency response measurements are missing from the table because at the end of the first clinic, we found our measurements were invalid due to an impedance mismatch in the test setup.)

In the table, measurements at 20 Hz and 40 Hz illustrate any deep-bass rolloff. At the high end, not a single tuner was really flat to 15 kHz, though the Sequerra and Yamaha supertuners came close. The measurement at 10 kHz indicates the de-emphasis error in each tuner; when there was any doubt as to whether the error at 10 kHz was really due to de-emphasis, we re-checked it at 4000 Hz. The difference between the measurements at 10 kHz and 13 kHz indicates the effect of the pilot filter. For example, the AR tuners exhibited an average response error of 0.75 dB at 10 kHz and 0.91 dB at 13 kHz. The de-emphasis error is a bit larger than we would like, but the small difference between the two values indicates that the pilot filter is well-designed, having little effect below 15 kHz. On the other hand, the Pioneer 9100 showed an average de-emphasis error of only 0.36 dB (excellent), but was typically down 1.3 dB at 13 kHz due to the early rolloff of the pilot filter.

Stereo Separation. The table includes separation data for the low end, midrange, and high end of the audio range. A separation of 30 dB or better over most of the range is first-class.

Comments on the Results

Two conclusions appear to emerge from these measurements. (1) Tuner performance is not strongly correlated with price. Some modestly priced tuners, such as the Pioneer 7500, provide a level of performance quite close to that of the expensive "supertuners." (2) *Tuner* performance is strongly correlated with age. Tuners which were widely respected a decade ago, such as the Dyna FM-3 and Scott LT-11213, are seen to be mediocre by today's standards. So if first-rate FM is important to you, upgrading to a tuner of modern design would produce a substantial improvement.

The Acoustic Research tuner was designed about eight years ago and was *the* first of the modern high-performance, low-distortion stereo tuners; since then the rest of the tuner industry has been gradually catching up to the AR's level of performance. Although a couple of defective units were encountered, our tests generally confirmed the AR's reputation as a classic. AR tuners in good condition were found to be sensitivie, quiet, with low distortion, good separation, accurate center-tune meters, and a de-emphasis curve that typically was very slightly down in the treble. So the occasional second-hand AR offered for $125 or less looks like a very good buy.

The Dyna FM-3, though better than many tuners of its generation, is simply outclassed today. The FM-3's reputation for a sweet, non-offensive sound appears to have been the result of a severe treble rolloff. A third FM-3 was checked and exhibited the same 5-dB loss at 10 kHz. This contrasts oddly with the favorable test reports published on the FM-3 a decade ago in the magazines.

The Dyna FM-5 (and its AF-6 twin with AM) was the major disappointment of these tests. In addition to the previously known problem of hum in stereo (due in part to the *wiring* for the stereo indicator lamp) and the bass rolloff in FM-5's having a series 23 number, we also found a shockingly high incidence of gross misalignment in the detector, multiplex, and Dynatune circuits. And unlike the FM-3, the FM-5 is not realignable by the user; even in the kit, all of the circuitry is factory assembled and aligned. We were able to improve the performance of some of the FM-5's with the aid of our test equipment, but basically the FM-5 does not appear to be capable of state-of-the-art performance. Obviously Dyna needs to pay much closer attention to alignment and testing procedures. Beyond that, perhaps it's time for Dyna to admit that the FM-5 is a lemon and replace it with a freshly designed FM-7 having negligible hum, ruler-flat response, better stereo performance (using the 1310P phase-locked loop IC instead of the unnecessarily complex stereo decoder in the FM-5), and a Dynatune circuit whose alignment can easily be checked and corrected by the user. Meanwhile, as one contemplates the differences between our measurements and those published in the various favorable magazine test reports, one can't help wondering whether the tuners Dyna sent to the magazines *were* carefully doctored *samples rather than typical units* selected at random from inventory.

Perhaps the most amazing thing about the popular Pioneer 9100 was its consistency. With only rare exceptions, the Pioneers were uniformly very sensitive, extremely quiet, very low in distortion, with accurate de-emphasis and excellent separation. When one considers how many thousands of these tuners were shipped from Japan, the uniformly good alignment of the tuners in our random sample is impressive. Our only complaint about the 9100 is that its tuning point for minimum distortion varies with signal strength, usually being center-tuned with weak signals and somewhat off-center on the tuning meter with the strong signals commonly provided by a roof antenna. Curiously, the newer Pioneer 9500 and 7500 tuners failed to meet their IHF sensitivity specs, but in the important areas of distortion, separation, and flatness of response they were superb. Given the apparent consistency of the Pioneer tuners, the excellent performance of the one 7500 *we measured*, and personal experience with at least two other 7500's, and in view of its widespread availability at a discount price under $200, the Pioneer 7500 is probably the "best buy" on today's tuner market.

The expensive supertuners offered various points of excellence. For example, the Accuphase T-100 was distinguished by extremely low distortion, accurate center-tuning, extremely wide and uniform separation, and a genuinely useful multipath meter. The Yamaha CT-7000 was very quiet, very easy to center-tune, and especially low in distortion in its wide-IF mode. The Sequerra Model One was exceptionally sensitive, it was the only tuner whose de-emphasis curve was absolutely flat, and its spectrum-analyzer display was a joy to use.

Finally, it should be noted that there are important areas of tuner performance which we did not test. For example, as high-power FM transmitters continue to proliferate in urban and suburban areas, the tuner's ability to handle very strong signals becomes increasingly significant. When fed 200,000 µV or more, many high-sensitivity tuners overload and cross-modulate. Because the Sound Technology 1000A FM signal generator puts out a maximum of about 50,000 µV, we could not test tuners for overload. But if you are using a high-gain roof antenna (to minimize multipath interference) within five miles of a major FM transmitter, overload is a real risk. For example, in Charlestown the signal strengths from the stations I want to hear (WGBH, WBUR, WHRB, and WCRB) are all within the optimum range from 1000 µV to 100,000 µV; but in order to get WCRB, my roof antenna has to look past WBCN only three miles away, which pumps about 500,000 µV into my tuner. In order to get a usable signal from WCRB, it is necessary to use a signal attenuator (Radio Shack no. 15-1114) to cut down the signals; without it, all stations within 2 MHz on either side of WBCN are affected.

Another untested area is the tuner's ability to cope with multipath interference, ignition noise, etc. Our measurements correlate well with the tuner's sound as long as the tuner is fed a clean, multipath-free signal from a good roof antenna in a good reception area. But if the tuner is in a bad reception area or is fed signals from a poor antenna, then its ability to produce a clean audio output depends on aspects of front-end and IF performance for which no standard test has been developed. This is an area of tuner performance that is correlated with price; the more costly tuners do tend to be able to cope more successfully with bad signals than do medium-priced tuners.

Model and Serial Number	Mono IHF Sensitivity (μV)	Sensitivity for Full S/N (μV) Mono	Sensitivity for Full S/N (μV) Stereo	S/N, A-Weighted (dB) Mono	S/N, A-Weighted (dB) Stereo	Distortion (%) 400-Hz THD Mono	400-Hz THD L	400-Hz THD L+R	12-kHz THD (IM) Mono	12-kHz THD (IM) L	12-kHz THD (IM) L+R	Frequency Response (dB) Bass 20	Bass 40	Treble 10K	Treble 13K	Stereo Separation (dB) 40	400	10K	Notes
Accuphase T-100																			
B1X361	3.0	350	1800	82	74	0.08	0.50	0.07	0.10	0.72	1.5	−1.0	0	−0.7	−1.2	47	46	45	CT (center tuning coincides with minimum distortion).
Acoustic Research																			
TO 2498	1.4	200	800	78	73	0.09	0.30	0.23	0.10	0.9	2.7	0	0	−1.0	−1.1	38	38	31	CT.
TO 3423	2.1	200	900	75	71	0.18	0.60	0.36	0.14	2.5	6.	0	0	0	−0.2	40	40	28	CT.
TO 2356	1.9	120	800	80	74	0.20	1.6	0.33	0.15	3.4	6.	0	0	+1.0	−1.1	29	29	30	
TO 1345	2.0	300	800	75	72	0.14	0.5	0.2	0.11	2.4	4.0	0	0	−0.6	−0.9	40	42	33	CT.
TO 1168	25	1600	10K	77	72	0.43	0.9	0.7	1.2	3.	6.	0	0	0	−0.9	27	28	25	CT. Defective front end.
TO 2623	1.4	—	—	74	71	0.2	0.4	—	—	—	—	0	0	−1.0	−1.1	30	30	25	
Aligned	*	300	1000	79	73	0.25	0.33	0.44	0.10	2.0	4.8	0	0	−0.8	−1.0	47	48	30	*Static.
TO 0802	*	250	800	74	70	0.23	0.6	0.2	0.26	2.2	5.	0	0	−0.8	−1.0	35	35	33	CT. *Static.
TO 0734	9	400	1200	77	73	0.16	0.5	0.22	0.16	2.3	6.2	0	0	−0.7	−1.0	31	31	27	
VO 0183	2.0	250	1300	74	72	0.14	0.6	0.22	0.8	3.6	—	—	—	—	—	41	46	30	
LL 27925 (receiver)	2.5	200	1000	76	70	0.38	2.4	3.4	1.1	28.	43.	0	0	+0.6	0	−33	−34	−17	CT. MPX decoder misaligned, channels reversed.
Dyna FM-3																			
Unknown	10	400	—	74	51	0.9	1.9	1.8	0.6	19.	19.	0	0	−4.3	−6.	25	29	10	Mono THD 3% when center-tuned.
16601914	5	1000	2500	73	51	1.3	2.8	1.6	1.5	34.	40.	−0.2	0	−5	−6.5	22	26	9	
Dyna FM-5																			
Unknown	2.0	200	1600	76	66	0.32	0.9	0.36	0.12	6.2	7.6	—	—	—	—		Negative		Hum in stereo (S/N 52 dB unweighted). MPX misaligned, channels reversed.
Aligned	2.0	200	1600	76	66	0.35	0.8	0.4	0.12	2.4	6.6	+1.2	+0.2	+0.6	0	27	41	32	Aligned MPX.
23 151846	3.0	300	500	76	62	0.4	0.8	1.1	—	4.5	11.	−4	−1.6	0	0	16	22	18	Dynatune misaligned.
22 505073	4.0	200	1400	75	68	0.4	1.7	0.4	0.4	3.0	4.	+0.2	+0.2	+1.7	+1.3	7	7	9	Hum in stereo (48 dB).
Aligned	—	—	—	—	—	0.4	0.6	0.46	—	2.6	—	0	0	+0.5	−1.0	39	43	26	Aligned MPX.
23 215115	3.0	200	1000	72	63	0.14	4.0	0.33	0.6	2.0	4.1	−5	−2.2	−0.3	−0.3	16	19	19	Detector, Dynatune; and MPX misaligned.
Aligned	—	—	—	—	—	—	0.9	0.3	—	2.4	9.	—	—	—	—	18	35	21	Aligned MPX. Dynatune impossible to align.
23 207105	1.7	300	1400	75	71	0.5	1.9	0.7	0.3	5.3	8.	−4	−2	0	0	14	21	19	Detector misaligned. Distortion increases for signals above 10,000 μV.
22 432070	7	300	1300	79	69	0.58	0.76	0.4	0.1	3.0	5.	+1.4	+0.3	+0.7	0	27	40	30	Hum in stereo (55 dB).
Dyna AF-6																			
26 524010	2.0	250	1000	81	71	0.30	5.0	0.5	0.2	4.0	7.	0	0	0	−1.2	24	24	20	Hum in stereo (56 dB).
26 335151	2.0	300	1000	81	70	0.22	0.42	0.24	0.10	3.4	8.5	+2.2	+0.5	+1.2	+0.4	26	31	28	Hum in stereo (57 dB).
26 512108	3.2	130	900*	70	61*	0.33	4.7	0.8	0.24	4.0	10.	−0.6	0	+2.0	+1.4	33	29	28	*Suffolk test. Hum in stereo (50 dB). Detector and Dynatune misaligned.
With PLL	—	—	—	—	—	0.29	0.28	0.27	0.30	4.5	8.5	—	—	—	—	24	22	18	Stereo separation and 12-kHz THD data reflect pilot leakage.
Heath AR-14																			
716 9224	40	300	2000	67	46	1.6	1.7	2.3	2.7	16.	14.	0	0	−2	−4	17	19	11	Misaligned.
Heath AJ-15																			
836 2045	6	500	3000	76	72	0.16	0.40	0.25	0.7	4.7	17.	−0.6	0	−0.3	−1.8	34	44	24	
951 2184	14	1000	3000	74	72	0.29	1.1	0.28	0.2	4.4	17.	−0.4	0	−0.8	−1.3	24	27	25	
Repaired	3.2	200	600	82	60*	0.21	0.70	0.17	0.2	5.0	24.	−1.0	−0.2	−3	−3.2	34	44	22	*Suffolk test.

Model and Serial Number	Mono IHF Sensitivity (μV)	Sensitivity for Full S/N (μV)		S/N, A-Weighted (dB)		Distortion (%)						Frequency Response (dB)				Stereo Separation (dB)			Notes
						400-Hz THD			12-kHz THD (IM)			Bass		Treble					
		Mono	Stereo	Mono	Stereo	Mono	L	L+R	Mono	L	L+R	20	40	10K	13K	40	400	10K	
H-K Citation 14																			
921 0346	1.7	160	1000	86	73	0.13	0.30	0.09	0.14	3.0	7.	−0.2	0	−0.8	−1.7	37	37	34	Suffolk test.
Kenwood KT-7000																			
060 429	2.1	300	1000	82	66	0.80	1.7	0.7	1.1	19.	8.	−1.4	−0.3	−2.8	−2.4	19	19	17	
950 215	2.3	200	1000	79	65	0.28	1.6	0.34	1.2	20.	29.	−1.2	−0.4	−1.6	−2.4	29	29	22	Pilot leakage. IF misaligned.
With PLL	1.9	—	—	—	—	—	—	—	1.2	40.	50.	—	—	+0.7	+1.1	31	35	19	Suffolk test. Pilot leakage. IF misaligned.
Lux T-310U																			
F5101148	1.9	200	1200	81	74	0.13	0.58	0.09	0.12	1.1	2.7	−0.5	0	−1.9	−3.0	45	41	29	CT.
McIntosh MR-78																			
AD 1326	5.0	450	2200	80	53	0.56	2.0	2.0	0.8	7.5	—	−1.0	−0.2	−0.7	−1.3	21	28	16	In MR-78 and MX-110, stereo S/N, THD, and separation data reflect pilot leakage (no 15-kHz filter).
McIntosh MX-110																			
62221	2.2	500	1600	78	56	0.14	2.7	2.6	0.7	32.	30.	−5	−2	0	−0.3	21	39	16	
McIntosh MR-71																			
88 B 25	2.5	1200	4000	87	58	0.22	1.0	0.75	0.50	23.	28.	0	0	−1.3	−2.8	35	37	20	
73 B 19	3.0	—	—	78	64	0.62	1.8	0.70	1.6	16.	19.	−0.6	0	−0.5	−1.0	11	12	8	MPX misaligned.
Marantz 24																			
5900143	5.5	350	2000	82	71	0.33	2.0	0.33	1.3	26.	30.	−1.3	−0.3	+1.2	+1.7	22	24	23	
Marantz 150																			
1533	2.2	300	1000	79	71	0.18	0.26	0.15	0.15	1.9	—	—	—	—	—	26	43	32	
2200	2.3	200	1400	79	72	0.42	0.42	0.2	0.10	1.4	4.2	−2	−0.5	+0.5	+0.4	28	44	26	CT.
Marantz 10B																			
10-4065	7.0	—	3000	80	75	0.28	0.26	—	0.50	20.	—	—	—	—	—	10	11	8	
Pioneer TX-9100																			
UE 370 8117H	1.6	300	1000	85	74	0.28	0.30	0.23	0.10	2.5	7.0	−0.2	0	−0.3	−1.3	37	40	34	
UG 370 9334M	1.4	220	1000	83	73	0.19	0.45	0.17	0.24	2.7	9.	0	0	−0.3	−1.1	37	42	36	
UG 370 9072M	1.8	300	700	86	74	0.26	0.30	0.24	0.22	1.7	5.0	0	0	−0.5	−1.7	30	31	30	
TH 360 1636	2.0	250	1500	84	75	0.05	0.38	0.08	0.10	1.6	5.0	0	0	−0.4	−1.3	28	29	33	
UC 370 5321M	1.6	200	800	87	75	0.19	0.34	0.16	0.20	1.5	4.5	0	0	−0.3	−1.3	42	49	36	CT.
TL 370 3577	3.5	500	3000	84	74	0.13	0.42	0.11	0.15	1.5	4.6	0	0	−0.6	−1.6	34	35	37	Bad front end.
UE 370 8471M	1.4	190	600*	86	72	0.14	0.30	0.29	0.13	1.3	6.4	0	0	0	−1.0	41	47	40	*Suffolk test.
TD 360 1040	1.4	200	800	85	71	0.14	0.86	0.19	0.13	2.0	5.3	−1.0	−0.4	0	−1.0	37	41	33	
UA 370 4456M	1.4	180	900	84	69	0.13	0.18	0.13	0.15	2.2	8.	0	0	−0.4	−1.3	46	49	35	
TL 370 3846	1.3	200	800	85	65*	0.19	0.23	0.19	0.13	1.3	5.	−1.4	−0.5	−0.3	−1.1	34	37	35	*Pilot leakage.
Pioneer TX-9500																			
VG 390 2071M	2.0	300	900	84	74	0.19	0.13	0.25	0.10	0.86	3.	0	0	−0.3	−0.8	33	45	41	
VA 390 1084N	3.0	220	1400	82	74	0.10	0.46	0.13	0.15	1.3	3.3	0	0	−0.3	−1.0	33	50	38	
VG 390 2063M	6*	300	2000	84	74	0.28	0.34	0.25	0.12	1.1	3.4	0	0	0	−1.0	32	38	36	*Static.
Pioneer TX-7500																			
VF 390 1296M	2.6	200	1000	79	72	0.10	0.13	0.11	0.11	1.5	5.3	−0.8	0	−0.4	−0.9	37	45	37	CT.
Pioneer 535 Receiver																			
UE 390 5282M	10	300	2000	78	52	0.35	1.2	1.6	0.3	14.	18.	−1.0	−0.2	0	−1.1	38	38	24	
Pioneer 626 Receiver																			
SD 381 0609	6	300	2200	76	60	0.15	3.6	0.45	0.32	5.7	7.8	−2.4	−0.8	−0.8	−2.7	14	20	21	

Model and Serial Number	Mono IHF Sensitivity (μV)	Sensitivity for Full S/N (μV)		S/N, A-Weighted (dB)		Distortion (%)						Frequency Response (dB)				Stereo Separation (dB)			Notes
						400-Hz THD			12-kHz THD (IM)			Bass		Treble					
		Mono	Stereo	Mono	Stereo	Mono	L	L+R	Mono	L	L+R	20	40	10K	13K	40	400	10K	
ReVox 10030	1.8*	90	200*	80	56	0.06	1.6	2.0	0.13	9.	11.	-1.0	-0.3	-0.4	-0.6	21	32	22	CT. *Into 75-ohm antenna input. Stereo S/N and THD reflect pilot leakage.
Sansui 9500 213110073	1.4	300	1300	84	74	0.11	0.60	0.14	0.10	1.3	3.4	-2.2	-0.6	-1.1	-1.8	19	20	18	
H.H. Scott LT-112B 1119140	12	200	1000	73	55	0.63	3.6	0.84	0.62	6.4	7.6	-4	-1.4	+1.5	+1.9	28	33	27	
1086238	3.0	2000	4000	76	58	1.1	14.	1.3	1.3	8.0	13.	-4	-1.4	+2	+2.2	32	34	28	CT.
H.H. Scott 4310 191233	2.1	400	1600	74	58	0.30	0.66	0.87	0.40	2.8	3.3	-1.1	-0.2	+1.7	+1.6	24	26	19	Tuning critical; low THD unobtainable by user.
H.H. Scott 310E 220599	2.3	800	1800	76	53	0.48	2.8	1.8	0.9	9.	14.	-6	-2.3	+1.0	+1.2	25	28	22	
H.H. Scott 341 538478	4.0	2000	10K	66	56	0.34	0.84	0.95	0.30	6.4	7.7	-1.8	-0.6	-2.3	-2.7	31	33	22	Hum.
Sequerra One 1232	1.5	80	400	73	69	0.09	0.16	0.11	0.30	3.0	9.8	-0.5	0	0	0	46	47	33	CT.
Sherwood 2300 272 794	3.0	200	25K	76	67	0.31	1.6	1.7	1.1	34.	46.	-2.0	-0.5	-0.3	-0.4	36	39	17	
Sony 5000 1259	—	—	3000	—	71	—	0.70	0.31	—	2.8	10.	0	0	-1.0	-2.5	33	42	34	No mono mode.
Sony 5130 80 1500	1.6	250	500	82	75	0.18	0.42	0.19	0.12	4.6	17.	-1.4	0	-1.3	-2.8	24	24	20	CT. Tuning very critical.
Yamaha CT-7000 1833 IF narrow	3.0	500	1600	84	74	0.15	0.20	0.11	0.09	1.3	4.6	-1.6	-0.5	-0.2	-0.5	40	46	36	CT.
IF wide	3.0	500	1600	84	74	0.11	0.14	0.09	0.09	0.70	4.8	—	—	—	—	40	46	36	CT.

An FM Antenna Survey

Steve Seto

The purpose of this article is to inform rather than to recommend; the data in the accompanying table are provided to give insight into the range of antennas available rather than to isolate the "best" or "most sensitive" antenna in the sample. The fourteen companies listed represent returns from over fifty requests for information sent to companies both in the United States and in Japan during the first half of this year. Of the information that was returned, only that pertaining to FM antennas, or to antennas that have a definite FM application, was chosen for presentation, under the assumption that these antennas have the most to offer the audiophile (as opposed to the videophile, a relatively new breed of perfectionist).

The tables themselves are divided into three major sections: physical characteristics, electrical characteristics, and comments, this last being a catchall section in which as much miscellaneous information as possible has been included. Another important feature is that the address of each company has also been included. Some of these companies offer extensive lines of useful accessories, such as signal splitters and attenuators, and these are noted at the end of the article.

Reading across the top of the table, the column headings are as follows:

- Company/Model—Self-explanatory.
- Type—The code indicates the type of antenna design: Y = yagi, LP = log periodic, T = turnstile, S = S-shaped dipole, C = conical, RE = rabbit ears.
- Length—The length in inches of the central boom.
- Width or Turning Radius—Wherever possible, the turning radius has been listed, because this is probably the most useful measurement. Otherwise, the width of the longest element is listed. Units are inches.
- Weight—The net weight in pounds. Whenever a "less than" symbol (<) appears, the weight listed is a shipping weight.
- Construction—The relative grade of antenna construction. MC = MATV/CATV, generally an industrial grade of construction with heavier tube walls and fittings; H = Home, generally a consumer grade of construction, suitable for most home applications.
- Gain—The relative increase in signal strength in dB delivered compared to a reference dipole. Where a range of numbers is given (e.g., 8.9-9.9), the first number refers to lower frequencies, the second to higher frequencies. These numbers represent the maximum and minimum gain over the entire FM band.
- **F/B** Ratio—The front-to-back ratio of the antenna in dB. If a range of numbers is given, it should be interpreted in the same way as noted for the gain. This ratio denotes how well the antenna rejects signals coming from the rear, such as multipath reflections. Higher numbers are better.

- Impedance—The complex "resistance" of the antenna in ohms. This is merely an operating characteristic of the antenna.

- **HP** B.W.—The half-power beam width is the angle in degrees between the half-power points on the antenna polar pattern. This parameter is directly related to the directivity of the antenna; if a very directional antenna is required, then the smaller the beam width the better.

- VSWR--The voltage standing wave ratio can be thought of as a measure of the amount of signal the antenna receives versus how much of that signal actually gets into the downlead headed for your stereo. If a range of numbers is given, it should be interpreted in the same way as noted for the gain. Lower ratios are better.

- No. of Elements—From the pictures accompanying the product brochures, I have attempted to determine the number of driven and director elements. Because some error is possible, these numbers should be used only as a relative indication of *how the* antenna is *constructed and how* much signal it is likely to deliver. In general, the larger the number of directors, the more narrow the beam width.

- Comments—As explained in the text above.

It is beyond the scope of the article (and the patience of this writer) to give really detailed definitions and explanations for many of the terms listed above; in the interests of space I have given only the simplest interpretations of some terms (such as VSWR) which are actually quite complicated. Fortunately, there is a good article in the July 1976 issue of Popular Electronics, entitled "Guide to Choosing TV & FM Antennas" by Julius Green (pp. 61-65), that presents several very informative graphs. In particular, a graph that shows gain as a function of antenna boom length is very useful and seems to correlate well with the data in this article.

Finally, for those who are interested in antenna accessories, Blonder-Tongue, Jerrold, JFD, and Winegard all publish extensive catalogs. They are the most logical starting point for finding the items you need. If you are looking for an antenna, you will find that Blonder-Tongue, Finco, Scala, and Sitco look particularly good in the tables. Each of these companies provides very good product documentation.

One closing caution: if you decide to buy an MC antenna, be sure that you install it (or have it installed) properly. Improper installation can easily neutralize all that expensive performance.

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DATE JULY 76

COMPANY/MODEL	TYPE	LENGTH (in)	WIDTH (W) OR TURNING RADIUS (T) (in)	WEIGHT (lb)	CONSTRUCTION	GAIN (dB)	F/B RATIO (dB)	IMPEDANCE (ohm)	HP BW (deg)	VSWR (x:1)	NO. OF ELEMENTS DIRECTOR	NO. OF ELEMENTS DRIVEN	COMMENTS
	PHYSICAL					ELECTRICAL							
BLONDER-TONGUE													ONE JAKE BROWN RD, OLD BRIDGE, NJ 08857
YH-FM10	Y	182	66 W	-	MC	12.5	27	75	48	1.5	TOTAL OF 10		HEAVY-DUTY (Y) OR VERY HEAVY-DUTY (YH) CONSTRUCTION,
Y-FM10	Y	182	66 W	-	MC	12.5	27	75	48	1.5	TOTAL OF 10		75 OHM GAMMA-MATCHING FEED, USES F TYPE CON-
YH-FM5	Y	98	66 W	-	MC	9.6	23	75	63	1.5	TOTAL OF 5		NECTOR AND 59 TYPE CABLE OR .412 DIAMETER CABLE.
Y-FM5	Y	98	66 W	-	MC	9.6	23	75	63	1.5	TOTAL OF 5		PRICES IN THE $300 RANGE OR LESS.
Y-FM2	T	-	-	-	MC	-	-	75	-	-	-	2	TWIN ½-WAVE DIPOLE, HEAVY-DUTY CONSTRUCTION
STEREO 8D	LP	141	84 T	-	H	8.5	26	300	50	-	3	8	TOP HOME ANTENNA
STEREO 8	LP	105	51 T	-	H	6.5	26	300	60	-	-	8	SAME AS 8D BUT WITHOUT 3 ELEMENT DIRECTOR
STEREO 5D	LP	105	67 T	-	H	5.5	16	300	60	-	3	5	
STEREO 5	LP	68	35 T	-	H	4.0	16	300	70	-	-	5	SAME AS 5D BUT WITHOUT 3 ELEMENT DIRECTOR
CHANNEL MASTER													ROUTE 209 N., ELLENVILLE, NY 12428
4408	Y	-	-	-	H	8.9-9.9	19.3-18	300	-	-	5	4	DRIVEN ELEMENTS ARE FOLDED DIPOLES, STACKING = +2.5 dB GAIN
4409	Y	-	-	-	H	-	-	300	-	-	3	3	DRIVEN ELEMENTS ARE FOLDED DIPOLES
4401	Y	-	-	-	H	-	-	300	-	-	3	2	DRIVEN ELEMENTS ARE FOLDED DIPOLES
4403	T	-	-	-	H	-	-	300	-	-	-	2	TWIN DIPOLE TURNSTILE
4405	S	-	-	-	H	-	-	300	-	-	-	1	"S" SHAPED SINGLE DIPOLE.
3805	RE	-	-	-	H	-	-	300	-	-	-	1	INDOOR RABBIT EARS
COLOR/CRAFT													13434 RAYMER ST., NORTH HOLLYWOOD, CA 91605
FM30	Y	-	-	-	H	-	-	300	-	-	8	2	STRONGLY RESEMBLES THE FINCO FM-5 IN CONFIGURATION
XM30	Y	-	-	-	H	-	-	75	-	-	8	2	SAME AS FM-30 BUT 75 OHM
FM20	Y	-	-	-	H	-	-	300	-	-	4	2	STRONGLY RESEMBLES THE FINCO FM-4G
XM20	Y	-	-	-	H	-	-	75	-	-	4	2	SAME AS FM-20 BUT 75 OHM
FM10	T	-	-	-	H	-	-	300	-	-	-	2	TWIN DIPOLE TURNSTILE
FINCO													THE FINNEY CO., 34 W INTERSTATE ST., BEDFORD, OHIO 44146
FM-5	Y	120	71 T	<8.5	H	8.9-11.6	15.5-20.7	300	36	1.25-1.40	8	2	DRIVEN ELEMENTS ARE FOLDED DIPOLES, STACKING = +3.5 dB GAIN
FM-4G	Y	93	56 T	<6.5	H	6.8-9.6	16.5-22.5	300	42	1.20-1.35	4	2	DRIVEN ELEMENTS ARE FOLDED DIPOLES, CAN BE STACKED
FM-3	Y	70	49 T	<5	H	4.9-7.8	11.4-9.4	300	50	1.30-2.10	3	1	DRIVEN ELEMENT IS FOLDED DIPOLE, CAN BE STACKED
FMT	T	-	-	<3	H	<0	-	300	-	-	-	2	TWIN DIPOLE TURNSTILE
MODEL 5	S	-	-	<1	H	<0	-	300	-	-	-	1	"S" SHAPED SINGLE DIPOLE.
FM-10GMC	-	186	-	-	-	11.9	21.6	-	56	1.15	-	-	
FM-6GMC	-	120	-	-	-	9.2	18.1	-	58	1.15	-	-	CUT-TO-FREQUENCY FM ANTENNAS, NO OTHER
FM-3GMC	-	70	-	-	-	6.4	15.6	-	62	1.2	-	-	INFORMATION AVAILABLE.
FM-2GMC	-	24	-	-	-	2.8	10.2	-	68	1.3	-	-	

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COMPANY/MODEL	PHYSICAL: TYPE	LENGTH	WIDTH (W) OR TURNING RADIUS (T)	WEIGHT	CONSTRUCTION	ELECTRICAL: GAIN	F/B RATIO	IMPEDANCE	½P BW	VSWR	NO. OF ELEMENTS: DIRECTOR	DRIVEN	COMMENTS
JERROLD													200 WITMER RD., HORSHAM, PA 19044
J33-FM	Y	74	–	<10.5	MC	6-7	18	75	–	<1.5	TOTAL OF 5		HEAVY-DUTY CONSTRUCTION, GAMMA MATCHING, F-61 CONNECTOR
QFM-9	LP	101	62 T	4.5	H	6.8-7.5	20	300	–	–	4	5	TOP HOME ANTENNA
FM-5	Y	–	–	–	H	–	–	300	–	–	TOTAL OF 5		NO OTHER INFORMATION AVAILABLE
C677M	T	–	–	–	H	–	–	300	–	–	–	2	TWIN DIPOLE TURNSTILE
JFD													1462 62ND ST., BROOKLYN, NY 11219
LPL-FM10A	LP	166	97 T	–	H	9.9	26	300	43	1.5	2	8	JFD CAPACITOR-COUPLED ELEMENTS, TRIPLE BOOM
LPL-FM8A	LP	121	84 T	–	H	8.7	20	300	46	1.8	2	6	JFD CAPACITOR-COUPLED ELEMENTS, TRIPLE BOOM
LPL-FM6A	LP	98	72 T	–	H	8.3	18	300	48	1.5	2	4	JFD CAPACITOR-COUPLED ELEMENTS, DOUBLE BOOM
LPL-FM4A	LP	63	63 T	–	H	6.5	16.6	300	49	1.6	–	4	JFD CAPACITOR-COUPLED ELEMENTS, DOUBLE BOOM
AFM 450	S	–	–	–	H	–	–	300	–	–	–	1	"S" SHAPED SINGLE DIPOLE
AFM 150	C	–	–	–	H	–	–	300	–	–	–	–	"STEREO-CONE" ANTENNA, OMNIDIRECTIONAL.
C 119	C	–	–	–	H	–	–	300	–	–	–	–	CONICAL ANTENNA ESPECIALLY FOR WINDOW MOUNTING
TA 705	RE	–	–	–	H	–	–	300	–	–	–	1	INDOOR RABBIT EARS, OTHER MODELS ALSO AVAILABLE.
KAY-TOWNES													BOX 593, TURNER CHAPEL RD., ROME, GA 30161
FM-10G	Y	146	79.5 T	< 7	H	–	–	300	–	–	8	2	TOP HOME ANTENNA, DRIVEN ELEMENTS ARE FOLDED DIPOLES
FM-6G	Y	99.5	56 T	< 6	H	–	–	300	–	–	4	2	DRIVEN ELEMENTS ARE FOLDED DIPOLES
FM-4G	Y	68	45.5 T	< 4	H	–	–	300	–	–	3	1	DRIVEN ELEMENT IS FOLDED DIPOLE
FM-2G	Y	32.5	36.5 T	< 4	H	–	–	300	–	–	1	1	DRIVEN ELEMENT IS FOLDED DIPOLE
FMBD-MPG	Y	24	33 T	< 3	H	–	–	300	–	–	–	2	DRIVEN ELEMENTS ARE FOLDED DIPOLES
FMND-1G	T	–	–	< 2	H	–	–	300	–	–	–	2	TWIN DIPOLE TURNSTILE
FMOD-G	S	–	–	< 2	H	–	–	300	–	–	–	1	"S" SHAPED SINGLE DIPOLE.
MORTEK													2409 MAIN ST., EVANSTON, ILLINOIS 60202
OCL-200	–	7.5	4 W	–	H	–	–	300	–	–	–	–	INDOOR FM ANTENNA WITH MULTIPATH COMPENSATION [8-25 MI.]
CL-100	–	7.5	4 W	–	H	–	–	300	–	–	–	–	INDOOR FM ANTENNA WITH MUTIPATH COMPENSATION [0-9 MI.]
RCA													2000 CLEMENTS BRIDGE RD., DEPTFORD, NJ 08096
1BG06	LP	58	43 T	–	H	6	13	300	65	–	1	5	FAIRLY GOOD SYMMETRY OF POLARS
SCALA													1970 REPUBLIC AVE., SAN LEANDRO, CA 94577
CLFM	LP	90	68 W	30	MC	7.0±½	25	75	52	1.4	–	8	VERY HEAVY DUTY, EXCELLENT POLAR SYMMETRY, TYPE "N" CONNECTOR UNLESS OTHERWISE SPECIFIED, ALSO AVAILABLE IN 50 Ω IMPEDANCE, ABOUT $400
													SCALA ALSO OFFERS THE FMO OMNIDIRECTIONAL FM ANTENNA AT ABOUT $200; NO OTHER INFORMATION

4

Company/Model		Physical: Type	Length	Width (W) or Turning Radius (T)	Weight	Construction	Electrical: Gain	F/B Ratio	Impedance	HPBW.	VSWR	No. of Elements: Director	Driven	Comments
Sited														10330 NE Marx St., P.O. Box 20456, Portland, OR 97220
CA-8-	1-FM	Y	146	63 W	15	MC	15.0-12.0	22	75	32	-	7	1	Single bay, "F" connector, F/S Ratio = 17 dB, about $104
CA-32-	4-FM	Y	146	174 W	110	MC	20.5-9.0	39	75	22	-	28	4	4 Bay array of CA-8-1-FM, 70" high, F/S Ratio = 21.7 dB, $540
MA-5-	1-FM	Y	71	63 W	11	MC	11.0-8.0	16.9	75	35	-	4	1	Single bay, "F" connector, F/S Ratio = 18.5 dB, about $67
EM-5-	1-FM	Y	71	63 W	24	MC	-	-	75	-	-	4	1	End mounted MA-5-1-FM, about $92
EC-8-	1-FM	Y	146	63 W	27	MC	-	-	75	-	-	7	1	End mounted CA-8-1-FM, about $124
EM-10-	2-FM	Y	71	63 W	45	MC	-	-	75	-	-	8	2	End mounted, 2 vertical bay MA-5-1-FM, about $182
EC-16-	2-FM	Y	146	63 W	51	MC	-	-	75	-	-	14	2	End mounted, 2 vertical bay CA-8-1-FM, about $248
EM-10-	25-FM	Y	-	-	46	MC	-	-	75	-	-	8	2	End mounted, 1/4 λ staggered MA-5-1-FM's, 2 bay, $196
EC-16-	25-FM	Y	-	-	53	MC	-	-	75	-	-	14	2	End mounted, 1/4 λ staggered CA-8-1-FM's, 2 bay, $266
SHD-8-	1-FM	Y	-	-	<15	-	-	-	300	-	-	7	1	Single bay, about $89
SHD-16-	2-FM	Y	-	-	<31	-	-	-	300	-	-	14	2	2 vertical bay SHD-8-1-FM, about $182
SHD-9-	1-FM	Y	-	-	<19	-	-	-	75	-	-	8	1	Single bay, about $100
Spico														Spirling Products, Henrietta & Duffy, Hicksville, LI, NY 11802
	FM-5	RE	-	-	<2	H	-	-	300	-	-	-	1	Indoor rabbit ears with tuneable dipole and impedance matching. Other models also available.
Telrex														Box 879, Asbury Park, NJ 07712
TCMS 10-FM		Y	150	67 W	21	H	11.9-11.1	19+	300	-	1.23	9	1	Ruggedized, driven element is folded dipole, $105
SX100/FM		Y	60	67 W	6.5	H	10	22	300	49	1.3	4	1	Ruggedized, driven element is folded dipole, STKG = +3 dB GAIN
MBXFM		C	-	-	-	H	10.0-10.8	20	300	55	-	-	-	4 vertical bay array, about $75
Winegard														3000 Kirkwood St., Burlington, Iowa 52601
CH-	6065	LP	142	78 T	7	H	9.6-10.6	20	300/75	60-47	-	5	5	Fair polar symmetry, top home antenna
CH-	6060	LP	60	42 T	5	H	8.6-8.2	20	300/75	70-63	-	2	4	Good polar symmetry
	CB-45	Y	94	52 T	-	H	-	-	300	-	-	5	3	End fire array
	CB-40	Y	34	38 T	-	H	-	-	300	-	-	2	2	End fire array
	FM-3T	T	-	-	-	H	-	-	300	-	-	-	2	Twin dipole turnstile
	FR-1	S	-	-	-	H	-	-	300	-	-	-	1	"S" shaped single dipole
	PB-10	RE	-	-	-	H	-	-	300	-	-	-	1	Indoor rabbit ears, other models also available

A Flexible Approach to Preamplifier Controls

Charles T. Pike

[Ed. Note: We all are continually looking for a combination of minimum distortion and noise with maximum flexibility in our preamplifiers. This may explain why the Ace Zero-Distortion "preamp" has proven so popular. You will be happy to note that you can do better yourself; Massachusetts member Charles Pike has done so, and while raising his unit to a much higher degree of flexibility than the Ace offers, he has retained its *passive* nature, and thus the inherent low distortion and noise of non-powered devices.—J.B.B.]

Due to the proliferation of separate preamplifiers lacking gain controls and switching, the passive switching arrangement that I use with my preamplifier might be of some interest. Figure 1 is a schematic diagram of my switching arrangement. By using double-pole, double-throw switches with a center off position, we may select either of two 'bus lines." Let us call the *bus* that goes to the gain controls A and the other bus B. Note that any L-R pair of jacks may be used as either inputs or outputs.

An example of the way this system works follows: The inputs of a tape recorder are plugged into input 1, the tape recorder's outputs are plugged into input 2, and a phono preamp is plugged into input 3. Then if inputs 1 and 3 are switched to bus B and input 2 to *bus* A, the phono preamp is connected to the tape inputs, and the tape outputs go to the gain controls for monitoring. By returning the input 2 switch to its center (off) position, thus disconnecting the tape outputs from bus A, another source, such as a tuner connected to input 4, may be switched to bus A, thus permitting one source to be taped while listening to another without disconnecting any leads. I have found this arrangement to give great flexibility; however, <u>the user must remember to return the unused input switches to their center positions.</u>

I gave quite a bit of thought to the design of the gain controls. At first I planned to have separate balance and gain controls, but this adds complexity. Therefore, I chose separate gain controls for left and right channels. By using stepped logarithmic attenuators, some of the disadvantages of separate controls are overcome. If the controls are set for correct balance and it is desired to increase gain, it is necessary only to increase both controls by the same number of steps, maintaining the same offset, and balance is maintained. Figure 2 is a diagram of one gain control. A single-deck, 24-position switch *is* used. The total resistance *is* 10 kfl, and the resistance ratios are chosen to give 2-dB steps (except the last few at the low end, which must be larger in order to get to 0). With 2-dB steps the balance may always be set within 1 dB, which I have found satisfactory.

The parts should be mounted on an enclosed metal box such as a "Minibox" for shielding. The double-pole, double-throw switches must have a center off position. I used large toggle switches (Lafayette 99R 61483, $0.99 each) because they were cheaper and more readily available than the miniature type at the time. Radio Shack lists a miniature type (275-620) for $1.79 each. Any number of switches may be connected to the bus lines. Five are shown as an example only. The resistors for the gain controls are 1/4-watt carbon resistors with 5% tolerance. The 24-position switches are Mallory 13124L.

It is important that the equipment to be used with the 10 MI gain controls be capable of driving this impedance. I understand that the Davis-Brinton phono preamp will drive this without difficulty. Switching flexibility could be improved by connecting another DPDT switch to connect the gain controls to either bus A or bus B; however, it is important that the output impedance of associated equipment be such that the 10 MI loading does not cause an apparent gain change when switched to the tape input bus while recording.

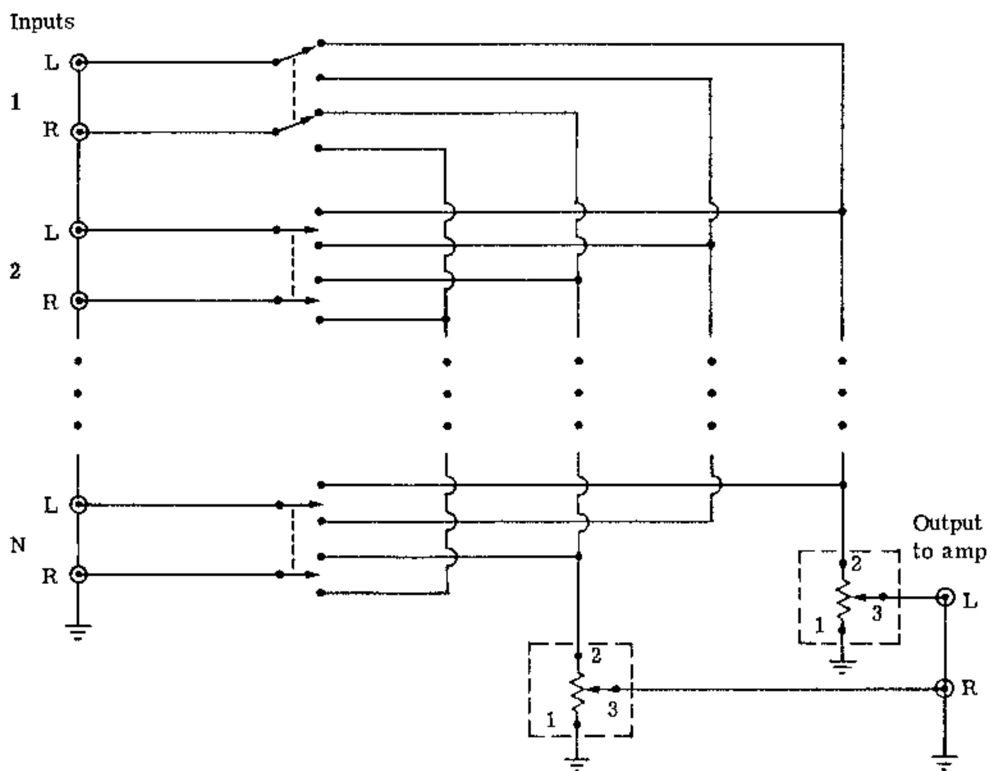

Fig. 1. Schematic of preamplifier switching arrangement

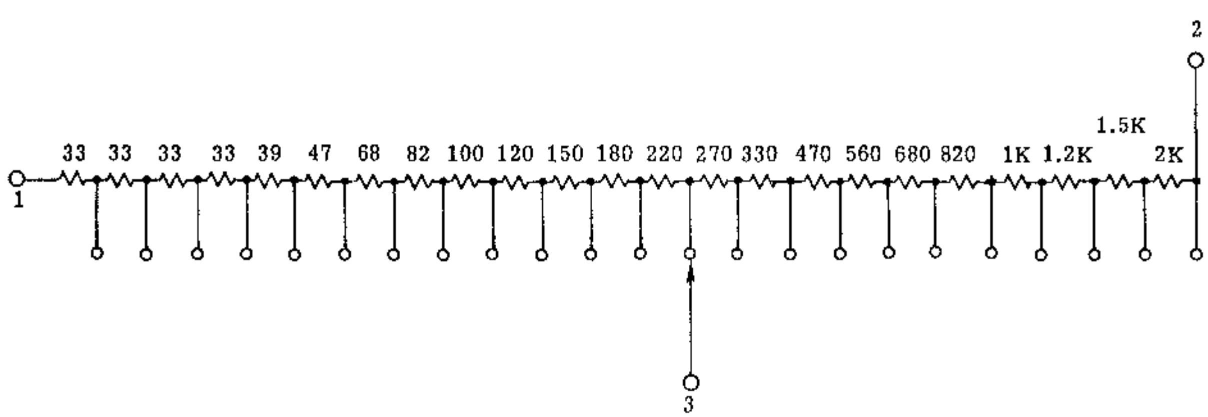

Fig. 2. Gain control schematic

Equalizing With the BSR Sound Level Meter and Other Instruments

Daniel Shanefield

The inexpensive sound level meter (SLM) that used to be sold by Radio Shack is now available from Lafayette and other BSR dealers. When using this instrument to equalize my playback system, I always got lousy-sounding results. So I tried to calibrate the SLM against a standardized microphone. With the first arrival of warm weather, I set up a comparison out of doors, mounting all the transducers up on ladders. A frequency sweep was played through an old EPI 100 (circa 1973), which came with its own individual anechoic response curve and has a super flat response. The SLM was compared to an AKG 202E (also supplied with an individual curve) and a Thermo-Electron 814 (not encased). Except for the 814's well-known small peak at 15 kHz, the EPI, the AKG, and the 814 were all in general agreement. By comparison, the SLM exhibited a response that was very close to a C-weighted noise measurement curve. This type of meter response is supposed to be –6 dB at 20 Hz, –2 dB at 50 Hz, flat in the range from 90 to 3000 Hz, down to –4 dB at 10 kHz, and –12 dB at 20 kHz. Actually, my meter was –6 dB at 10 kHz.

When I had equalized with this meter, thinking its response was flat, I had mistakenly increased the response of the playback system by 6 dB at 10 kHz. My system consequently became shrill, and strongly accentuated tape hiss.

This SLM was reviewed in Audio (July 1971), and the approximation of a C-weighted curve was mentioned. A more detailed review appeared in Stereo Review (August 1972), in which the C-weighting was also noted and a frequency calibration curve was plotted. The sample that Stereo Review got (from Radio Shack) was even worse than mine; it was down 10 dB at 10 kHz.

I called up Mr. Kurt Mathieson, the technical service man at BSR in Blauvelt, N.Y. He said that their SLM is weighted because it was originally meant to be used as much for noise abatement measurements *as* for hi-fi. When I told him that there must be a lot of audiophiles who have been led into using wrong equalization curves because of this, he said the meter is now sold by BSR for use with their test record, which is "probably" C-weighted to compensate for this. He wasn't sure!

The strange thing *is* that the instruction manual doesn't warn the user about the weighting. I have a feeling that such "weighting" is pure accident because the C curve itself is rarely used, except for hazard estimations with very loud sounds. I'll bet it's just what they happened to get with a cheap microphone.

You can probably do better by making your own equivalent to a sound level meter, as suggested by Jim Nichol in the September 1975 Speaker. Or, just run the output of a good microphone into a tape recorder and use the recorder's VU meter. Most such meters have flat response (with Tandberg's being an exception).

But with any equalization process, I strongly suggest getting into the ballpark with instruments such as an SLM, and then fine tuning the equalization by ear.

For a typical complaint by an audiophile who got "disappointing" results when trying to equalize without using the ear-plus-music fine tuning, see High Fidelity magazine (May 1976, page 24). I have heard many similar complaints from others. Also, for guidelines on what to listen for when using the ear method, see my article in Stereo Review (May 1976). That method has worked well for many people.

The initial process of getting into the ballpark with the 13SR SLM is much improved if the C-weighted curve is assumed for the meter and is compensated for by adjustments of opposite sign. In other words, any reading on the SLM for 50-Hz sound should be increased by 2 dB, and at 10 kHz by about 4 dB, and so on. (Note that the readings are to be increased, not necessarily the equalizer settings. They come later.)

There is a different method for playback equalization that I think is the very best way to go, though it leans pretty heavily on instrumentation. Here is the procedure:

1) Attach the playback power amplifier output to a VTVM or transistorized voltmeter. Run the VTVM's output (which normally feeds the meter) through an RC low-pass filter. Now run the output of the filter into an automaitc "strip chart" voltage recorder, so you can record the power amplifier's output. I prefer a linear voltage scale rather than a logarithmic dB scale, for reasons explained below.

2) Play a standard 20 Hz to 20 kHz swept frequency signal recording through the system. Adjust the low-pass filter's R and C values so that the recorder trace barely wiggles at 20 Hz. Then the "ballistics" of your strip chart recorder will be just about right for good sensitivity during a later step. (Without the filter, many modern strip chart recorders wiggle excessively at the lower audible frequencies.) The trace should be fairly flat from 30 Hz to about 17 .kHz, which I think is the important audible range.

3) Now unhook the power amplifier from the input of the VTVM and attach a good omnidirectional microphone with flat response, such *as* the 814, to the VTVM. Switch the VTVM to a more sensitive range, and place the microphone where your head would be during ordinary listening. Hook the loudspeakers up to the power amplifier of your playback system.

4) Play the standard swept frequency recording again. The sweep should take about a minute, so that it is slow enough to allow the buildup of room resonances. (In my experience, pink noise changes too fast and therefore is not as good as continuous sine waves for achieving good-sounding equalization.) Note the recorder trace, with all its horrible room resonance effects. You'll probably see from 50 to 100 major peaks and valleys !

5) Insert a graphic equalizer into the playback system, and adjust it until the overall trend of the trace is as flat and non-horrible as possible. When a linear voltage scale is used in the recorder rather than a dB scale, the weighting of a peak is very much exaggerated, as opposed to the weighting of a valley. But this is just the way our subjective impressions of the sounds register on our minds: peaks are thousands of times more unpleasant than valleys. (Well, maybe hundreds of times.)

When excellent recordings such as some of the most recent releases by Philips and DGG are played over a system that has been equalized by this method, very little fine tuning is necessary, and some really good sounds can be achieved.