THE B.A.S. SPEAKER

EDITOR-IN-CHIEF: Michael Riggs

COORDINATING EDITOR: Michael Riggs

STAFF: Robert Borden, Joyce Brinton, Frank Farlow, Scott Kent, Mark Saklad, John Schlafer, Jack Stevens

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In This Issue

This year's hottest audio topic is almost certain to be Bob Carver's "Sonic Hologram" circuit. In this month's feature, Al Foster dives right into how the thing might work and some of the complications involved. This is one you won't want to miss.

Elsewhere, Roy Allison and Victor Campos take a close look at Dick Greiner's recent articles on loudspeakers and power requirements. Scott Kent proposes a polarity standard for tape heads, and Dick Greiner returns with further comments on the damping effectiveness of the Stanton brush and on loudspeaker design. Plus we have some interesting notes on new audiophile discs and the demands they make on cartridges.

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*Several members of the Saint Louis Audio Society have had a local machinist fabricate 45 rpm adaptors for Linn Sondek turntables. These have been completely satisfactory, and the "extras" made are available to BAS members for $22 each. These are for 60 Hz use only. Send a check or money order to 45 rpm, 8319 Ardsley Drive, St. Louis, MO 63121.
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*Classical records by Musical Heritage Society, large selection, played once, as-new condition. Will sell singly or in bulk. $3 each, or best reasonable offer. Must be picked up at my home in central New Jersey. Send SASE for list. Call (201) 949-4579 weekdays, or write Ronald B. Freeman, Room 2G411, Bell Labs, Holmdel, NJ 07733.

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*Grace 707 Mk. II tonearm; Signet TK7E cartridge (aluminum cantilever); Linn Sondek turntable. Mike Lulejian, (404) 321-4036 (Atlanta) after 7 PM.

Discounts

Videotape

BAS members might like to know where to find videotape at reasonable prices. Sound Machine, 2836 Kennedy Boulevard, Jersey City, NJ 07306, (800) 526-6070, has L-500s and L-750s for the absolute rock bottom prices of $12.99 and $16.99 respectively, plus shipping. They take Visa and Master Charge, and you can order over the phone toll free. They're fast, too. They handle Sony, Fuji, and Scotch (I think) tapes, but I have found Sony to be the best. Sound Machine seems to have very low prices on hardware, too.

-- Bob Craig (Illinois)

Stereo Discounters Defended

In the March issue of the Speaker, there is an article critical of Stereo Discounters in Baltimore. Over the past three years, I have dealt with them approximately six times and have been well pleased with both their prices and their service. In fact, the only "slow" shipment I've encountered was when U.P.S. was on strike and my order was shipped by the U.S. Postal Service.

There is one firm that I've had bad experiences with, and that's Stereo Corporation of America in Brooklyn, NY. One order I placed was never filled, and the refund of my money took approximately three months. On another occasion they were very slow in shipping a Pickering cartridge. -- Buford Reynolds (Tennessee)

Corrections and Complaints

From Ball Brothers

I just reviewed my copy of the February Speaker, and I compliment you on your thoroughness in reporting on the talk Bob and I gave at the meeting. There are a couple little points of misunderstanding I would like to clear up.

On page 22 in the section on wear and friction, the records described after 125 plays are not those shown in figure 2. Figure 3 should be referenced in paragraph 4 of that section. It shows a cleaned record and one with an experimental coating (not Sound Guard) after wear testing. The purpose of this photo is to demonstrate how the amount of wear varies on different areas of records because of some variation in the record itself. Similar tests with records coated with the Sound Guard preservative show almost none of the visible wear debris.

On page 24, paragraph 4, there is a little confusion on the use of record cleaners. The statement is made that the cleaners should be allowed to evaporate to be sure the record is dry before using the preservative. It is important with any wet cleaner to wipe off as much of the fluid as possible because, the last area to dry is down in the grooves, and if the fluid evaporates it will redeposit the contaminants in this most critical area.

In the same paragraph, there is a misunderstanding about how many sprays of the preservative to use per side of the record. I failed to make myself clear. It is twelve sprays per side, not six sprays per side (twelve per record). Also, on page 24, paragraph 5, we did not evaluate the Keith Monks machine, but the type of fluid that is used in his system. As stated, the recommended fluid is the alcohol/water mixture. -- Dr. Virgil Friebel, Ball Brothers Corporation

From Stephen Temmer

Someone has sent to me an undated excerpt from your magazine in which you have made statements about my involvement with the Digital Standards Committee and have reprinted (without permission) all of the material that was mailed to former members of that committee. If you were among the small group that was permitted publication rights, please excuse the accusation.

First of all, I would like to correct those people who have stated that there was a "court injunction," that I "stopped the DSC, together with the Department of Justice," etc. Neither of these statements is true or even remotely accurate. Your own printing of my involvement states the matter clearly: the AES lawyers advised the AES Board of Governors, of which I am a member, not to pursue the digital standards matter and in an appearance before the Board advised them to accept the newly formulated guide lines on standardization, which had been quickly assembled from similar documents of other societies as a result of the "threat" of an investigation. As of this date, I know of no contact between the Justice Department and the A. E. S.

A Jim Brinton is quoted in those pages as saying "since digital recording offers so much potential to the audiophile ..." I would like to refute that statement as a bottom-line summation, even though I am willing to admit that there is some benefit from that technique. As the first engineer in New York to operate a broadcast Ampex recorder in 1948, I have seen even that "great benefit to the audiophile" replaced by records costing three times as much as ordinary ones, just to eliminate the tape recording. The digital aftermath, I believe, will be much much worse.

-- Stephen F. Temmer, President, Gotham Audio Corporation

From dbx

It would appear that Jim Brinton has acquired a crystal ball that mysteriously allows him access to information of somewhat dubious credibility. I am referring to Mr. Brinton's note in the January Speaker concerning dbx. Though he is correct in stating that dbx is now a part of BSR, that dbx's high-quality customer service remains, and that the company seems financially sound, these are the only points about which he is correct.

BSR would hardly need to acquire dbx, as Mr. Brinton suggests, to use dbx's low-cost manufacturing expertise "to make its (BSR's) own lines more profitable." BSR is, after all, the largest manufacturer of turntables in the world, producing more turntables than everybody else combined. Any British company that can beat the Japanese at making anything, especially hi-fi gear, must already have plenty of low-cost manufacturing expertise. It might be more to the point if BSR were suspected of desiring to make use of dbx's high level of electronics technology.

Although Mr. Brinton may be aware of "management funding decisions (read: cuts)," and "circumstances which have crippled the company's advanced development capabilities," those of us here at dbx have seen nothing of the sort over the past few years. We introduced three new products at the November 1978 AES Show: the Model 148 broadcast noise reduction system and the Models 163 and 165 compressors. At the January 1979 CES Show, shortly before I received my copy of the Speaker, we introduced three more new products: two dynamic range expanders in the 3BX family, the 2BX and 1BX, and a remote control device for the 3BX, the 3BX-R. I suggest a listening session with some of our new products might be in order before deciding whether our "advanced development capabilities" have been "crippled." (Presumably any recent funding cuts would have hit after these products were out of the design pipeline. -- MR)

Though it is understandable that the BAS does not hold editorial sway over contributions to the Speaker, it does seem inappropriate that Mr. Brinton, while holding the positions of Publisher of the Speaker and President of the BAS, would not undertake a bit more research than he evidently did before writing such a statement. Perhaps a phone call to dbx, or better yet, to a BAS member who also works at dbx, might have served to set the record straight.

-- Leslie B. Tyler, Chief Engineer, dbx, Inc.

Allison and Campos Reply to Greiner

In the introductory remarks to his article "Loudspeakers -- Power Requirements, Distortion and More" (March 1979 Speaker), Dr. Richard Greiner accuses many loudspeaker designers of ignoring the laws of physics. He is probably right about some of us, but then he goes on to do exactly the same thing himself. A little exaggeration and worst-casing may be forgiven someone promoting his point of view. Dr. Greiner, though, is so far off in his premises and relies on so many half-truths that his conclusions on loudspeakers have no credibility. In this he is not alone, of course; it seems that there is no other area of high fidelity where so many writers ignore the laws of physics.

I should like to address some of the major problems with Dr. Greiner's presentation in the order of their appearance. But first, let us look again at the expression for loudspeaker system low-frequency output:

$$\text{Power output} = (\text{amplitude})^2 \times (\text{diameter})^4 \times (\text{frequency})^4 \times (\text{a constant}).$$

It is clear that, if we wish the power output to remain constant with frequency, the cone-motion amplitude must be inversely proportional to the square of the frequency. It is quite possible to arrange this for the frequency region from first resonance up to 1 kHz or so, depending on the cone diameter. But to achieve flatness of power output, the Q of the first resonance must be close to 1. If it is less than 1 (if there is too much damping), the power output will roll off at low frequencies. If the Q is much higher than 1 (not enough damping), there will be a peak at the bass resonance, with some audible ringing if it is high enough. The point I wish to make is that there is a proper value for Q, and it is not the smallest value obtainable.

1. In the middle paragraph on page 2, Dr. Greiner states that "small boxes and heavy small-diameter pistons" have two major problems with no solutions. The two problems cited are:

A. "If the piston is made massive, as it must be to get a low resonant frequency, the Q of the speaker rises and the little-box/piston combination booms at one frequency." This is sheer nonsense. Piston mass is only one of the factors determining Q; the complete expression is $Q = (2\pi fM)/R$, where M is total moving mass including air load, and R is the sum of all resistances (including electromagnetic damping) in the equivalent circuit. It is true that if we start with a system that is optimally designed (with the correct value of Q) and do nothing to it but increase the diaphragm mass, the Q will increase. But even the least sophisticated designer knows that the proper compensatory measure is to increase R, preferably by increasing the Bl product. That will restore the original Q. A heavy cone does not cause poor transient response or boominess. Those problems are caused by the wrong Q, and Q is controllable by the designer.

B. "Second, to give enough output the piston must move so far at low frequencies that both amplitude and frequency modulation distortion become a problem." We might ask what is "enough" output, what size piston are we talking about, and what frequency range is to be covered? It certainly is true that all woofers have output limits, and when those limits are reached we have distortion problems. But they have nothing to do with the size of the box. They are completely determined by the size of the piston, its linear excursion capability, and the frequency range it is expected to reproduce.

2. The same paragraph contains other misleading generalizations. It is stated that "the long voice coil necessary for long-throw pistons reduces the efficiency of the speaker and raises the Q even more." Again, if we start with an optimized system design and simply add turns of wire to the voice coil, the statement is true, because we have, in effect, added a resistance in series with a system not designed for it. But no rational designer would do such a thing. He might, instead, increase the diameter of the voice coil wire sufficiently to obtain the desired additional length with the original resistance value. Because of the larger wire cross-section, there would then be fewer turns in the magnet gap, decreasing the Bl product. Increasing the value of B (flux density) would restore the original Bl product. Result: same response, same Q, and same efficiency as before, but with the needed amount of coil overhang. The only cost of the overhang is a bigger magnet.

3. On page 3, sound pressure level on axis is compared with the reverberant field SPL in a listening room. It is stated that SPL in the reverberant field is typically 3 to 5 dB lower than the anechoic SPL one meter on axis. This relationship actually depends on the size of the room and the absorptivity of its surfaces and furnishings, and also on the dispersion of the loudspeaker system. I have found that the two SPL figures are very close -- typically within 1 or 2 dB -- for wide-dispersion loudspeakers in normally furnished rooms of normal size (2, 500 to 3, 000 cubic feet). Perhaps Dr. Greiner's listening room is atypically large.

4. On page 4, a four-way speaker system with a 15-inch woofer is recommended. Dr. Greiner asserts that "high efficiency can be achieved with such a system since a short-throw woofer can be used." In fact, the attainable efficiency is determined by the box size and the low-frequency extension of the system -- not by the woofer size or "throw" (excursion capability).

5. Dr. Greiner warns us on pages 4 and 5 not to use pads, or even crossover chokes; that pads "degrade the damping of the system and allow it to resonate on its own" and that "the low pass inductor in series with the woofer has the serious flaw of introducing series resistance between the amplifier and woofer voice coil ..., (which) alone will change some otherwise good driver-enclosure combinations into boom boxes." Related to this is the statement: "All drivers should have large magnets so that the driver motor is strong and the electrical system has good control over the mechanical system." All of these assertions are based, apparently, on the naive assumption that the more damping there is, the better; that the lower the Q of the system, the better. But there are no rational grounds for such a belief. Too low a Q sacrifices low-frequency response with no audible benefit in better "transient response." As we have seen, Q is controllable in various ways, and the designer of a complete system can -- and should -- design the drivers so that Q values will be correct with the crossover network elements in the circuit. In doing so, he must select the proper magnet size for the purpose: not too small, but not too large either.

Matters such as these really are understood by competent designers of loudspeaker systems. Obviously they are not well understood by everyone, and it is dismaying to see someone of Dr. Greiner's stature simply repeating the old myths -- especially in a tutorial context.

On the other hand, his use of the Beers and Belar formula for calculation of frequency modulation distortion puts him in very respectable company. The Beers and Belar paper has been accepted as accurate ever since it was written, and I do not question Dr. Greiner's reliance on it.

Even so, the fact that the Beers and Belar analysis is generally accepted does not make it valid. Loudspeaker tests in anechoic chambers are generally accepted also, but they have been proved to be invalid at low frequencies. And the Beers and Belar FM distortion formula gives results that are too high by a factor of 3.5.

The root of the problem, again, lies in the test conditions. Given the conditions specified in the Beers and Belar paper, the formula is correct. Those conditions, however, do not represent the situation of a loudspeaker system heard in a normal domestic living room, and the differences are significant.

Consider first the question of amplitude, that is, cone displacement for a given low-frequency power output. The environment assumed by Beers and Belar is that of a cone in a flat baffle. This is equivalent to a radiation angle of 2π steradians (half-space). In a typical listening room, however, the woofer will not be more than a few feet away from a three-boundary intersection. At 30 Hz, this is a very small fraction of the wavelength being radiated, and the effective radiation angle for the woofer at this frequency is only $\pi/2$ steradians. The actual power radiated is increased by a factor of four. Consequently, for a given power output at a given low frequency, the diaphragm amplitude in a typical listening room is half the amplitude given in the Beers and Belar formula. Because the FM distortion products are directly proportional to displacement amplitude, their predicted value will be cut in half by this one factor alone.

But wait -- there's more! The formula only quantifies the relative distortion content in the power emitted straight ahead, in alignment with the cone displacement vector. It does not allow for the fact that loudspeakers emit low-frequency energy omnidirectionally, which the listening room then redirects to the listener; nor does it take into account the fact that there is no frequency modulation of energy emitted at right angles to the cone motion.

Let us define the direction of cone motion (on-axis) as 0°, and the amplitude of displacement as A. If we consider the woofer's radiation at any angle θ relative to the axis, the effective displacement amplitude at that angle is $A\cos\theta$. Put another way, as we look at the radiation emitted at angles varying from on-axis to 90° off axis, the effective component of cone displacement changes gradually from A to 0 as $\cos\theta$ changes from 1 to 0.

Finally, we must consider the relative amount of radiated power represented by each value of θ from 0° to 90°, and we find that it increases with the angle. That is so because, as θ increases, each equal increment of θ represents a ring of larger diameter on the surface of an imaginary hemisphere with the woofer at its center; that is to say, larger values of θ represent larger solid angles of space. More total power is radiated from the woofer at larger angles of θ, for which $A\cos\theta$ is small, than at smaller angles for which $A\cos\theta$ is large. This additional weighting factor is $\sin\theta$.

To make the Beers and Belar formula represent reality in a normal listening room, then, we must take the following steps:

1. Multiply the displacement amplitude values by 1/2, thus compensating for the difference in radiation loading provided by a real room in comparison with half-space;

2. Multiply the adjusted (halved) value of A by $\cos\theta\sin\theta$, so as to obtain the correct proportion of distortion sideband power to the total power radiated by the speaker. The average value of $\cos\theta\sin\theta$ computed from 0° to 90° is 0.32. Because distortion is specified on the basis of sound pressure rather than power, we must take the square root of 0.32, which is 0.56. One-half A times 0.56 is 0.28A.

Therefore, the FM distortion figures given in the chart on page 3 of Dr. Greiner's article should be divided by 3.5 to give a reasonably accurate picture of the dimensions of the problem. When this is done, we find that even a single 10-inch driver can produce 0.1 acoustic Watt at 30 Hz with a relatively inoffensive 1.1% FM distortion of a 250 Hz simultaneous tone. More important, we find that in all practical cases any FM distortion will be completely masked by the nonlinear distortion of the system produced at the same cone displacements. Nonlinear distortion still is the important limiting factor in quality of bass reproduction at high levels and probably always will be.

-- Roy Allison (Massachusetts)

I would like to comment on Dr. Greiner's scholarly dissertation on "Loudness, Room Size and Power Requirements, " which appeared in the February issue of the Speaker.

It is hard to define an "average level" for music, because this level, even when averaged over a long period will vary depending on the type of music being played, the size of the orchestra or band, and the person leading the orchestra or band. For example, when the BSO was rehearsing, recording, and performing for an audience Berlioz' "Romeo et Juliette, " the peak level in the hall was read to be 119 dB SPL during the Finale of the piece. This did not vary by more than 1 dB in any given play-through, and many sections were replayed during this undertaking, so measurements could be made repeatedly. This was true even though, at times, an audience was present. At no time did any reading vary more than 3 dB or so, regardless of the position of the equipment in the hall. For those who will invoke the inverse-square law, it must be remembered that past Row N in the orchestra the reverberant field is pretty much dominant and, aside from variations in the readings caused by standing waves and other similar phenomena, the field will be found to be rather homogeneous. I chose Rows N, O, and P, center, as the basis of reference because they were more-or-less equidistant from the different sections of the orchestra and the ratio of difference in distance between, say, the first violins and the percussion section was not a substantial portion of the overall distance. In all sections of the piece (and remember, multiple and repeated measurements of the same sections were possible), the peak-to-average ratio was a consistent 15 dB. So the maximum average level was 104 dB. Conversely, a small baroque orchestra, such as the Brandenburg Ensemble, generates a peak level of about 90 dB, with an average of 75 dB, in Symphony Hall.

Now, when Ondeko Za rehearsed the new composition "Mono-Prism, " which was performed at Tanglewood, in Symphony Hall, the peak level in the hall read 155 dB. To take care of any reaction, let me say that most of the orchestra members had cotton stuffed in their ears and, after the initial sections, moved as far away from the drums at the edge of the stage as possible. The drum was so big that it fit only part-way through the big side doors of the stage, so that only the front of the drum was onstage, and the rear protruded into the backstage area. As an interesting aside, after the rehearsal, which was accessible only by special invitation, I went to the Green Room and asked Seiji Ozawa what the name of the humongous drum was, to which he replied, as I now recall, "Shudeiko. " When I asked him to translate, he looked at the composer of the piece, Ischii, spoke a few words in Japanese, turned to me and said "Biiig Drum. " At any rate, the experience is impossible to describe -- akin only to that of one standing on the downtown platform of the 8th-Avenue Subway in New York when an Uptown Express passes overhead. The first balcony, and the seats, in which I was sitting during part of the rehearsal, literally trembled, and the sonic impact was actually felt on the face, chest, and abdomen. Such levels and low frequencies are truly beyond the scope of reproduction of any presently available system. Indeed, the equipment I had could not really measure such low frequencies with any degree of accuracy. But to believe this, it must be experienced. Incidentally, these measurements were possible only through the courtesy and permission of the Boston Symphony Orchestra's Members' Committee, without whose cooperation I could not have brought the equipment into the hall.

Over the years, I have made measurements in many halls, and the results seem to be pretty consistent. Large orchestras with large brass and percussion sections pretty much hit absolute peaks of 120 dB SPL in most medium-to-large halls and with most conductors. The variation one generally finds is in spectral energy distribution with the same group, from hall to hall, even if the "sound" identified with the group (that is, Philadelphia Orchestra, BSO, etc.) varies only a little. (An interesting conundrum, as "Car Talk" would say, but the subject of a long dissertation on its own.) There are exceptions, however. When I made some measurements a few years back in the Marlboro Music Festival Hall, which is kind of dry and non-reverberant with an audience, and on the small side as well, I was surprised to read piano peaks of 120 dB with an average of 105 dB while Peter Serkin was performing Messiaen's "Oiseaux Exotiques, " with the percussion ensemble directed by Leon Kirchner. Which prompted me to call him the "loudest piano-playing artist alive, " a title he continually questions me about whenever I have the pleasure of seeing or talking to him. Now, traditionally, the last piece performed at the Marlboro Music Festival is Beethoven's "Choral Fantasy" (Opus 80), and over the years the maximum peak SPL during the Finale has varied only from 117 to 119 dB. Some other absolute peak measurements of interest are (when all are playing fortissimo): French horns = 105 dB, cellos and basses = 87 dB, woodwinds = 90 dB, voice quartet with piano = 92 to 95 dB, timpani = 111 dB. These figures are given just for interest's sake, as these wavefronts may be obtainable only at a rehearsal and may become part of the overall sound during the actual performance. Incidentally, the chorus was composed of 97 singers and the orchestra of 20 violins, 8 violas, 7 cellos, 2 bass viols, 2 flutes, 2 oboes, 2 clarinets, 2 bassoons, 2 horns, 2 trumpets, timpani, and piano.

Now in this small hall, which is almost ideal for chamber music, a string quartet playing Mendelsohn's F Minor Quartet (Opus 80) peaks at 97 dB maximum with long-term average energy between 60 to 80 dB. Incidentally, the background noise in the hall was 52 dB, and certain parts of the music fell below this level, and only the discriminatory capacity of the ear/brain system made it feasible. To throw a real curve in here, let me state that the Finale of the recording available in the stores of the "Romeo et Juliette" was compressed at a ratio of some 10-to-1 just to make it "feasible" as a recording to be played back in the home.

At this time, it should be pointed out that Dr. Greiner seems to have pretty much glossed over a very important facet of sound reproduction through transducers. I refer to linearity (I'm speaking of input-output; not describing frequency response, which is an incorrect usage of the word).

In my not inconsiderable experience with loudspeakers, I've found that the very best in this area of discussion are linear up to an input of approximately 100 Watts and an acoustical output of about 105 dB at 1 meter from the speaker. Above these points, loudspeakers become increasingly nonlinear. The statement above pertains only to direct radiators and not horn-loaded designs (although other negative factors in performance may affect these, size being one of them). These points are pretty much defined in the design of a driver if reasonably extended and low-distortion, low-frequency response is expected. Because nonlinearity increases with a decrease in frequency, a sort of "balance" in design parameters defines so-called efficiency. Generally speaking, the more efficient a driver is the more nonlinear it is in its low frequency region. In other words, a certain amount of voice-coil overhang, and consequently wasted power, is necessary to insure linearity. (Other equivalent factors govern the behavior of electrostatic loudspeakers -- most specifically, hysteresis in the transformers and the linearity of the suspension of the diaphragms; indeed, no full range electrostatics I know of can achieve orchestral peak levels in a regular, 3500 cubic-foot room.) The preceding is stated purposely ignoring factors such as overshoot, ringing, etc., which make an analysis such as the one Dr. Greiner is attempting even more difficult. It should be brought to mind, nevertheless, that relatively little study has gone into the psychoacoustical effect of such factors on a person. For example, during the preliminary studies that resulted in the live-versus-recorded demonstration Acoustic Research gave with a live set of drums and cymbals, no audible indication of integration of peaks was found, although the difference between the live sound and the recorded sound on peaks (although small) was measurable. Another phenomenon of interest was that although one could "clip" an amplifier, with instantaneous recovery from overload (with no ringing), rather seriously on cymbal crashes, the clipping was inaudible -- at least to any great degree. In fact, the most obvious differences audible could be attributable to timbre rather than anything else.

Lastly, Dr. Greiner, at least to me, and because of the power chart, seems to have made the assumption that all amplifiers deliver rated power into all loudspeakers. This absolutely is not the case. Available power into a loudspeaker from an amplifier depends a great deal on the impedance the amplifier sees in any given frequency range, the specifics of design of the safety circuits integrated into the amplifier, and the ability of the output stage of the amplifier to "sink" the current being fed into it by the speaker's back-EMF while simultaneously delivering current to the speaker. So one may find that a given amplifier may deliver different maximum power levels into different speakers and sometimes never as much as power as it delivers into a resistor on a test bench. An amplifier that delivers rated power into all loudspeakers is the exception rather than the rule. A calibrated oscilloscope on the output of an amplifier is, to me, one of the greatest of revelations. This, along with the phase-load and impedance characteristics of a loudspeaker, a spectrum analysis, by section, of the music being played are minimal requirements when trying to determine just what power is going into a loudspeaker.

The point of the preceding is not to demean Dr. Greiner's efforts. On the contrary, it is meant to show, that if anything, he is too conservative in his estimates, and that, in all probability, even more rated power is required for the necessary dynamics.

The entire audio industry seems to be becoming more cognizant of these factors, as more and more powerful amplifiers are becoming available and the mean power being sold is far higher than it was, or could have been, ten years ago. And although these amplifiers are sold at retail purportedly so one can play "louder," the fact remains that at least more of the dynamics are being preserved than used to be the case. Education through experience is much slower than that garnered through schooling or total immersion. However, the public seems to be learning much faster (and so seem the salesmen) than in the case of refrigerators. For example, do you know (or can you find out) what the loss in BTUs/hour is through the walls of your refrigerator?

Thank you for the opportunity to make this comment -- and for reading.

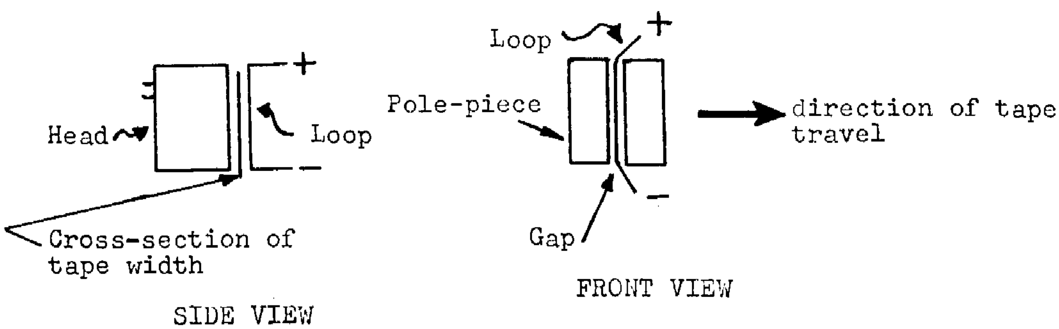
-- C. Victor Campos (Washington, D.C.)

Tape Head Polarity Standard Proposed

This is a suggested polarity standard for magnetic tape heads. Although conventions exist for polarity of loudspeaker drivers and microphones, none appears to be accepted for polarity of magnetic tape heads. (Or for anything else. Given the recent concern over the audibility of polarity reversal, it might be nice to have standards for all equipment. -- MR)

I would appreciate your comments regarding adoption of this standard. Any repetitive asymmetric signal may be used to drive the induction loop, and the signal from the tape head may be viewed on an oscilloscope.

The positive terminal of the tape head shall be that terminal which exhibits a positive polarity when excited by a single-turn induction loop placed parallel, at the tape contact surface, to the gap of the head; where the induction loop positive terminal is nearest to Channel 1 or to the narrow edge of the head gap 90° counterclockwise to the direction of normal operating tape travel, when viewing the tape contact surface. -- Scott Kent (Massachusetts)



More on the Stanton Brush

I have just read Mr. Alexandrovich's very nice response to my comments about the damping effectiveness of the Stanton brush. I have found that I am able to duplicate the response curves he shows for the Stanton 881S with the STR100 test record. I am pleased that we can agree on the data.

When I look at these curves, I see that the peak at 7 Hz is only about 1.5 dB lower than that at 5 Hz without the brush. On the other hand, Mr. Alexandrovich is quite right in noticing that the response for warp frequencies below resonance is reduced by a considerable amount.

I see the glass half empty and he sees it half full. We agree on the data and the method of measurement. As soon as I am able to find the Ortofon test record, I will repeat my tests at full speed. (I still think the advertisement is a tiny bit on the optimistic side.)

I also feel that the Stanton 881S is, in terms of tracking ability and frequency response, one of the top few pickups I would consider using.

-- R. A. Greiner (Wisconsin)

Comments on Music, Physics, and Design

I was really pleased to see Mr. Kapplinger's comments on the difficulties with reproducing the entire sonic frequency spectrum with a single loudspeaker driver. His comments overlap and reinforce my own experience with astonishing accuracy. Page 13 of the March Speaker should be reproduced and mounted on every wall as a reminder of the extraordinary frequency range required of a loudspeaker. It also serves as a reminder of just which frequency ranges contain the primary energy and information in the reproduction of sound.

Because 650 Hz is the approximate geometric center of the audio range, I only wish that the octave below were considered the lower mid-range and the octave above the upper mid-range. The treble clef runs from C to A' (roughly); this is 250 to 800 Hz. I have contended that this important range should be covered with one transducer, so that no crossovers and time alignment problems appear. I am therefore a bit wary of seeing a crossover put at 500 to 600 Hz as is suggested in the chart. In principle, I certainly agree that a four-way or even a five-way system may be essential to really first-quality reproduction. I am pleased to see the absence of the term "sub-woofer." As one who has had speaker systems that reproduce 30 to 20 Hz for at least 30 years, I have always been annoyed with the term. The sub-woofer crowd did not invent the bottom octave, they just discovered it a half century late.

In addition to the frequency-range information provided in the chart, it would be well to remember that for average orchestral music, half of the energy is below 250 Hz and half of the rest in the next two octaves above and so forth. These considerations should be thought about by all audiophiles (and audiophobes) from time to time so that they gain perspective on the frequency and power requirements of loudspeaker systems.

-- R. A. Greiner (Wisconsin)

Notes on Records

Recorded Levels on Modern Discs

I have just added six of the newly released Angel 45 rpm discs to my record collection. These discs are remasterings of some of the standard EMI-Angel catalog. I have compared them in several cases to the EMI version and to the Angel version of these releases in a well controlled A-B-C type setup. There is no question that the Angel 45 rpm pressings are superior in every way. Even the very exciting EMI pressing of the Planets (Holst) is distant, dull, and bland by comparison. The advantages of higher groove speed are apparent.

On the other hand, these discs are very hard to play. Like so many of the current direct and 45 rpm discs, the recorded velocities are very high. I have set up a high-quality, high-tracking-ability pickup along with a flat, calibrated preamplifier and peak-reading equipment so that the actual peak recorded velocities on some of these discs could be documented. Without exception, the Angel discs had many passages with recorded velocities of 25 to 35 cm/sec. In a few cases, I found 45 cm/sec.

It is therefore essential that these records be played only with a pickup that will "stay in contact with the groove wall" at these very high velocities. An inadequate pickup will mistrack and smash up the groove walls with only a few playings. Unfortunately, only a few select pickups will do the job. Always test your pickup (properly set up) with a Shure TTR 103 test record or the equivalent. It must track at least 30 cm/sec, or I would advise not playing these records (or any direct discs). If there is the slightest hint that these discs or any of the direct discs available are distorted in the high level passages, it is bound to be a pickup problem. The pickup, not the disc, should be thrown out.

-- R. A. Greiner (Wisconsin)

Super Records and Why They Are So Hard to Play

I have recently had some sobering experiences with playing high-quality records. First, several persons have brought me records that they considered to be overcut or in some way defective. As it turned out, their copies had been damaged by the pickups they were using. My own copies were perfect. I decided to measure some of the levels recorded on these discs to see just what tracking velocities were necessary to play them without groove damage. A pickup with very high tracking ability (over 40 cm/sec), a flat calibrated preamplifier and peak-reading instruments were used. The problem records had, without exception, levels of 25 to 35 cm/sec. In two cases, 40 to 45 cm/sec velocities were seen several times on the disc. In order to see if these levels were unusual, a selection of forty direct, digital tape, and 45 rpm discs were measured. In addition, a selection of fifty good quality, but otherwise standard, discs from a broad selection of companies and music types were measured. In all cases, levels of 35 cm/sec were common. Many discs had levels of 25 to 35 cm/sec, and over 30 percent had levels that reached 35 to over 40 cm/sec. This was the case whether the recorded program was jazz, full orchestra, or solo harpsichord. It was the case for American, European, full-price or budget labels. It is thus clear that if a pickup does not track at least 30 cm/sec and preferably higher, it will not play many modern discs. Not tracking means that the stylus is flailing about in the groove, smashing it to pieces. Once smashed, the groove will never be the same. Always check a pickup with the Shure TTR-103 tracking test record, or equivalent. If it does not track band four, throw it out (the pickup, not the test record). One playing with a groove smasher will lower the quality of the record and make any further judgments about the quality of the record or of other pickups questionable. Regrettable as it may seem, only a few pickups are able to play all that is recorded in the grooves of a high-quality disc.

-- R. A. Greiner (Wisconsin)

RCA to Issue Audiophile Piano Recording

According to a report in Billboard, the music industry trade publication, RCA will issue a new album of Chopin selections played by Peter Serkin and produced by long time RCA classical specialist Max Wilcox (formerly staff for two decades and now freelance). Wilcox thinks audiophiles will be pleased with the piano reproduction and ambient quality of the recording, made at the American Academy of Arts and Letters.

Many of the "back-to-basics" techniques advocated by sound purists were incorporated into the Serkin disc. Signals from two Schoeps omnis were fed through a custom preamp to a 30 ips recorder. No noise reduction or signal processing was used. The lacquers were cut directly from the edited studio master tape, and plated by Europadisk. Wilcox hand-carried the metal parts to Indianapolis and personally supervised the pressing operation to make sure he got what he wanted.

According to Max, "there's an open, free sound in the top, and with the omni-directional pickup, you're hearing the hall as well as the direct sound. It's state-of-the-art analog recording. I wouldn't know what else to do with it to make it any better." The disc will retail for $7.98.

RCA also has an upcoming release of a Soundstream digital album of the Bartok Concerto For Orchestra with the Philadelphia Orchestra, which will be priced at $9.98.

Perhaps corporate clout can produce a listenable product, if it is forced to do so by seeing substantial dollars flow to the likes of Crystal Clear, Telarc, Direct-Disk, et al.

-- Dick Lewis (Massachusetts)

Stepped Volume Controls

I recently completed the stepped volume control project described in the February issue of Audio. For those of you who, as I do, own a Dynaco PAT-5 in either stock form or in any of its various reincarnations know that a major problem with it is its volume control, as the results of the BAS preamp survey bear out. In an attempt to solve the problem of channel mistracking, I tried an Allen-Bradley Mod-Pot. I was forced to use a unit made up of four sections as Allen-

Bradley did not make a 15k module in the 20% log taper I desired and tracking between channels was very bad to say the least (7.5k ohms for each of the four modules, to simulate a dual 15k control). So when I saw Marshall Leach's volume control project, I started in immediately. Using the taper suggested in the article and substituting 15,000 ohms for the 20,000 ohms he used for the total resistance in the control, I set out to calculate the individual values for the 22 resistors used on each of the two wafers in the Centralab PA-4002 switch. Table 1 gives the attenuation in dB for each position, the calculated resistance value and the value I used in constructing the control (1% RN55 metal film types).

Taking to heart the necessary precautions about being sure the resistors were soldered properly and in the correct order, construction presented no real problems. When attempting to install the switch however, you need to do some work. You will probably notice when you first open the box in which the switch is packaged, that the shaft is somewhat longer than the stock control in the PAT-5 and needs to be cut down. Also, the hole in the mounting panel needs to be enlarged to accommodate the locating tab on the front of the switch, which is not the same distance from the shaft as it is on the stock control.

In use, it is definitely stiffer than many of the other stepped controls found in more luxurious equipment, but this should change with extended use. In my particular listening environment, the volume increases a little too quickly over the first five positions, but is not too objectionable. This of course can be changed by recalculating the resistor values using different attenuation factors in the equation found in the article. -- Thomas Mishler (Michigan)

Table 1

Resistor	Attenuation	Calculated r	1% mil-bell value
1	60 dB	15.00	15.0
2	54 dB	14.93	15.0
3	48 dB	29.79	30.1
4	42 dB	59.45	59.0
5	38 dB	69.69	69.8
6	34 dB	110.46	110
7	30 dB	175.07	174
8	28 dB	122.83	124
9	26 dB	154.64	154
10	24 dB	194.67	196
11	22 dB	245.07	243
12	20 dB	308.4	309
13	18 dB	388.5	392
14	16 dB	489	487
15	14 dB	615.45	619
16	12 dB	775.05	768
17	10 dB	975.6	976
18	8 dB	1228.2	1240
19	6 dB	1546.2	1540
20	4 dB	1946.4	1960
21	2 dB	2450.55	2430
22	0 dB	3085.05	3090
			R_t = 15000.9

Preamp Notes

Preamplifier/Power Amplifier Interfacing: A Missing Link

We recently have found that a number of power amplifiers and accessories designed for connection to the outputs of preamplifiers draw DC input current to bias their input devices during a "warm-up" phase, or even continuously. This first came to light with tube-type power amplifiers with a direct connection to the input grid. In this case, the input tube was worn past its prime and was drawing grid current. A change to a new tube repaired the condition. Subsequently, we have found that it is fairly common for amplifiers to draw such current on the order of 10 to 50 microamps during a "warm-up" time. The time required may be many tens of seconds, even for transistorized power amplifiers. It is apparently the time required for the input coupling capacitor to fully form and reduce its leakage current contribution to negligible levels.

The consequence of this problem in power amplifiers for preamplifier design is that a low output impedance must be maintained at all times, even when the preamp is turned off. This will prevent any transients arising from a changing DC source impedance as the preamplifier comes on. In the Apt/Holman preamplifier, for example, the muting relay used to suppress system transients grounds the output when the preamplifier is turned off. When the relay transfers to the signal, a relatively low, 18 kOhm DC resistance is present, thus keeping the offset voltage caused by the input current low.

Units that have been found to have this problem are listed below. In some cases, the problem is endemic to the design of the power amplifier; in others, the problem was only a sample defect in one unit of the design and may not be indicative of performance of other units of the same make and model in the field. It is certain that this list is not comprehensive.

Audio Research D-150
G.A.S. Ampzilla, Son of Ampzilla
Harman-Kardon Citation 12
Dyna Stereo 70 (with worn input tubes -- changing the tubes cured the condition)
Phase Linear Andromeda III Equalizer

-- Tomlinson Holman (Massachusetts)

More on the Preamplifier We Deserve

Here are some feelings of mine on "The Preamplifier We Deserve," in the same numbered order as the original:

1. Peripheral equipment capability is not a problem with me. I use one turntable, one tape deck, and one tuner.

3. Level balancing would be better without the pot trimmers, which can get noisy, or worse, introduce some undesirable character to the incoming signal. Manufacturers should supply an instruction sheet for adjusting levels with resistors.

4. System checkout signals add unnecessarily to the cost. Test records or external signal generators can be purchased by the more technically oriented.

5. Recording capabilities are better left for the Russound TMS-1.

6. Signal processors are better left for outboard connection.

8. Signal processing is better left for outboard connection.

9. No comment, except, how many people use an extra mono amp?

10. Loudness contour is probably best controlled through the tone controls.

11. Why waste energy with extra lights?

14. Headphones. Why not have the same arrangement as PAT-5? One can drive even electrostatic phones, because the power amp output is first connected to the preamp and then distributed through the special selector switch on the preamp. This eliminates the need for a skimpy external junction box and does not incur the expense of a special amp for the headphones.

This has been my pet peeve since I let go of my PAT-5. I've been very sorry about this aspect. And frequent headphone listening gives a very useful perspective needed to make certain aural verifications of the speaker systems.

Actually, the most convenient place for this type of headphone connection is right in the power amp. But almost nobody ever has done this before. So, why should someone want to start meddling with this? And, would any self-respecting audiophile ever allow himself to plug anything into the power amp (such as a phone plug)? Unthinkable!

15. Safety. My experience is that if you try to keep your equipment to yourself because other people might damage it, they are more attracted. My best solution is to avidly invite the people in the household for a workshop on how to use and enjoy the system. They quickly turn to the transistor radio. Muy rapido.

Then again, you might gain excellent criteria and a good pair of ears if one of those household members is a woman. They hear better than men.

16. Reliability. Very important, although I have never had any problem with hi-fi equipment, in over fifteen years.

17. Manufacturers should maintain servicing facilities. They know their product better than the local repairman, er, person.

My concept of the preamp we deserve is that of the PAT-5 format with only two additions: switchable phono capacitance loads and switchable phono resistive load.

-- Carlos E. Bauza (Puerto Rico)

Hi-Fi Choice: Cassette Decks & Tapes III

This is the third time around for cassette machines in this series, which now numbers eleven books. As with the two previous cassette deck editions, Angus McKenzie is the author. Fifty models are reviewed, thirty-six of which are new to this edition. Fourteen reviews are taken from previous issues, and these are not strictly comparable with the new reviews, as new measurements have replaced old ones in updated books. McKenzie really gives the cassette decks a workout, testing each machine for: replay noise; dynamic range with ferric, chrome, and ferric chrome tapes; line and DIN input noise and compatibility; metering; input, replay amp, and overall distortion with the three tape types; stability; azimuth setting; wow and flutter; overall response, again with three different tape types; user presets; sound quality; and value for money. There is also a comparison chart on tapes.

The test program came from "very high quality master tapes, ... recorded at 38 ips with Dolby 'A' processing." For the tests this was played back on a Studer B67 professional reel-to-reel machine. All levels were very accurately checked before being played back through Chartwell 450 loudspeakers via Quad 405 amplifiers. The program on the master tape was as follows:

1. Tone recording on left only, right only, then left and right simultaneously. These were used for setting recording level accurately, and also for gaining an impression of distortortion and wow and flutter. McKenzie later states that measurements and listening tests for wow and flutter do not correlate. "Somewhat surprisingly, there was better correlation with the measurements when listening on headphones."
2. Pink noise.
3. A speech recording of McKenzie's voice done in an anechoic chamber. This was used in testing Dolby processing, stability, HF compression, distortion, amplifier clipping, and recording meter characteristics.
4. Steinway piano recorded by the author at Queen Elizabeth Hall. This was used to determine transient stability, distortion, response, and wow and flutter.
5. Pop recording.
6. Another pop cut.
7. Part of Stravinsky's "Rite of Spring" recorded by the author in the Royal Festival Hall. This was used for testing the clockwork models, "and it was noteworthy that the very loudest passage recorded satisfactorily on many machines, whilst on some the same passage sounded excruciating."
8. Nagra stereo recording "of underground trains entering and leaving Golders Green station

to show, very clearly, transient positioning, very low frequency performance and high frequency compression, often noted when signals and points hissed as they changed. Many recorders showed bad HF compression on this track, whilst a few showed no compression at all."

9. Elton John's "Rocket Man" for HF sibilance and sharply percussive sound tests.

The test program was listened to perhaps 300 times during the testing of all the decks, "and so we are all heartily sick of it by now."

The decks tested were as follows (BB = best buy, R = recommended): Aiwa AD1250 (BB), AD6300 (R), AD6550/6400 (BB), 1800 (R), AD6800; Akai CS702DII (R), GXC725D (R); B&O 5000; Eumig Metropolitan CCD; Hitachi D220 (R), D850 (R), D900 (BB); JVC KD720 (BB), KD65 (BB); Marantz 5010; Nakamichi 350, 550 (R), 600 (R), 1000II; Neal 103 (R), 302 (R); Philips 2538/2534; Pioneer CTF4040 (BB), CTF7070, CTF9191 (R), CTF1000; Sankyo STD2000; Sansui SC1110/1100, SC3110/3100; Sanyo RD4028 (R), RD5300-2 (R); Sony TC136SD (R), TCK5, TC158SD (R), TC138SD, TC206SD, TCK8B/TCK7 (BB); Tandberg 320 (R), 340A (BB and rave review); Teac A103 (BB), A303; Technics RS615 (BB), RS631 (R), RSM85 (R); Toshiba PC4360 (BB), PC5460 (R); Trio (Kenwood) KX1030 (R); Uher CR240 (R); Yamaha TC511S, 800GL (R).

The following tapes were tested. Group 1: Agfa LNS; Ampex 370; Boots Micro Ferric; EMI Standard; "Poor Own Brand" Pyral Hi-Fi & Sprint; Scotch Dynarange. Group 2: Agfa Ferrocolor and SFD; Ampex Plus 371, 2020, and Grand Master; Audio Dynamics Plus and Super; BASF LH and Super; EMI Super/Boots UDV; Fuji FL; Maxell LN; Memorex MRX2; Philips Super; Pyral Maxima; Pyral Optima/Dixon C99XP; Scotch High Energy; Sony HF; TDK D; Woolworth's Winfield Alpha Plus. Group 3: Audio Magnetics XHE; BASF Ferro Super LH1; EMI Hi-Fi; Fuji FX and FX1; Maxell UD/Dixons Prof. and UDXL1/Hitachi; Scotch Master 1; TDK AD; Woolworth's Alpha Super; Pyral Superferrite. Group 4: Agfa Carat; BASF Ferrochrom and Super Chrome; Fuji FX2; Maxell UDXL II/Hitachi; Scotch Master II and III; Sony Ferrichrome (FeCr); TDK SA.

All tapes were tested for bias requirements, mid-frequency and high-frequency sensitivity, optimum bias, 333 Hz and 10 kHz maximum output level, dropout performance, wow and flutter, noise, dynamic range, printthrough, housing, leaders, and head cleaner. Boots is a pharmacy chain in England, and Dixon is a camera store chain.

This book plus Hi-Fi Choice: Loudspeakers 2, Tuners, and the very latest issue Turntables & Tone Arms are available from me for $5.50 each or $7.00 each via first-class mail at RD2 Box 120E, Milton, Delaware 19968. -- John Tooley (Delaware)

More Personal Experience on Room Equalization

Gerald Larsen is quite correct in his discussion of the need for room equalization. No matter how perfect the source material and the electronics of a system, seldom if ever is the room perfect. I have used equalization for twenty years in many rooms and found that it is always possible to improve the sound with some amount of equalization. Those who deny the use of an equalizer are quite naive about the acoustical properties of rooms and the effect they have on sound quality.

At the same time, it is essential to tread slowly and carefully when trying to equalize a room. It is not possible to get a room "flat" except at one point. Thus, I recommend that several readings, or settings, be done over a reasonable sized listening area and sort of average or compromise setting be used. Excessive settings on the equalizer usually indicate an attempt to get the system too "flat" at one point in space. I have devised a four-microphone averaging system that gives reasonable equalizer settings for a selected area of the room.

If there is enough interest in this technique, I will write a note on it for the BAS Speaker.
-- R. A. Greiner (Wisconsin)

(Let us know if you would like to read more about Greiner's technique. -- MR)

Upcoming Discs from Crystal Clear

During the week of March 18, 1979, the Cathedral of Christ the King in Atlanta was the scene of some of the most glorious, awesome, and stupendous sounds I have ever heard. The occasion was a series of Crystal Clear direct-to-disc recordings of the Cathedral's Fratelli-Ruffatti pipe organ and the twelve-piece Atlanta Symphony Brass Ensemble. I understand four records will be issued; two of organ and brass and two of solo organ. Organist Richard Morris seemed to have an excellent command of the material, and the performance of the Symphony Brass, with virtuoso first trumpeter John Head, was well nigh perfect. Much of the music was overflowing with pomp and splendor and seemed to blend well with the visual surroundings. Sitting there amidst the stained glass windows while hearing the sonorous bass drum whacks and soaring trumpets in the opening bars of Aaron Copland's Fanfare for the Common Man was an exhilarating experience. Another impressive selection, and one which may well turn out to be the showpiece of the entire series, was an obscure and heretofore unrecorded piece from the music for the court of Queen Elizabeth.

Regarding recording techniques, it was good to see the use of a simple, three-point omnidirectional pickup. The microphones used were special, small-diaphragm instrumentation types. Apparently these have much better off-axis high-frequency response than typical omnidirectional condensers (Schoeps, AKG, et al), and no doubt have much to do with the superb sound we have come to expect from the Crystal Clear label. (In microphone design, the capsule diameter is a trade-off between noise and off-axis response. Smaller diameters are generally noisier, but may be used for applications in which this is not a serious concern. -- MR)

A word about the acoustics. The Cathedral of Christ the King was constructed in 1938 and the Ruffatti organ installed in late 1972. The building is of Gothic style, with high ceilings, stone walls, and terrazzo floors. Distribution of bass energy in the vertical plane is somewhat uneven -- it is strongest near the floor and grows progressively weaker with increasing elevation. When the church is empty there is a fair amount of reverberation (decay time: 3 seconds) and a tendency toward upper-mid-range and high-end brightness. These two qualities combine to produce a majestic, spacious sound, albeit one which is very loud and rather bigger than life. Those who crave "pin-point imaging" may not care for such an overwhelming acoustic setting, but I found it ideal for the type of music presented. Let us hope that Crystal Clear has captured it in full measure.

It will be interesting to see how the discs (especially the ones with brass) compare with the real thing. If they're even close, they will have the potential of representing direct-to-disc recording at its finest hour. -- Nick Lombardi (Georgia)

Head Configuration for Auto-Reverse Cassette Recorders

Everybody has his own favorite choice of features he wants to see in a cassette recorder. Let me describe what I want: a timer-operable recorder with auto-reverse recording/playback and tape monitor. The present 45-or-60-minutes-to-a-side limit is just too short, but the C-180's are not hi-fi. And I want three heads for best performance.

Problem: The most convenient arrangement uses an erase head and a dual-head (separate record and playback gaps) assembly. There is space for a second erase head (mirror symmetric with the first erase head), but what about the record and playback heads, each with its optimized headgap? I suggest leaving them the way they are. Therefore, on recording in the forward direction, the deck operates as before, with tape monitor. In the reverse direction, continue using the record head, which, now being downstream from the playback head, makes it impossible to monitor. That's fine: if I use the auto-reverse record feature, I am not looking for the tape monitor -- I want convenience while still enjoying high-quality recording. If I want to monitor, I will flip the cassette over. And if it's on timer operation, who's going to monitor anyway? Reverse playback presents no problems. Please do not give me rotary heads; I do not trust them, and the cost must be high.

This proposed arrangement simply requires an additional erase head, some extra switching, and motion sensing. Because dual-capstan drives can be made to reverse well without difficulty, the additional cost should not be too high.

You manufacturers out there can use this suggestion without charge; I promise to buy the machine when it appears. -- Cary Lu (Massachusetts)

Information for the Masses

For those who may not have seen the "How to Shop for Stereo" article in the May 14, 1979 issue of Business Week, here is some of what you missed:

"Don't forget that turning the volume control all the way up and feeding too much power into small speakers can produce distortion and might even harm the delicate cones."

"Most likely you will want a standard 'moving magnet' cartridge that the store or salesclerk will mount in the tone arm, a delicate task requiring careful adjustment of tiny screws."

"On the better machines, a special electronic circuit known as Dolby can be switched on to remove a semiaudible hissing sound made by the tape as it glides over the recording and playback heads."

"A frequency response of 35 Hz to 18,000 Hz is considered good, one of 100 Hz to 15,000 Hz is acceptable."

"Do not let an overly aggressive salesclerk load you up with electronic marvels that are expensive and are important only to the most dedicated audiophiles. Among them are 'reverb,' or time-delay gadgets ... (and) ... new 'digital' record albums made by computer."

-- Nick Lombardi (Georgia)

In the Literature

Audio, May 1979

*The Forum (p. 6): On the proposal to eliminate clear-channel AM stations.
*Behind the Scenes (p. 8): Bert Whyte's notes on the Winter CES: Carver's preamp, fancy turntables and cartridges, etc.
*Audio ETC (p. 18): On jargon and careless rhetoric.
*Tape Guide (p. 26): Q & A.
*Obituary (p. 28): On Clarence Moore, founder of Crown.
*Letters (p. 34): Including an objection to the high noise level in airplanes. Wear earplugs.
*Line Surge and Hash Protection (p. 38): Filters and other notions from Frank Stifter.
*Catastrophe Theory and its Effect on Audio, Part 3 (p. 42): Just what you wanted to know about trimodal stability and the butterfly factor; it all reveals that emotional states play a large role in product evaluation. Surprise!
*Build a Headphone Amp (p. 56): An unnecessarily powerful amp for home use.
*Equipment Profiles (p. 60): Stanton Permostat record treatment (it works impressively, but it costs about 80 cents per treated disc). Nikko Gamma 1 tuner (quite good except for an audible de-emphasis error). Nagatronics HV9100 ribbon cartridge (superb performance, output level extremely low). Audio Technology 510 stereo LED level display (accurate, intelligently designed, convenient, can be switched to read either line level or power amp output). Advent 500 SoundSpace time-delay system (a rave review, "the best and most realistic" time-delay he has heard to date). Visonik 8200 direct-drive turntable (good, convenient).
*European Letter (p. 80): Notes from across the sea.
*Top of the Pile (p. 90): On the cutting of Mobile Fidelity superdiscs, and reviews of other audiophile discs and tapes.

The Audio Amateur, 1979 No. 2

*Advent MPR-1 Microphone Preamplifier (p. 5): When Advent stopped making this excellent preamp, TAA scooped up Advent's remaining stock of parts for it, including the input transformer, and now offers the unit as a $42 kit. But beware: the metal chassis is not included in the kit. Many audiophiles are handy with a soldering iron, but lack metalworking facilities and cannot punch large holes for XLR or phone sockets in a steel box (required for magnetic shielding), so this is a crucial omission. One practical solution, requiring only the use of a 1/4" drill, is to use toggle switches instead of slide switches (thus requiring only round holes) and not install input jacks; instead, solder cables to the input transformers, lead them out through holes, and attach female XLR sockets (Switchcraft A3F) to the cables.
*A Comparison of Preamp Construction Projects (p. 11): Notes on the Williamson 20/20, the SWTPC Super Op Amp, a homebrew employing the LM381A IC, and the Marshall Leach.
*Direct Cut Myths and Problems (p. 16): On Umbrella's direct-discs and Teldec's pressing operations.
*Suppression of Surface Noise (p. 22): The full operating theory of today's tick-and-pop suppressors, as described by Williamson in 1953!
*Soldering (p. 24): Some basic data and good advice.
*Audio Aids (p. 30): An A/B switcher, improving the SP-3a-1, de-buzzing the Hafler 101, calibrating the Kenwood 1030, etc.
*Test Reports (p. 40): On the Heath audio load and sweep generator.
*Letters (p. 48): Including a lengthy report on the audibility of very small RIAA deviations, from Stan Lipshitz et al.

Audio Engineering Society Journal, March 1979

*FM Distortion in Phonographs (p. 121): Analysis of the frequency modulation of a steady tone on a test disc reveals the vertical and lateral tracking angles, the effective tip radius, and the needle-drag distortion of the pickup cartridge. One conclusion: "measured vertical angles of present-day pickups usually fall in the range of 20 to 36 degrees, with a mean of 29 degrees." Are our cartridges really 10 degrees away from the presumed 20-degree standard of modern cutters? Stay tuned to this channel.
*A Low-Frequency Horn of Small Dimensions (p. 141): A reprint of Paul Klipsch's original 1941

paper on the design of the Klipschorn woofer.

*CCIR/ARM (p. 149): Dolby's recommendation of CCIR weighting with an average-responding meter for noise measurements.

*The Stereo-180 Microphone System (p. 158): On the discovery that a "binaural" sense of image breadth and spatial reproduction can be achieved through loudspeakers if the recording is made using a pair of hypercardioid mikes angled at 135 degrees and spaced 46 mm apart.

*Competition and Technical Innovation (p. 178): From the U.S. Department of Justice, an essay on the legal antitrust implications of the making of standards (as in the case of standardizing digital recording formats).

Audio Times, April 1, 1979

News items: Connoisseur Society is releasing an extensive series of pre-recorded cassettes, dubbed from 15 ips masters onto BASF super-chrome tape, to sell at $11 list. TDK is developing a 5-hour VHS videocassette. Now that Phase Linear is owned by Pioneer, it is going to expand its product line broadly, including a $700 turntable and a line of speakers.

Audio Times, May 1, 1979

The large array of new Marantz products to be introduced at the summer CES will include a TV audio tuner ($185) and a slew of two-speed cassette decks. (It is understood that American manufacturers have a loophole in their Philips cassette license, which permitted B.I.C. to introduce the two-speed cassette deck, while the Japanese license lacks that loophole; Marantz products, except for speakers, are made in Japan, but Marantz is technically an American company, being owned by the Tushinsky family of California.) The IHF is becoming affiliated with the EIA after years of warring. Ben Bauer has died at 65; his long career with Shure and CBS Labs included over 100 patents, notably the cardioid microphone (1938) and the SQ quadraphonic recording system. Andy Petite and Frank Reed, formerly of Advent, have started a new speaker company in East Boston to be called "Boston Acoustics." EMI tape from England will be distributed here by Empire. TDK will bring out its metal-particle cassettes in June, instead of waiting until fall as earlier planned. Other tape makers are cautious about metal's prospects. Advent's new president, William Anderson (formerly of Frigidaire, RCA, and Sharp) says that $2500 will remain the bottom price for high-quality projection TV for years to come; no major price cuts are conceivable.

Gramophone (England), April 1979

*Commentary (p. 1791): A look at EMI's facilities for transcribing ancient 78s for re-release. Plus Harry Maynard's report on the winter CES display of new products, and a look at Bib, the accessories maker.

*Reviews (p. 1799): JVC MC-2e moving-coil cartridge, QL-47 direct-drive turntable, and P-3030 preamp (preamp superb, with flexible cartridge loading adjustments and a quiet head amp; turntable okay, but arm is rather massive, yielding a 6 Hz resonance with the JVC cartridge; cartridge excellent). Allison Three loudspeaker (excellent in terms of bass and treble range, power-handling, dispersion, low distortion, but midrange transparency just slightly lacking). Technics 9011 preamp, 9021 power amp, 9031 tuner (tuner outstanding, amp and preamp good, except for poorly chosen high and low filters in the preamp).

*Philips Compact Disc (p. 1810): Details on Philips' new digital disc format, a 4 1/2-inch disc whose player is anticipated to sell for under $300. Laser playback is used, and playing time is one hour.

HiFi Stereophonie (Germany), May 1979

Several articles devoted to discussing the image of the sound engineering profession, among them:

*Inner Views of the Medium (p. 581): The mutual effects of musical and technical aspects of recording.

*Studies in Berlin (p. 592): About the Department of Music Recording at the College of Arts in Berlin.

*Possibilities and Limits of the Recording Techniques in the Popular Music (p. 606).

*Alban Berg's Lulu in Paris (p. 612).

*Record Awards 1979 of the Deutsche Phono-Akademie (p. 614).
*Test Reports (p. 646): The Mini-Series of Mitsubishi, Sony, Technics, Toshiba, and Uher (the miniaturization does not entail quality impairment; according to the reviewer, the real future of minis is in the tuner/preamplifier units). The Revox B77 Dolby and the ASC AS-6002/38 tape recorders (the Revox has brought no important improvements compared to the A77; the ASC is ideal for high performance home use). (The B77 has substantially more headroom in the record electronics than does a stock A77. -- MR)
*The Recording Level Problems (p. 674): How to use VU or peak level meters considering the musical contents.
*Level Setting Using a Dual Tone-Burst (p. 678).

High Fidelity, May 1979

*Equipment Reports (p. 29): H. H. Scott 480A integrated amp (flexible, well-chosen filters, conservative specs). Hitachi D-7500 three-head cassette deck with Hall-effect play head (elevated hum and poorly-written manual, otherwise good). Onkyo TX-4500/II receiver (has overly aggressive tone controls and filters, lacks tape dubbing, otherwise fine). Ultralinear 228 speakers (mediocre). Akai Pro-1000 open-reel deck (an unusually flexible semi-pro machine, performance fine except for bass rolloff). Uni-Sync 50 power amp (excellent).
*Too Hot to Handle (p. 48): Odd questions.
*Whither Good Radio? (p. 53): Classical music broadcasting, past and future.
*Car Stereo (p. 63): A cursory survey of some high-priced goodies.
*The Untamed Philly (p. 67): Recording sites of the Philadelphia orchestra.

Modern Recording, May 1979

*Interfacing Auxiliary Equipment (p. 42): A basic tutorial on the uses of the many in/out patch points on a studio recording console.
*Jazzing Up PCM (p. 52): How PCM digital recording works, details on Denon's PCM recorder, and a series of jazz sessions.
*Lab Reports (p. 76): Denon 750 cassette recorder (superb performance, but at $1400 its omissions are inexcusable -- no tape monitoring, no Dolby rec cal controls, low headphone output, lousy manual). Spectra Sound 1000B graphic equalizer (excellent, but expensive). Hitachi HMA 8300 "class G" power amp (excellent). Neutrik AD4 analog delay unit (adequate for PA systems).

Boston Phoenix, May 15, 1979

The Sound Ideas supplement, about 75 pages long, includes numerous articles by and about BAS members -- on FM tuner design, audiophile discs, the Sound Concepts Concert Machine, one-third-octave spectra of noise levels and speaker response in a car, the high current demands placed on amplifiers by low-impedance loudspeakers, John Allen's Esplanade stereo system, etc. To order, send $1.00 to the back issues department of the Phoenix, 100 Massachusetts Avenue, Boston, MA 02115.

Popular Electronics, May 1979

*Stereo Scene (p. 21): Notes on some of the new products and trends seen at the winter CES.
*A Step Beyond Stereo (p. 25): A report on preliminary tests and listening sessions with a prototype of the new Carver 4000 preamp with its "sonic hologram" imaging circuit, autocorrelator noise reduction, etc. The hologram imaging turns out to be strongly dependent on the speakers and room used, and listener positioning is critical, but at its best the effect is dramatically convincing.
*Audio Reports (p. 39): Eumig CCS cassette deck (an unorthodox Austrian design, expensive, mostly superb performance, but poor headroom and thus mediocre S/N). Pioneer TVX-9500 TV audio tuner (mediocre sensitivity and excessively high muting threshold, but when fed strong signals from a roof antenna, performance is outstandingly good, better than specified).
*Test Report (p. 83): Sabtronics 8100 frequency counter kit (a rave review -- the unit is far better than its specs suggest).

Radio Electronics, May 1979

*Amplifier Buyers Guide (p. 47): The introductory article is good, but the compilation of specs is filled with misinformation (because of some manufacturers' listings being in accordance with the new IHF standards and others' not), and so is not to be trusted for comparisons.

The Real Paper, May 19, 1979

The supplement on stereo includes an article about the BAS.

Stereo Review, May 1979

*Audio News (p. 30): On proposals to alter station allocations to make room for more AM and FM broadcasters in the U.S., and the status of RFI rules.
*Audio Q&A (p. 36): Good advice on turntable maintenance, why dealers bad-mouth certain product lines, and the dismal prospects for lo-fi AM stereo.
*Audio Basics (p. 40): On devices for noise reduction and other useful tricks.
*Tape Talk (p. 42): Peak-reading displays are better than slow meters.
*Technical Talk (p. 45): Julian Hirsch fires his cannons at the arrogance and strained hyperbole of some subjective reviewers.
*Test Reports (p. 46): Carver C-4000 preamp (Hirsch engages in some hyperbole of his own in this rave review of a hand-made prototype of a product that does not exist in production form yet, followed by a special report by Larry Klein on assorted listening sessions). * Fisher CR-4025 remote-control cassette deck (good for its price, and convenient). SAE 180 parametric equalizer (clean, quiet, surprisingly flexible for only a two-band unit). SME Series III tonearm (laborious to install, but worth it; extremely low mass, very effective damping products outstandingly stable tracking). Sherwood 7650 receiver (fine performance, very conservatively rated).
*Las Vegas Audio Show (p. 79): An exhaustive report on the mountain of new products shown at the winter Consumer Electronics Show. -- Peter W. Mitchell and Jiri Burdych

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*Whatever Bob Carver's status as a product designer may be, he surely wins the prize as the best PR man in audio today. In a survey of the spring's magazines, Carver got a larger total volume of press coverage, and a larger amount of enthusiastic praise, for two non-existent products than most manufacturers get for several years' worth of designs. All this praise has been generated for a "magnetic" amplifier that is not magnetic (more accurately described as an efficient power supply that is amplitude-modulated by the audio signal, an extreme form of "class H" operation) and for a "hologram" imaging circuit that is based on an old idea described in detail by M. R. Schroeder of Bell Labs fifteen years ago and previously implemented as a commercial product by JVC in their "biphonic processor." Eventually, when Carver's products are fully designed and go into production, we shall see how well their performance justifies all the advance hype they have received. (I am not implying anything negative about the products, only marveling at his skill at generating publicity. Given Schroeder's experiments and the performance I have heard from the biphonic processor, it seems clear that the hologram generator can indeed produce impressive imaging when the recording, speaker setup, and listener positioning are carefully chosen.) Incidentally, since the basic operating principles of the hologram generator are well known and the required circuitry is not very elaborate, it may be possible to develop a homebrew kit in the $50 to $100 range. We're looking into it. -- PWM

Carver claims that his circuit is not a straight adaptation of Schroeder's work, but involves refinements that allow it to work well in live rooms with ordinary program material. If true, this certainly would make it different from JVC's device, which is intended for use with binaural recordings. Whether we can develop a kit to do the same thing remains to be seen. Meanwhile, the whole thing bears watching -- but perhaps nothing more until the smoke begins to clear. See Al Foster's article elsewhere in this issue. -- MR

April BAS Meeting

Business Meeting

Frank Farlow called the meeting to order with the announcement that both Jim and Joyce Brinton were ill and unable to attend. As there was no official business, an extended open forum was held while awaiting the appearance of amplifier and speakers for the meeting feature.

Several previously announced items were about to expire: Dick Glidewell's offer of the Draco/ Sherwood Micro-CPU tuner and the FCC Notice of Inquiry into RFI were only open until the end of April.

Dave Weinberg asked for those interested in a possible group purchase of capacitive-discharge ignitions to contact him. Frank Farlow announced the continuing availability of order slips for Audiomart, a monthly used-equipment flyer. Peter Mitchell announced the upcoming Shoptalk Auction, scheduled for the 12th and 19th of May on WBUR-FM. Various audio manufacturers have donated products for the auction. Also, Peter said that the latest Audio Amateur has plans (and a kit -- $42) for the Advent microphone preamp; unfortunately, the kit does not include a steel box, which is essential for shielding. Also, Tom Holman said that the supplied board is screened for a TIS97 transistor, but the 2SC1345E (Hitachi) transistor supplied does not have the same basing. Be careful.

Lebow Labs does carry the Integrex Dolby™ decoder, but has no information on it. (The address for Integrex is P.O. Box 747, Havertown, PA.) Lebow Labs will give BAS members a $5 discount, on the two-channel version only.

Brad Meyer reported that the "flood of test-equipment questionnaires" had abated, and the information would be digested within a couple of weeks. The overall response was favorable, including those out-of-state members who responded. Many people want the BAS to have an Ivie spectrum analyzer. There have been approximately 140 replies received, or more than 10% of the membership.

Peter Mitchell said that the BAS had tried to get Robert Carver to come to a meeting, but that he is too busy just now -- "possibly next fall." As Carver has refused outside financing for his new company, the preamp will be in short supply for a time. However, we can expect that, in about two years, the Sonic Hologram will be released separately. The Hologram is apparently based in part on a circuit published some time ago in the Audio Amateur, and is similar in effect to the JVC binaural processor.

Andy Petite was asked and refused to comment about his new speaker. His company, Boston Acoustics, located in East Boston, plans to produce a three-way acoustic-suspension loudspeaker system, to retail "around $400 each," starting this summer. The midrange and tweeter will be purchased from vendors; Boston Acoustics will make their own woofers.

Al Foster gave a preliminary report on the Turntable Test Clinic. Results will be published over two issues of the Speaker, because there is so much data. The most significant finding is that over half the participants have overly compliant cartridges in excessively massive arms. As at least ten people put in upwards of eighteen hours each, there may be another clinic, but not soon.

Cary Lu gave an extensive discussion of the pending FCC proposals on AM and FM broadcasting intended to increase the number and availability of channels. This was also covered in Stereo Review magazine. There is not yet an open Notice of Inquiry. Briefly, the FCC proposes to reduce AM channel separation from 10 kHz to 9 kHz, extend the band upward to 1800 or 1860 kHz, and eliminate the designated "clear channel" stations; to reduce FM channels from 200 kHz to 150 kHz or even 100 kHz spacing and reassign station frequencies according to computer models, which should minimize interference. In both AM and FM, present station allocations would change.

One major effect would be the obsolescence of present digital-frequency-synthesis tuners. Another might be increased adjacent and alternate channel cross-modulation in the USA only, because most of the rest of the world has comparatively uncrowded broadcasting. Basel, Switzerland, has only nine stations, for example, and Japan (where many tuners are built) has only two or three.

Following a short break, Frank Farlow introduced Bob Berkovitz, Director of Research at Acoustic Research, and Dave McIntosh, formerly with AR and now an independent consultant.

Meeting Feature

After a short delay, caused by trouble with the analog PA system, Bob Berkovitz unrolled a scroll (which he denied obtaining while mountain climbing in Framingham) and explained that he would first cover the basic concepts of digital technology and play for us some samples of tapes made of the Boston Symphony and the MIT Symphony. Dave McIntosh would then demonstrate some techniques in digital measurement of loudspeaker parameters, using the Apple II microcomputer.

Bob started with a description of the "Banker's Problem": how can we store and recall numerical data? To answer this, we need to understand the physics of tape recording. Tape is composed of many small magnetic particles that do not move, but whose polar orientation can be changed. For a zero average field, statistically, about half the fields point each way. This is never exactly half, however, and the variance from place to place on the tape is the residual noise level. Ray Dolby and others have made fortunes trying to reduce the influence of this noise without being able to reduce the noise itself.

Assuming a perfectly linear system, setting 1 μA (microampere) equal to $1, we could record up to $\pm$\$350.00 maximum, with an rms error of $\pm$6.25 cents, or a signal-to-noise ratio of 75 dB (one part in 5600), on a good recorder. For an account that averages around zero, the error can equal the signal.

If, instead of relying on level, we insist that the bank account be either full or empty, then we can accurately separate magnetically saturated places on the tape from ones that are approximately zero, or saturated the other way. By giving everyone sixteen accounts, with binary place value (2^0, 2^1, 2^2 ... 2^{15}), we virtually eliminate the chance of error and also increase our S/N to over 96 dB, or one part in 65, 535. For example, \$5000 would equal the sum of bits -- or accounts -- 13, 10, 9, 8 and 4. The maximum error is half of the smallest account, or bit: bit 1, called the LSB (least significant bit) equals one, or one times 2^0.

We can accurately describe a musical waveform by a list of 16-bit numbers, each of which is a measure of the signal amplitude at some known (relative) time. By measuring the waveform repeatedly and recording the number, we have a description that enables us to reconstruct that sequence with a maximum error of $\pm$1/2 LSB. That's not just noise -- that includes distortion, too.

An additional advantage over recording the waveform itself (analog recording) is that, because the time relationship of samples is known, we can reconstruct that timing to arbitrary accuracy on playback, thus eliminating wow, flutter, amplitude or frequency modulation, and other time-base distortions that might arise because of an unstable tape transport.

Bob then played three examples of digital recordings. Equipment used was a Sony PCM-1 digital converter (11-bit floating-point with two bits ranging and extra error-detection bits), JVC U-matic video recorder model CR-4400U, Dunlap-Clarke 500 amplifier, and a pair of AR 90 loudspeakers (with all driver controls set flat).

First a test tone was played to demonstrate the complete absence of wow or flutter or apparent dropouts. Then a short selection of the Bartok Concerto for Orchestra from a Boston Symphony Orchestra concert last February was played. This included a few minutes of a rehearsal, with Sir Georg Solti talking to the orchestra. The hiss, clearly audible when the orchestra paused, was from the Transcription Trust's Neve mixing console -- analog, of course. (I have to wonder why they spent \$60, 000 on a board with that much noise. -- MR)

The third selection was quieter in background, though noisier music -- the MIT Symphony in a recording session with John Newton. Here it was clear that the digital tape was capable of a greater dynamic range than the ambient noise of the recording environment permitted, because when the faders were pulled down, the slight noise went away.

Questions and discussion from the floor followed. The obvious advantage of digital recording is the complete lack of deterioration through copying. Editing is difficult, but existing video techniques using computers can be employed. Because of the incredible S/N capability, we are finding that 20 dB SPL microphone noise is really noisy.

To avoid aliasing error, very sharp anti-aliasing band-limiting filters are used -- the amplitude response is within +0, -1/2 dB from 2 Hz to 20 kHz, but there is phase shift. Scott Kent pointed out, however, that his phase-corrected harpsichord recording needed over 1000 degrees of shift correction at 20 kHz, and no one claims to have heard it.

Polygram and others have recently conducted studies that indicate that a 15 kHz limit satisfied any listener they could find on any program they had; in The Audio Amateur, Stanley Lipshitz claims that 0.2 dB change in response between 15 kHz and 20 kHz can be audible, however. Obviously, there are questions still unresolved.

After another short break, Dave McIntosh and Bob Berkovitz demonstrated the use of an Apple II microcomputer to measure the impulse response and spectral decay of loudspeakers. The Apple was chosen because of its excellent graphics package and because it is more like the minicomputer AR uses in the laboratory than other microcomputers seem to be.

Dave started with the statement that the frequency response in amplitude and phase is merely the Fourier transform of the impulse response measured as a time function. By exciting the speaker with a narrow pulse (in this case, a square pulse of 10 μsec duration) and recording the output for 5 milliseconds from the start of the pulse, then performing the operation called the Fast Fourier Transform (FFT) on some portion of that interval, we get a spectral analysis of the speaker's response to the impulse.

If we look only at some subset of the interval recorded, say one msec, and perform an FFT on that, we get the spectral content of that portion. By moving our "window" along in time, looking at successive portions of the whole 5 milliseconds following the initial response to the impulse, we can see how the spectral output of the loudspeaker decays with time.

There are several other tricks we can perform. We can compare the behavior of any microphone to that of any other under identical conditions, and calibrate the differences in amplitude and phase response. This way we can effectively convert an $11 microphone into the equivalent of an expensive calibration microphone.

Additionally, because room reflections are incoherent, but the direct sound from the loudspeaker is repetitive (for repeated inputs), by using a series of impulses and averaging the responses we can suppress the effect of far reflections ("far" means greater than the microphone-loudspeaker distance). The average of sixteen impulses suppresses room effects 12 to 15 dB. This makes any environment more like an anechoic chamber than it would otherwise appear to the computer.

Tests were run on several loudspeakers, amid much discussion of the results. Bob and Dave both claimed that, so long as most of the energy that is going to emerge from a speaker does so in a relatively short time, the time order of emergence of the frequency components doesn't seem to matter much. A difference between two speakers in a stereo pair is audible, however, presumably as a smearing of the image.

We are talking about times shorter than the integration time of the human ear, of course, and at low frequencies this can be longer than 200 milliseconds (one-fifth of a second).

Although moving the microphone (relative to the speakers) only a few inches produced (sometimes) dramatic changes in the measurements, some results obtained (as a demonstration of the techniques involved only) were:

Dahlquist DQ-10: ± 5 dB 230 Hz to 21 kHz on axis. (This provoked the tongue-in-cheek comment, "That's a good speaker!" from Dave McIntosh.)

DQ-10, 10 degrees off axis: approximately 20 dB hole in response around 3 kHz.

The Avid 330A spectral decay shows a fairly prolonged resonance at about 2500 Hz. The tweeter cuts off very sharply when the input stops, however.

Finally, the ADS 200A minispeaker showed an overall slope of 6 dB from 230 Hz to 20 kHz, and appeared not to change much over several angles of measurement.

Because of the excellent graphics capability of the Apple II, these displays were of good quality even on the Advent 1000 Videobeam used for the demonstration.

A very long evening ended with the comment that the software used is proprietary to AR, but that there are FFT programs in BASIC that will do some of the same things, but take forty times as long (about four minutes per plot). -- Mark Fishman

A Publication of the BAS

Carver's Sonic Hologram: How It Might Work

Alvin Foster

If you are like most audiophiles, you are eagerly awaiting a chance to audition Carver's new preamp, or at least some of its auxiliary functions, i. e., the Sonic Hologram, the time delay system, the auto-correlator, etc. But if you are handy with a soldering iron, there is no need to wait for the Sonic Hologram.

The operating principle of the Sonic Hologram is not new. Shroeder, working at Bell Labs, observed the same thing years ago with the aid of a large computer, and two years ago JVC demonstrated at the New York AES Convention a device that did the same thing but required four speakers.

How It Works

The system works on the same principle that makes binaural recordings more realistic than stereo. Essentially, the headphones (required for binaural listening) prevent the right channel sound from reaching the left ear. The result of this isolation is full left-to-right imaging and depth, i. e., front-to-back relationships. Binaural reproduction requires special recording procedures. Only two microphones can be used, and they must be placed on a dummy head and located where the ears would normally be positioned. The French ORTF system will also produce nearly identical results. To realize the full benefits of the Sonic Hologram, recordings that utilize binaural recording techniques most closely will work best; however, if the reviews are correct, most recordings will benefit to a lesser extent from its application.

Remember, headphones are ideal for reproducing binaural recordings because they prevent the left or right channel output from reaching the opposite ear; perfect isolation. You can obtain very similar isolation by building a wall down the center of your room, between the speakers. To provide this same type of isolation, Carver has designed some exotic circuitry -- circuitry that would have been too expensive for a consumer product ten years ago. To obtain the desired channel isolation, he probably uses a time-delay bucket-brigade IC to build a "cross-feeding cancellation" system. The circuitry consists of the IC chip, an attenuation network, equalization, and a phase inverter. By simultaneously feeding the signal going to the right channel to the cross-feeding cancellation system, the left ear is permitted to hear the right speaker hardly at all. This situation is reminiscent of headphones, albeit less perfect. To accomplish this muting effect for your left ear, the output of the cross-feeding cancellation system is fed to the left speaker.

Because the left ear is further from the right speaker and the head lies between the two ears, some frequency response alteration and attenuation of the right speaker's output will occur before it reaches the left ear. The cross-feeding cancellation system must adjust for the resulting frequency response and level to obtain the desired cancellation. The system works because inverting a signal and mixing it with the original causes cancellation. The delay circuit is added to the cross-feeding cancellation system because the information from the right speaker will take slightly longer to reach the left ear than the right ear. To provide the desired cancellation for your left ear, the inverted signal and the right-channel information must arrive at your left ear at the same time, properly attenuated and equalized. To prevent the speaker output of the cross-feeding cancellation system from reaching your right ear, the right channel cross-feeding cancellation system must go into operation to render the left speaker's muting system inaudible. This flip-flopping goes on indefinitely, or until the signal is below audibility. The entire process, before a signal is effectively muted, lasts but a few hundredths of a second.

Timing is important. If the two signals do not meet at the same time, little or no cancellation will result. That is why your head must be confined to a relatively small area; any movement out of the "listening zone" will cause the two signals to arrive out of synchronization and to become audible. Carver's preamp may include an adjustment to vary the size of the listening zone. Making the zone too wide will cause more diffuse imaging. Perfect adjustment would permit instruments to move in space from your far left shoulder to your extreme right, a 3-D, or "holographic, " effect.

What Recordings and Components Will Sound Best?

Binaural recordings played back over the system described above will sound best. Recordings made with two closely spaced microphones will also sound very good. Multiple-mike recordings will have the least chance of obtaining the maximum benefit of the Sonic Hologram. Front-firing loudspeakers designed to minimize external cabinet reflections (e. g., the AR9 and 90 and the KEF 105) will work better than bi-directional or multi-directional speakers. Cartridges that preserve accurate phasing between the channels (e. g., the Shure V15-IV and Acutexes) will also contribute to the 3-D effect.

An Explanation of Figures 1, 2, and 3

The accompanying charts were made by psychoacoustician Mark Davis, a Ph. D. candidate at MIT. The three charts were obtained by placing tiny microphones within his ears while he sat in an anechoic chamber. The frequency response and time-delay difference introduced by the size of his head, etc., were accurately traced on a chart recorder. The charts illustrate the four functions the cross-feeding cancellation system must perform: (1) introduce approximately 675 microseconds of delay; (2) invert the signal; (3) equalize the signal; and (4) attenuate the signal.

Figure 1 shows the time difference between the two ears when the signal source is placed 60° to the left of the listener. The initial interception of the all-frequency pulse is represented by the large rise in the top half of the chart. The right ear hears the same signal about 675 microseconds later, greatly attenuated. A pulse occurring directly in front of the listener and lying at an equal distance between the two ears would reach the two ears at the same time.

Chart 2 displays the frequency response differences experienced by the two ears when the same pulse is 60° to the left of the listener. Below 200 Hz there is very little difference in level or frequency response, but above 200 Hz the difference becomes more pronounced.

Chart 3 shows the frequency response difference between the two ears. As you can see, the rolloff is rather gradual and predictable from about 200 to 2000 Hz. The various dips and peaks are introduced by the ear lobes, shape of the head, distance, etc. To enhance the effect of the Sonic Hologram, a filter would not have to conform to the exact frequency response of Chart 3, but merely follow a gross approximation of the curve.

Ten years ago it would have required a large computer to adequately delay, invert, attenuate and equalize the signal. Now you can buy two bucket-brigade delay chips from Reticon for $13 each, add your own equalization and attenuation approximating these curves, invert the signal, and you will have a home-built sonic hologram.

If you do decide to roll your own, you can determine how wide you would like your listening zone merely by varying the delay around 675 microseconds. The listening zone will increase as the delay is lengthened (however the stage width will be reduced). The best method would be to measure your head, ear lobe to ear lobe, dial in the necessary delay, and then place your head forever in a locked vice. Throw away the key, and enjoy nirvana forever.

The Sonic Hologram is not a departure from reality or a gimic. It is, instead, an enhancement of realism, designed to remove some of the distortions introduced by stereo.

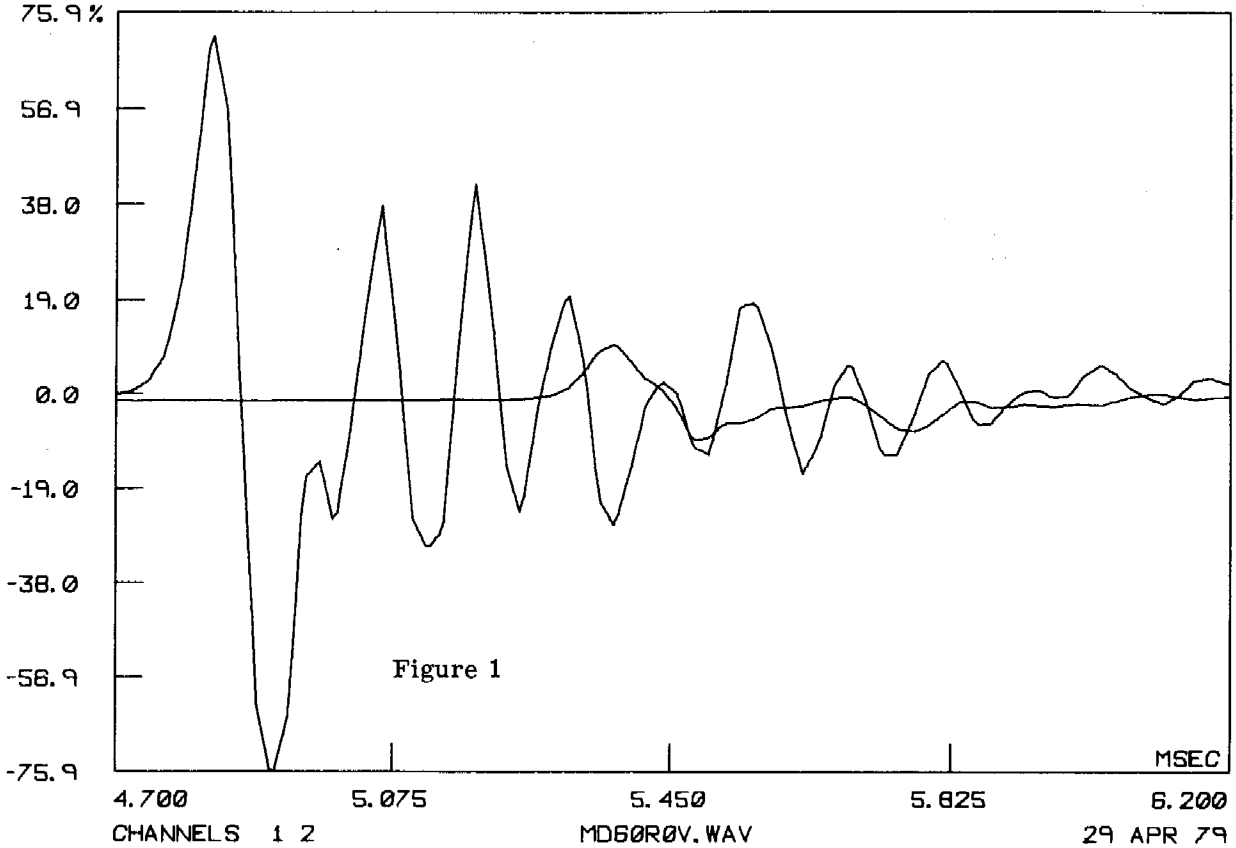

Figure 1

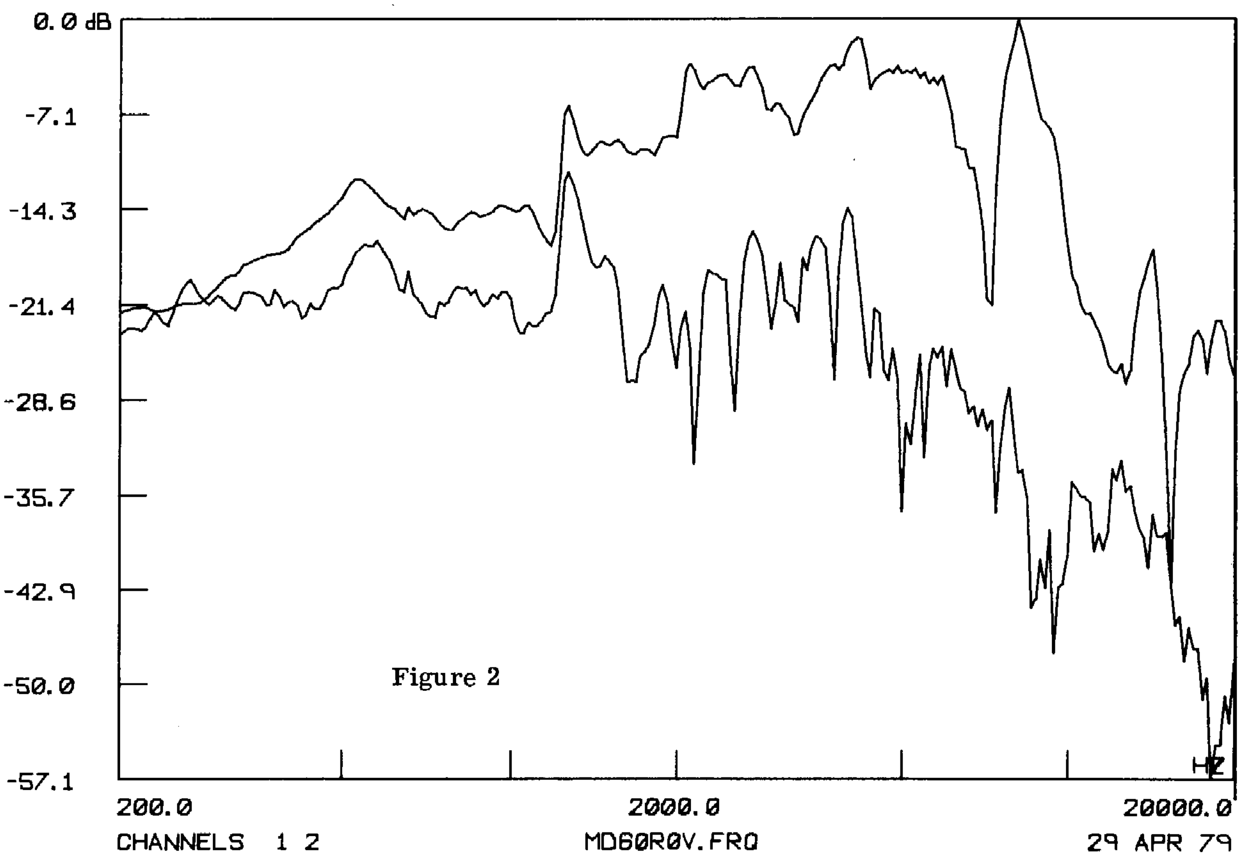

Figure 2

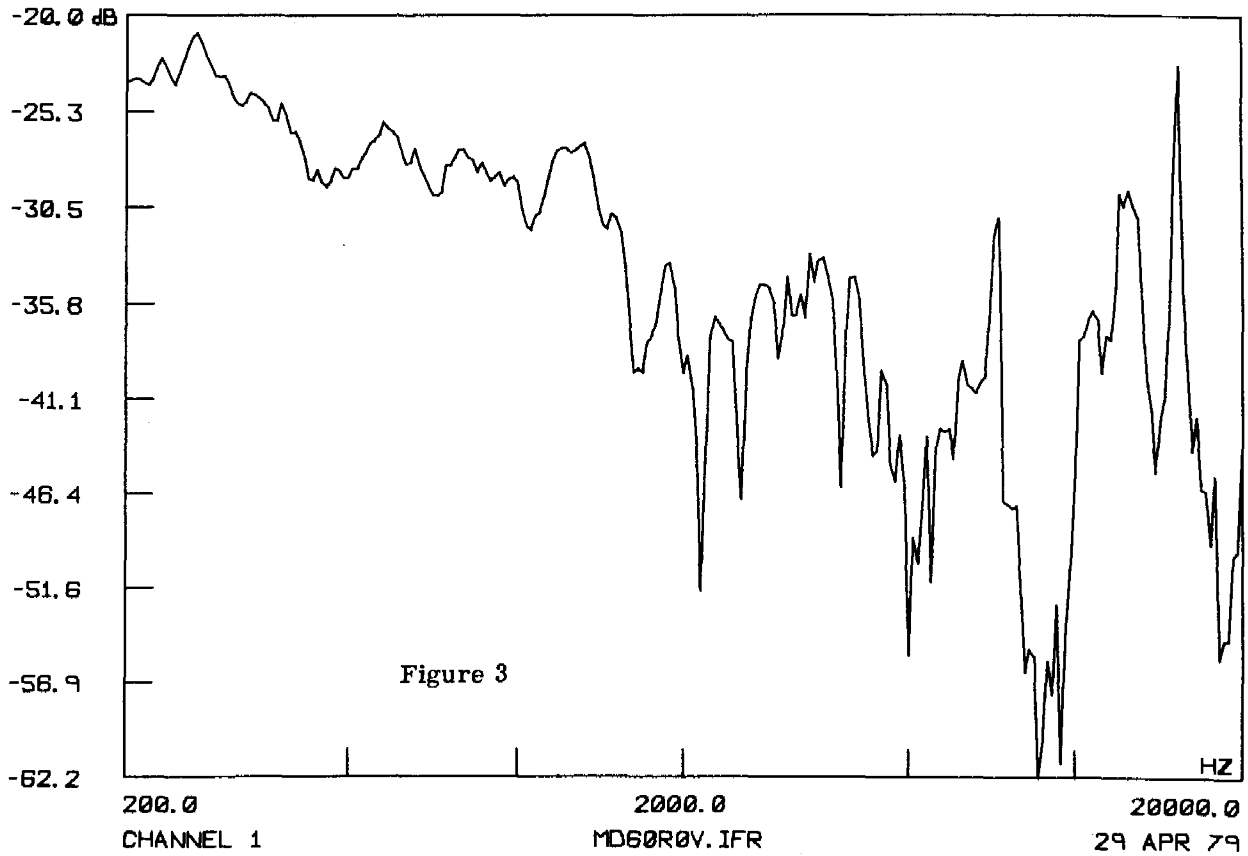

Figure 3

Update on the Sonic Hologram

Just after completing the above, I sent a copy to Bob Carver, president of Carver Corporation. He phoned shortly thereafter to compliment me on the parts of the article that are consistent with his approach and to provide additional insight into how his device actually works.

He explained that rolling your own is possible only if you plan to listen in an anechoic chamber with prerecorded, binaural material. A device based on the data I have supplied, which is consistent with the current research, will work, but only under non-reflective conditions. Extensive research was undertaken to get a device to work in real time and with existing stereo playback material. And, according to his patent attorney and three feet of research literature, the Sonic Hologram is the first device that works in a listening room and with available stereo recordings.

Amplitude-to-Phase Converter

The cancellation of the left or right signal is pretty much as I have described. However, an additional circuit was required to enable the unit to perform satisfactorily with all types of program material, including multi-miked recordings. He calls it the "amplitude-to-phase converter" system. Essentially, it reacts to the amplitude of a signal by reorganizing it into some complex phase relationship, after which, the signal is fed to the proper loudspeaker. Psychoacousticians have known for a long time that stereo localization is entirely amplitude related, i.e., it sounds louder coming from your left speaker, therefore, the performer is to your left. Carver uses this phenomenon to provide the required lateral perspective. The existing phase relationships on the record supply the information required for the front-to-back, or depth, relationships.

The Sonic Hologram will have a two-position switch. One position, the "normal mode," will allow the playback of regular stereo, multi-miked recordings, while the "theoretical" position allows the playback of recordings in which the natural phase relationships of the performance were recorded intact, i.e., Blumlein recordings, spaced omni recordings, etc. Both types of material will yield identical holographic results.

Head Shadow and Equalization

Carver was quick to point out that the equalization required to compensate for the frequency response alteration of the signal as it passes around the head to the opposite ear is quite complex, head-shadow equalization. Earlier I stated that only a gross approximation of the frequency response is required. This is not so. If the proper equalization is not added, the signal will sound as if it is coming from within your head, as in binaural headphone reproduction. Carver uses a twelve-pole filter below 1 kHz and a five-pole filter for higher frequencies.

Delay

I mention above that the required delay of around 675 microseconds can be accomplished with a chip readily available from Reticon. That may be, but Carver uses a bunch of wide-band FET amplifiers to get the necessary delay.

Carver says the quality of the holographic effect does vary with the characteristics of the room, but even under the worst conditions, lateral head position need only be confined to a space of about two feet, and front-to-back positioning only six feet.

Software

Record companies have already approached Carver to ascertain whether the Sonic Hologram information can be encoded directly onto a disc while it is being made. According to Carver, all records and tapes can be recorded in such a manner, and they would be completely compatible with existing stereo playback systems. He has already patented such a box and a ready-made package for the record companies. His unit is completely compatible with current recording techniques.

"Magnetic Amplifier"

A discussion of Carver Corporation products would not be complete without an update on the "magnetic amplifier." According to Carver, the unit has lost some weight, from 12 pounds as projected to 9.7 pounds. The price has risen to $349, but so has the amp's output, to 325 Watts per side.

When will the preamp and amplifier be available? According to Carver, all systems are set to go. He has sent fact sheets to dealers, and he expects to start shipping soon.