THE B.A.S. SPEAKER

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In This Issue

Our main feature is, at last, an index for Volume 4. We have both the traditional version, to appear in next month's issue, and the convenient John Gombos version, which appears here. Thanks to John for his work. We promise to get the Volume 5 index out in the first quarter of Volume 6.

Special attention is due this month's meeting summary. Not only does it cover presentations by Frank Van Alstine on his preamplifier and modifications to Dynaco equipment and by Peter Mitchell, Bob Berkowitz and Mark Davis on phase distortion, but also includes additional information the speakers did not have time to present at the meeting. You should read it even if you attended the meeting.

Other features include a subwoofer design by Jim Nichol, along with a cautionary note about building your own speakers. Dez Fretz reviews the Western Technologies 2S loudspeaker, Brad McCoy the Harman/Kardon ST-7 turntable, Bob Borden the Fons turntable and Formula 4 arm, and Dow Williams the Formula 4. We also have more on damping, this time from Brian Miller, who has paddle-damped his Black Widow, and on Disctracker.

Coming attractions include an article on modifying the Sony TC-377 tape deck, a review of the Dyna ST-150 both with and without Van Alstine's modifications, and a home rack-mount system.

Direct from Cleveland

We have sold the first batch of the Telarc/Advent Direct from Cleveland records but if you haven't ordered yours as yet, don't worry, a second batch is now here. The mail-order prices are:

United States		$8.75
Other countries, surface rate (including Canada & Mexico)		$9.45
Other countries, via air mail:	Europe & South America	$11.25
	Asia, Australia & Islands	$12.30
	Central America	$10.25

Membership dues are $14 per year (October 1 to September 30) or portion thereof. Dues include a one-year subscription to the BAS Speaker. (Note that almost the full amount of dues is allocated to production of the Speaker. The local activities of the BAS are strictly self-supporting.) For further information and application form, write to: The Boston Audio Society, P.O. Box 7, Kenmore Square Station, Boston, Mass. 02215.

For Sale

*Audio Research D-100, $750; Denon AU320 transformer, $75; Grace 707 tone arm, $100; Fidelity Research Mk. II, $75; GAS Sleeping Beauty, $140. All in excellent condition with all packing and shipping included. Write George Fulton, 554 Georgetown Lane, Beaumont, TX 77707.

*Dyna PAT-5, $125; Citation 15 tuner, $225. Tommy, (212) 255-1691, 10 p. m. to 11 p. m.

*Dayton-Wright XG-8 Mk. III speakers, factory rebuilt to current specifications, $1750; dbx 124, brand new, in box, $290; Technics SP-10 Mk. I, with tone arm, base and three detachable headshells, $345; two Dunlap-Clarke Model 10 preamp prototypes, $375 each; one Dunlap-Clarke Model 250 amplifier prototype, $399. Write DCE, 230 Calvary Street, Waltham, MA 02154.

*One Beyer M 101 N(C) dynamic omnidirectional microphone, $80 or best offer; one Beyer M 201 N(C) dynamic hypercardioid microphone, $100 or best offer. Microphones have standard 3-pin XLR connectors and are supplied with case, stand adapter, and cable. David Satz, (617) 492-2263.

*AGI 511 preamplifier; Accuphase T-100 AM-FM tuner; 1 pair of AR-7 loudspeakers; 1 pair of AR-MST-1 loudspeakers. Will sell very cheaply. Best offer accepted. Call (617) 687-6016, evenings and weekends.

*Your most important high fidelity component - - an oscilloscope: Heathkit model 10-102, 30 mV sensitivity, 5 MHz, single trace, recurrent sweep, similar to the current model 4541 but without triggering. Perfect working condition, $120. Dual-trace adapter for above or any other scope, electrically identical to the model ID-4101 Heathkit but in a package which matches the I0-102, $35. Dyna PAT-5, perfect, no modifications, wired from kit, $130. BSR FEW-III equalizer, $120. Headshells for Miracord 50H or similar, free. Need a second turntable or a gift? Lenco L-85, beautiful Swiss design, semi-automatic with no attachments to arm during play, with spare headshell, base, and dustcover, $100. Tired of audio? I also have a telescope. Harry, (617) 862-5500, x 5811 days or (617) 275-2171 evenings.

Wanted

*Pioneer or Kenwood audio scopes, Sony TC 651. Dick Glidewell, (617) 874-5740.

Errata

We have several errors to tend to. First, in the June issue we printed "Some Capsule Rock Recommendations" without credit to the author. We're very sorry for that, and if he will identify himself to us we will make amends. The second also appears in the June issue, in Scott Kent's "The Limits of Slew Rate, " into which we inserted "June" instead of "July" for the date of publication of A1 Foster's article on receivers. Finally, we made two mistakes in Foster's piece. This sentence is on page one: "The benefits are easily audible, particularly with speakers that are not damped at the lowest frequencies." "Audible" should be changed to "visible." Page eight contains the following sentence: "However, a design shortcoming of the phono section is its 4.5 dB rise at 100 kHz. " It should have been omitted altogether.

BAS Elections

BAS elections will be held at the September meeting. At this writing, the following people are running for the following offices: Jim Brinton for president, Al Foster for recording secretary, Frank Farlow for corresponding secretary, and Henry Belot or Peter Mitchell for treasurer. As you can see, this guarantees the first contested election in BAS history. Nominations will also be accepted from the floor. Out-of-state members can vote by sending a card before the September meeting. That will be on the 20th, so do it right away.

A Passive Radiator Subwoofer

After finding that my Speakerlab SK horn woofer had pathetic bass response below 100 Hz, I decided to attempt a subwoofer design, crossing over the SK at 80 Hz and letting a single subwoofer handle the deep bass. (I considered using an equalizer to boost the bass of the SK woofer but decided against it because of the need for increased amplifier power to drive the woofer at higher levels and because of the possibility of driving the woofers into nonlinearity.)

My first attempt failed miserably. I bought the PASSRAD passive radiator drivers from McGee Radio and installed them in a 1.6-cubic-foot box. The PASSRAD system consists of an eight-inch woofer and a ten-inch passive radiator. The McGee catalog ad claimed these drivers would go down to 33 Hz in the 1.6-cubic-foot box with less than "2 bd" (sic)- variation throughout its piston range. It looked good on paper, but when I built one I found it dropped off sharply below 90 Hz.

I then built three boxes having four-, six-, and eight-cubic-foot volumes and tried woofers of eight, ten, twelve, and fifteen inches, along with passive radiators of ten, twelve, and fifteen inches. I worked up to the eight-cubic-foot box, because the ever-increasing volume (with proper woofer and passive radiator) gave lower bass response.

The woofer that worked best in the final design was the CTX twelve-inch woofer 12W38C, which is available from McGee. The passive radiator was a CTX twelve-inch woofer (McGee model CTS-12ME6840, $12). It has only a 6.8-ounce magnet, but I was after the cone and suspension, so I cut out the spider, cut the speaker wires going to the voice coil and removed the voice coil by cutting off the apex of the cone just behind the dust cap. This leaves the cone floating on the suspension. You can also knock off the magnet with a hammer and chisel if you need a magnetic paper weight.

The combination that worked best used the above mentioned twelve-inch woofer and twelve-inch passive radiator in a box 2x4x1 feet to give the eight-cubic-foot volume. I realize this goes against the usual rules for box dimensions, but in doing my experiments I found it most convenient to use 3/4-inch plywood in standard 2x2 and 2x4 sizes to minimize woodwork. [JBL says that if the volume is sufficient, the dimensions are unimportant. -- Ed.] The effect of this box shape is reflected in the response curve, which is down 3 dB at 28 Hz.

In trying all these combinations of box volumes, woofers and passive radiators, I found that I obtained best results when the woofer and passive radiator were of equal areas. (This contradicts the results in Small's article on passive radiator systems; I have no explanation for this.) I also found that if a given combination of woofer and passive radiator in a small cabinet caused a peak in the low end, I could reduce the peak by adding weight to the passive radiator cone. However, if the response is good to start with, adding weight to the cone only introduces bumps and humps in the response curve.

I have noticed that McGee Radio has passive radiators in its new catalog. I intend to try them to see if I can make this system more cheaply without sacrificing performance. I'll report on the results. Also, I (and everyone else, for that matter) should include a standard compulsory warning that this is a ported woofer and that one should not even think of building it without making plans to include an infrasonic filter in his system to keep the record warps and tuner thumps far away from the woofer.

-- Jim Nichol (Massachusetts)

Many Negative Thoughts on Rolling Your Own Speakers

Having provided BAS members with data on speaker drivers currently on the market, I now propose (resting secure in my glass house) to throw rocks at the whole concept of building ones own speakers or even of building speakers from commercial speaker kits.

The first problem is quality control. During the BAS tour of the AR facility we were told of the calibrated tweeters in each driver testing chamber. Each tweeter is tested with a sine wave sweep and the results compared with a standard curve outlining upper and lower limits of acceptability. Failure rates range from 20% to 100%, and on one tweeter they are very happy if the failure rate is a mere 20%. As I recall, AR tests each woofer they manufacture. If the woofer fails quality control, they rip out the cone, voice coil, and suspension, and reconstruct the speaker for retest. If it doesn't pass quality control after three attempts, the basket and magnet assembly are scrapped.

Now, assuming a similar failure rate occurs with other speaker driver manufacturers (and Murphy's law guarantees that failures will occur), who is going to test all those drivers you buy from McGee Radio, Speakerlab and B & F Surplus? All three of these vendors merely distribute drivers, and I have yet to read of any of the driver distributors having quality control departments. True, some brag in their advertising about all the expensive test equipment they use in designing their speaker kits, but they don't brag in a similar manner about their quality control departments. For example, I had a bad experience in this respect when I ordered a CTS midrange from McGee. It arrived with RTV adhesive around the edge of the driver (it had been removed from a defective speaker) and someone had scribbled "BAD" on the box it came in. I returned the driver immediately with a scathing letter demanding satisfaction, which I eventually obtained.

My second point is the need for sound level measuring equipment to see if the crossover networks are actually doing their job, to verify that the relative levels of the drivers are set to achieve flat frequency response (you will probably have to do this if you build a speaker from an article in Popular Electronics, Audio Amateur, etc.), and to discover any gross dips or peaks in response (especially common in the bass).

To illustrate the need for this test equipment, allow me to relate the sad experience of a friend who was looking for a speaker system, cheap, and found a great deal at B & F Enterprises (an electronic and industrial surplus outlet). For only $15 he got an eight-inch woofer, a five-inch cone midrange and what looked like a typical CTS $3 dome tweeter. He wondered: "How can I go wrong? And how can I go wrong with their surplus or reject KLH cabinets and crossover capacitors for a few extra bucks?"

I measured the response of the midrange and tweeter and found that the five-inch midrange only went down to about 800 Hz, about 200 Hz below the low end cutoff of the tweeter. I talked him into getting a good midrange (the Phillips 5060/SQ8), which goes much lower, thus making crossover networks possible. There followed an afternoon of sabersawing holes for driver mounting, then winding lots of big crossover coils and wiring the whole thing up, screwing the front panel into place, sealing up all the leaks around the drivers and equalizing the levels of the drivers. When he took the final well engineered product home he discovered the woofer sounded rotten. He had spent $15, and the only useful driver in the batch was the lowly $3 CTS tweeter. That's how he went wrong.

-- Jim Nichol (Massachusetts)

Western Technologies Reference 2S Loudspeaker

It is always satisfying to see a worthwhile audio product come to the market through the energy of a dedicated music lover. Darrell Spreen is such a man, and his company Western Technologies (9819 McKnight St. N.E., Albuquerque, NM 87112) was formed to offer the Reference 2S speaker system to audiophiles. A systems analyst involved with laser research for the Air Force, he has spent the past fifteen years designing, building and evaluating loudspeakers in his leisure time. During a two-year search for a commercially available speaker for use in his home (he decided on the Celestion Ditton 66), he developed a hierarchy of qualities he felt important for speakers to possess. An outline will complement my review of the Ref 2S, as his design goal was to achieve as many of these qualities as possible in the final product.

The level 1 qualities are those which Darrell feels contribute to a speaker's accuracy; they include bandwidth, balance and smoothness. Level 2 qualities, low distortion and definition, result in transparency of sound. Those of level 3 are rare among speakers and contribute to realism of sound. They are musicality, depth and three-dimensionality, delicacy and aliveness. In designing the Reference 2S, he addressed level 1 and 2 qualities first. He then sought those of level 3 through a lengthy process of building many prototypes and comparing them both to the Celestion 66's and to a planar-magnetic-design speaker. Throughout all of this, he kept in mind the objective of building a modestly priced system.

Darrell decided upon a reflex design for the bass region. His goals here were good mid-bass transient response, extended low bass response and minimal cone excursion down to 50 Hz. An eight-inch driver selected from among ten promising offerings was required for a very low cutoff frequency in a moderate size enclosure and reasonable dispersion into the midrange. Through his own research, Darrell had developed a plastic coating to damp spurious vibrations in cone drivers. He uses this coating on the woofer and midrange drivers. After trying several internal damping materials, he selected a dense urethane foam.

Two cone drivers and a piezoelectric supertweeter handle the mid- and high-frequency regions. This combination provides many of the level 3 qualities but necessitates an unconventional crossover design which, while maintaining phase integrity in the crossover regions, introduces a phase shift to simulate a dipole radiator. (Shades of Andromeda.)

He designed a column-shaped enclosure to house the four-way floor-standing system, placing appropriate drivers near ear level. The loudspeakers measure 34x11x9 1/4 inches (hxwxd) and cost $500 per pair. They will be available at three or four selected stores nationally.

Subjective impressions: It is important to note at the outset that during the first month of subjective evaluation of the Ref 2S, I had no knowledge of either the design principles employed or the drivers used. I say this mainly because had I known piezoelectrics were used, prejudices about the alleged harshness of these drivers would have become a factor.

All the usual comments, such as "I can't believe that sound is coming from those little boxes," apply to these speakers. In this case, however, if the listener allows himself to concentrate on the music alone, he will be delighted by the coherent soundstage presented when a fine recording is played. Driver blending is superb, or this would not be true.

I hesitate to break the audio spectrum into frequency regions, for there is such a strong sense of a continuum with the 2S, but this is probably the simplest way to communicate my impressions clearly.

Bass response is honest, with the pair making no pretense at filling out the very lowest octave. It is neither full nor lean elsewhere and lacks the euphonic gutsiness common to some [Which? -- Ed.] recent acoustic suspension designs. The upper bass is not overly taut, yet possesses the highest resolution I have heard in this region.

The midrange sound can only be described as exciting. This needs clarification, for it is not exciting because of built-in frequency response aberrations, as are some of the JBL's. Rather, there is a sense of tremendous dynamic range, akin to that of live music. There is a quickness to the response that can be exhilarating or grating, depending on program material. This comment includes the high frequency reproduction and therefore involves the piezo driver. There is great clarity, but stand by if the recording is too bright. It will be revealed immediately.

Imaging is very good. As I've said, the soundstage created is quite convincing. The speakers do not "over-image" the way the overly-bright KEF 104's do. Sound sources take on a natural size. Reproduction of depth beautifully complements the good lateral imaging. I miss only the subtle sheen of cymbals imparted or reproduced by some electrostatics.

Overall, the Reference 2S is a fresh-sounding speaker that lets the listener through to the music. I commend to your attention the Nakamichi-miked "Ngiculela" on Stevie Wonder's <u>Songs in the Key of Life</u> when auditioning these (or any other) speakers. Though the majority of my own listening has been done using tube electronics, the speakers work well with high-quality, medium-power receivers (20-60 Watts/channel) and other solid-state equipment. It should be noted that in these cases, a touch of stridency may creep in sooner with many pop recordings. This speaker will appeal to music-lovers, especially those audio perfectionists on a budget.

In the Boston area the speakers can be heard by contacting BAS member Gary Rancourt, (617) 369-1949.

-- Dez Fretz (Florida)

Harman/Kardon Rabco ST-7

The ST-7 is the latest in Rabco's line of tangentially tracking tonearms. With the $430 ST-7 (and the new stripped down version, ST-6) Rabco has begun a new partnership with Harman/Kardon (H/K bought them out), which hopefully will solve the problems of the past. The ST-7 is a belt-drive turntable with manufacturer specs of -68 dB (DIN B) for rumble and .04% (NAB weighted) for wow and flutter. Neither rumble nor wow nor flutter from the ST-7 are audible to me.

The tone arm is very low mass (about six grams), which can be made even lower (to about four grams) by removing the sliding collar and balancing with the counterweight. The tone arm lifts up at the end of the record (triggered by a photo cell) but does not return to the starting position. This photo cell has been one of the major problems with the turntable. Unfortunately, it sometimes gets out of alignment and causes the tone arm to lift up in the middle of the record or not at all. I had this problem with my first unit (after I returned it twice they gave me a new one) but have had better luck with the second (serial number 2122574). Still it occasionally (three times in six months) lifts up at the wrong time for some unknown reason. (Maybe light from the room just happens to hit it right.) Harman/Kardon now includes an excellent technical manual, which describes how to make the adjustment of this cell yourself.

Cueing is accomplished by a lever which moves forward and back and operates very smoothly. The tone arm can be moved only with the cueing lever in the up position. On my unit there was some drift in the cueing on descent, which I fixed with some help from the previously mentioned technical manual (others manufacturers note).

The acoustic isolation from feedback is not too good, and the ST-7 can howl badly or just muddy up the sound if you like to turn your music up loud or if you like a lot of bass. Two solutions are the Netronics base ($17.50) or the Audio Technica feet ($25.00). The cheapest method of solving this problem is to put a piece of 1/2" to 1" foam under the turntable.

The turntable is turned on and off by a rocker switch located under the front of the turntable. I use this switch only when listening to some source other than records. Normally I leave the table turned on and control it through the switched outlet on the back of my preamp. The choice of speed (33 or 45) is selected by a classy lighted arrangement operated by touch. There is a strobe on the left side of the platter, which is fairly easy to see from overhead. The turntable has enough torque to use Discwasher fairly easily, and the Dustbug does not cause drag. There is, however, a slight amount of drift in warmup.

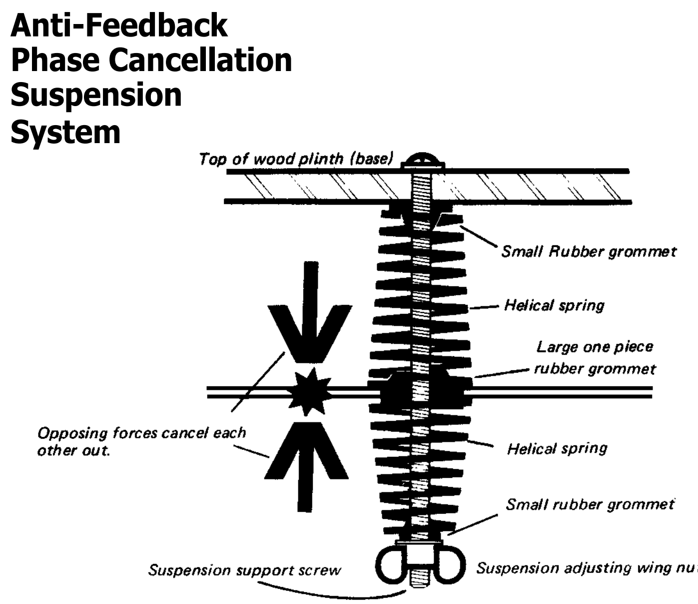
Like Mr. Shedd (review of ST-7 in the April Speaker), I have not noticed any brightness in the upper midrange. It could be that I just don't listen as critically as The Absolute Sound. I have found the turntable mates well with my ADC XLM. The ST-7 is an excellent turntable with a low-mass arm that includes all the advantages of tangental tracking (no tracking error, no anti-skating, etc.) and also includes the convenience of arm lift-up at the end of the record.

-- Brad McCoy (Ohio)

Fergus Fons and Formula 4

Mr. Hamish Robertson seems to have spent much of the past few years roaming the British Isles designing turntables. First he gave us the Linn-Sondek, then the Ariston (reviewed in the March issue), and now the Fons CQ30. Growth is evident in this progression: the single-speed Linn-Sondek followed by the two-speed Ariston followed by the infinitely variable speed (29 to 100 rpm) Fons. One can only wonder what he will do for an encore.

A common element in these turntables is the suspension system, with the turntable and arm being suspended on a subchassis by springs. The Fons uses a T-shaped subchassis, with the two members of the T being made of different metals to minimize resonances. This subchassis is suspended from the main chassis by three sets of double springs, which the manufacturer claims cancel upward and downward forces (see sketch), thus providing considerable isolation from acoustic feedback and vibration. It seems to work; my five-year-old daughter jumped rope within a couple of feet of the turntable while a record was playing with no audible effect -- through the speakers, that is,



The arm mounting board is cantilevered from one arm of the T, and herein lies the one real problem I've had with the Fons. Supposedly the turntable and arm mount can be leveled by means of wing nuts on the three spring mounts. Well, one or the other, but not both at the same time. I finally resorted to shimming the arm board where it attached to the T, and after several hours achieved modest success in getting everything level.

The Fons features belt drive (another Robertson characteristic) via a dc servo motor (a new touch). There are pushbuttons for Off, 33, 45, and 78 rpm and three twenty-turn potentiometers for speed adjustment, giving an overall continuous range of 29 to 100 rpm. Why would anybody want it, you ask? Fons suggests some possibilities. Record cleaning can be done faster at higher speeds. Taping discs can be speeded up by playing the disc at 66 2/3 rpm and running the tape recorder at, say, 15 ips and playing back at 7 1/2 ips, which I guess is all right if your tape recorder can make it up to 30 kHz at 15 ips and your Shibata stylus can stay in the groove at 66 2/3 rpm, or you really don't give a darn what your tapes sound like. I find it desirable to have some control of speed accuracy at 33 1/3, and about the only way to get that is to accept the other speeds. A nice feature is that the platter can be held by hand for cueing or when changing records without harm to the motor or belt.

The transformer for the power supply for the speed control circuit is located as far as possible from the arm and cartridge path, and hum is not a problem. (If hum were a problem with a particular cartridge, it would be a simple matter to outboard the transformer.)

The platter is a one-piece die casting and is supported by a bearing claimed to have a concentricity of ±0.00003 inch. Rumble is rated from -64 dB to -78 dB and wow and flutter at 0.03% to 0.06%, depending on weighting. In practice it is an extremely quiet turntable.

The platter, however, is not readily removable, making such usually simple maintenance as belt-changing a matter for the service shop.

The Fons is a ruggedly built and excellently performing turntable (as are the other Robertson designs), but cosmetically it is lacking. The manufacturer's literature makes a point of its being "handcrafted." I think this is a euphemism for "put together in the cellar." The lovely platter casting is topped off by a rubber disc of about 7-inch diameter at the center and a ring about 1/2 inch wide at the outer edge. Both the disc and the ring look as though they were cut to shape with scissors. They are both glued to the platter, and thus are not easily removable. Placing a level on the inner disc produces different readings depending on location, making the leveling job tenuous (and tedious) at best.

In the depression between the disc and ring, strobe markings are painted on the platter: very nice if you have a transparent record for setting speed. The speed selector pushbuttons are mounted directly on the pc board and tend to skew one way or another depending on how well they were fitted to the board. On my sample, at least, the cover lift (a strip of plastic glued to the dust cover) was hanging unglued at one end. The bottom cover is a thin piece of hardboard screwed on at the corners; it tends to sag below the base in the middle. Otherwise the woodworking is okay but not spectacular.

Whether these cosmetic blemishes outweigh the performance is a matter for individual agonizing. I found them to be either fixable or tolerable.

Mounting the Formula Four (Mayware) arm on the Fons is a relatively easy, straightforward task (except for the leveling problems noted above). However, be sure to buy your Fons with an uncut mounting board: many are precut for SME's. Installation requires drilling a 1-inch-diameter hole in the mounting board for the arm and a 1/4-inch hole in the base for the armrest. The latter hole is nicely taken care of by removing the screw holding the trim plate on the Fons, drilling out the hole, and substituting the armrest for the screw.

Comments appearing in previous issues of the Speaker regarding the fussy setup of the Formula Four are accurate, PHD's comment in TAS about setting up the arm in ten minutes notwithstanding. With some cartridges there could be a problem with clearance between the arm pivot and the turntable cover, even with the latest (lower) version of the arm. With a Sonus Blue the clearance is only about 2 or 3 millimeters.

Clearance under the base for the long cable connector on the Formula Four is barely adequate. I cut a large hole in the bottom cover and set the turntable feet on raised pads (about 1/2 inch) to insure adequate clearance.

The Formula Four is grounded by means of a wire contained within the cable sheath, and there is no easy way to break into this sheath to connect a ground wire from the turntable chassis. A separate ground wire must be run from the chassis to the preamp. The Fons provides a ground lug for this, but it could easily be overlooked.

The Fergus Fons and Formula Four combination, in addition to being alluringly alliterative, provides a nicely matched, high quality record playing system. I'm still trying to reach some conclusions about the Sonus Blue, which is proving to be something of an enigma. At the moment my reactions are positive -- but more about that later.

Bob Borden (Massachusetts)

More on the Formula 4

An article on tone arm geometry in the first issue of The Audio Critic stated, in effect, that few arms are designed for proper cartridge offset and overhang. It proceeds to give what they claim to be the correct setup for minimum distortion for all arms. [See also Mitch Cotter's presentation in the April 1977 Speaker. -- Ed.] Proof enough was there to cause me to check out my Formula 4. It was way off. The measurements are painstaking and tedious. I had to elongate the arm mounting hole in order to obtain correct overhang, and rotate the cartridge in the headshell slightly to obtain correct offset angle. The latter is easy with the Formula 4 because of slotted cartridge mounting. The results were more overhang and a greater offset angle than with the original setup. Skating force is increased, as overhang and offset angle are increased, so experiment with increasing the antiskate weights.

Was it worth it? Yes. In fact, that one "how to - no cost" upgrade was worth the $28 subscription price of the magazine. How? In reduced distortion, which was manifest not immediately but over extended listening, producing less strain and fatigue.

The details of measurements and calculations are far too involved to restate here. Formula

4 owners may also refer to the Speaker, Vol. 4, No. 8, and Vol. 4, No. 11/12, for further information.
-- Dow Williams (California)

Damping the Black Widow

Recently I bought an Infinity Black Widow tone arm and a Sonus Blue Label cartridge. I was quite pleased initially with the great definition and the open, airy quality the combination provided. But I soon noticed that the cartridge did not image quite as well as I had hoped. In fact, though better than the Shure V15-III in an undamped arm, it was not as good in this respect as the Shure in my old Technics SL-1300 paddle-damped arm.

The Black Widow's effective mass is quite low -- only three grams, according to Infinity. That should place the low-frequency arm/cartridge resonance above the range of record warps, so I doubted that damping would help. Nevertheless, I decided to give it a try. I made a paddle of balsa wood and attached it to the arm with Elmer's glue. The results were all I had hoped for. Even with this very low-mass arm, disc reproduction improved in the same ways noted with other arms and cartridges, i.e., more stable and clearly focused imaging, tighter bass and cleaner high-frequency reproduction.

I should note that my Sonus is, according to the serial numbers given in The Absolute Sound, an older, higher-compliance version. More recent samples may not benefit as much (perhaps not at all) from damping when used in the Black Widow. Also, I have found that I can alter the sound of the cartridge considerably just by varying the level of damping fluid in the tank. Too much STP and imaging becomes even more precise, but the sound becomes less open, rather constricted and flat. You must be careful not to overdamp.
-- Brian Miller (Illinois)

Disctracker and the Phoenix/Graham Damper

In the March issue of the Speaker, the Disctracker damping device is described, along with the Phoenix/Graham device. If I understand their operation correctly, it seems that their effect on tone arm/cartridge operation would be quite different because the Disctracker references the stylus to the record surface, while the Phoenix/Graham mechanism references the stylus to the turntable base, and assuming a platter with negligible axial runout, therefore to the top of the platter.

To illustrate how this difference could affect turntable operation, consider this example. Assume that the stylus were encountering an upward record warp. The Disctracker would act to move the arm upward also, thus minimizing stylus deflection. In the same situation, the Phoenix/Graham device would tend to hold the arm stationary with respect to the platter, thus inhibiting it from moving upward with the record, and increasing stylus deflection.

If my analysis is correct, it seems that the action of the Phoenix/Graham device could be counter-productive. Perhaps this observation would help to explain the controversy over this damping device.
-- Roy Mallory (Massachusetts)

If a pivot-damped arm is excessively damped, then the arm will indeed resist any change in direction, as Mr. Mallory suggests. In recent work I've done on tone arm design, however, I've discovered that if the damping is applied sparingly (optimum damping), and if the effective arm mass is low, that is, compatible with the cartridge, then near-perfect performance is obtained. Don't forget that the purpose of damping is not simply to ride out record warps -- it's also for minimizing tone arm resonances and improving reproduction, even from a perfectly flat disc.

Now, what about the top of the warp on a record? As the arm travels up the side of the warp, the Disctracker will stabilize the stylus all right, but at the crest of the warp, it seems to me that the only force (other than tracking force) working on the arm is inertia, which now has control of the arm. This means that the arm would like very much to jump off the disc altogether (if the warp is sharp enough and inertia high enough), and there really isn't any control over this, as the

Disctracker can't grab onto the air above it. In this instance, I think pivot-damping would have the advantage, because it would be able to control this effect.

What this all boils down to is this: all record playing devices are imperfect in one way or another, so far as I've seen. If the design is thorough enough, then the shortcomings will be slight. I suspect that the Disctracker will make a great improvement in undamped arms, and the theoretical weakness I pointed out may be unnoticeable in practical usage. Just so, a well designed pivot-damped arm can be made to behave properly and will yield a dramatic improvement in performance over most, if not all, undamped arms. (Perhaps the ultimate arm would be a good pivot-damped arm with the Disctracker attached. The theoretical shortcomings of one system would then be corrected for by the other. When the BAS tests the Disctracker, I'll investigate this possibility.)

-- Bob Graham (Massachusetts)

Effects of Spatiality on Musical Tones

Among the many presentations made at the recent meeting of the Acoustical Society of America was one intriguing paper entitled "Cycloramic Stereotone Positioners for Weaving Streams in Space." In the words of the author, David Goldsmith, "the introduction of spatiality as an integral dimension of musical tone exerts an influence upon the net character of the music produced that is so profound as to make such music generically distinct from the prior art."

The basis of Goldsmith's concept is the physical displacement (real or apparent) of individual tones within the harmonic structure of a chord or tonal motif. For example, suppose the listener is presented with a C-major triad, i.e., C-E-G, such that all the tones appear in front (at 0 degrees in the horizontal plane). Add to this the submediate tone "A," creating a minor sixth chord. Now, if the "A" is "moved" around the listener, the effect (theoretically) is a "weaving" of tones in space, with the stable triad holding its ground, and the seductive "A" asserting itself from various positions (terminology mine). Any or all of the tones can be likewise manipulated at the will of the composer, conductor, or even the listener.

The technical problems created by this concept may be enormous. Presently, Goldsmith employs a microphones-and-baffle system which can be rotated to pick up or reject incoming sound from various instruments. Alternately, electronic switching among stationary mikes would provide the necessary displacement cues.

The earphone-conveyed information would alter the subjective position of the listener, and visual cues should aid in the discrimination of sounds coming from in front and behind. No doubt, the system could be adapted for four- (or more) channel reproduction, which may actually enhance the experience as well as obviate the need for visual reference in direction determination.

According to Goldsmith, results so far have not been particularly impressive for several reasons, the main one being his use of general-purpose microphones designed more for P. A. work than recording. However, a glance at the baffle setup being used reveals some serious shortcomings. Cavity resonances and undamped panel vibration could be major contributors to sonic coloration. Mechanical noises are bound to creep in during rotation, regardless of how well the mikes are isolated. For purely technical reasons, then, one is tempted to view the system with a jaundiced eye.

Furthermore, in this author's opinion, consideration must be given to the esthetic value of such tonal manipulation. Although I hesitate to prejudge, it should be pointed out that the idea is not altogether new. Current trends in aleatoric music feature the transfer not only of tones, but of rhythmic motives as well. Some spectacular sonic punches, remarkably like those described, have been pulled by several of the "heady" rock groups, but, alas, only in the studio.

Goldsmith, however, maintains that his conceptualization is unique in all respects. If his idea catches on, we may witness the genesis of a musical dogma steeped in the soup of tonal migration. At least one form seems most likely to pick up the cue, and that, of course, is synthesized music. But whether or not modern composers will lend an unbiased ear is open for debate.

-- Dan Lilley (North Carolina)

More on Audio/Pulse

Re Gerald Larsen's report on the Audio/Pulse Model One that appeared in Vol. 5, No. 8, of The BAS Speaker. As a user of the Audio/Pulse I would like to offer the following additional comments:

1. I use as primary speakers the Crown 224 bi-amplified and as secondary speakers the small Braun (or ADS when manufactured in USA). The relation of available rms Watts between the primary and secondary amps is fifteen (375 Watts per channel vs. 25 Watts per channel). The result is very well balanced, even at high outputs (105 dB plus), which demonstrates the flexibility of the Audio/Pulse-based ambience system with respect to equipment compatibility.

2. My Audio/Pulse is driven from the main output of a Levinson JC-2 preamp. This procedure has the advantages of having the loudness control of the JC-2 act as master volume control of the whole system and of being able to fix, with all the necessary care, the relative outputs of the primary and secondary speakers, solving in this way the only critical calibration permanently and perfectly. On the other hand, with this procedure the level match of the Audio/Pulse has to be reset when significant changes in loudness are applied to prevent overloading of the Audio/Pulse circuits. The key word is "significant." I've found that for differences in output of the order of 15 to 20 dB one setting is enough. In this way the Audio/Pulse might flash the red overload signal, but there are no audible effects. It is better to overload than to underload; when underloading, noise from the Audio/Pulse can be distracting.

In other words, I think Mr. Larsen's comments on overloading with rock music are true visually (you see red lights) but not audibly. The alternative of driving the Audio/Pulse from the tape outputs of the preamp poses the problem of matching the output levels of both speaker systems for each position of the preamp volume control, something which is difficult to do correctly (in a hurry) and which has to be done perfectly if a good ambience effect is to be achieved.

3. It is true, as Mr. Larsen states, that when changing the level match in the Audio/Pulse, noise and oscillations are heard through the secondary speakers. To avoid this, any change in level match must be preceded by cancelling the output of the secondary speaker. This can be done by disengaging the decay time switches and engaging the appropriate one immediately after setting the level match.

4. Keep the Audio/Pulse on all the time. This is a very good way of avoiding equipment failures caused by on-off pulses, except for components containing tubes or motors.

\- H. Gallegos (Peru)

Soundcraftsmen Equalizer with Sound Concepts Delay Unit

Collins Beagle and Victor Campos are surely correct when they remark that any extra component in the signal chain will degrade the signal to some extent. On the other hand, stereo reproduction is a fake anyway, so the real question concerns the tradeoffs required to gain the best illusion in an imperfect world (cf. the discussions on tone arm damping). I offer some of my experiences with an equalizer and a delay line as examples.

I am blessed with a room of reasonable size and mostly non-parallel surfaces as well as the fortunate circumstance that decor dictates (would you believe) that my dipole radiating electrostatic screens be placed ten feet from the rear wall, thus providing me with about twenty milliseconds of built-in delay. However, aesthetic considerations and a militant spouse also require that the screens be elevated from the floor no more than seven inches. As a result, I have a broad 3 to 4 dB hump centered around 320 Hz. The subjective effect of such a hump is to obscure the airiness and openness Mr. Beagle and I both prize. Furthermore, my screens don't sound alike (nor did the Quads that preceded them), probably because of some unavoidable asymmetries in room placement. This, of course, makes for stereo images that wander slightly with frequency. My tradeoffs were clear: I could get a new room and a new wife, or I could get an equalizer. So that my quest for purity not lead me into the paths of vice, I purchased a Soundcraftsmen. The sound may be more veiled than before, but it seems less so.

I think most people are apt to suffer from too much upper or mid-bass. If room placement won't solve the problem, follow Daniel Shanefield's advice and squash the devil out of that mid bass. As he also remarked, the purchaser of an equalizer who has the BSR-Lafayette-Radio Shack SPL meter is likely to end up boosting the treble far too much, relying on meter readings rather than his ears to give him flat response. Several correspondents have noted that going for flat response can give unmusical results anyway, as many records are boosted at the high end to sound well on cheap phonographs (and many studio monitors). I have discovered another pitfall: the pink noise levels on the Soundcraftsmen equalization record I have are down several dB at the very high frequencies (above 10 kHz) when compared with those on the CBS 140 acoustical test records. Unless your instruments are very good, it would seem wisest in such matters to trust your own sense of proper musical balance, especially at the extremes of the frequency spectrum.

My other deliberate departure from purity came with the acquisition of a Sound Concepts delay line. My experiences with this device seem closely parallel to what Gerald Larsen reported about his Audio/Pulse; it is a striking advance in illusionism. I had always supposed my speakers to reproduce depth unusually well, but the effect I now hear is several orders of magnitude better. The effect is most noticeable on large choral works (try the Previn/EMI Alexander Nevsky for a revelation), but it is also pronounced on the Telarc Direct from Cleveland (does anyone else find the dynamic range of this record surprisingly restricted?) and definitely perceptible, but more subtly so, on the Leonhardt/Telefunken Goldberg Variations. I find that records with good natural ambience work best, especially if one tries to match the ambience already there rather than to drown it. Closely and multiply-miked records are a problem, since it is often impossible to integrate ambience and auditory perspective satisfactorily.

Mr. Larsen is also correct about the usual store demonstrations of delay lines. One of the local salesmen, a serious audiophile, had privately supposed that the units he was selling were just a gimmick, but his jaw dropped when he heard one in my living room.

On certain recordings -- especially some large choral works - the stereo stage actually spreads out to the left of the left speaker and to the right of the right speaker. This does not occur with smaller works; delay lines do not generate five-foot violins. In general, the image is not thickened and blurred, but, on the contrary, is often better focussed.

I have the impression that people are putting down the Sound Concepts in favor of the Audio/Pulse for insignificant reasons of the type so common among audiophiles: Audio/Pulse is (literally) flashier looking [Check out the new Sound Concepts SD-550. -- Ed.], Sound Concepts doesn't imitate Carlsbad Caverns as well, and Audio/Pulse has more electronics inside (at almost the same price, too). There are some more plausible-sounding advantages of Audio/Pulse: the rear channels are not phase-coherent, as they are in Sound Concepts, and concatenated bucket-brigade chains, such as those used by Sound Concepts, are allegedly prone to slap-back effects at long delays. In fact, I haven't found that either of these differences has been important to my in my listening, and I have found Sound Concepts to have its own minor virtues. For instance, delay and reverb are continuously adjustable, and I have found that I prefer about 35 to 37 milliseconds of delay for most material. This setting falls between two discrete delays on the Audio/Pulse. Furthermore, I have never had to adjust the input level of my unit, even though my preamp volume control has been turned to widely different positions.

This is the first four-channel system that has pleased me. My rear speakers, amp and a Sansui QS-1 had been gathering dust for over three years, and a four-channel fanatic with excellent equipment and sophisticated SQ decoders never succeeded in altering my opinion.

Finally, if you do get an equalizer or delay line, remember to employ them as Hamlet recommended:

> But use all gently: for in the very torrent, tempest, and, as I may say, whirlwind of your passion, you must acquire and beget a temperance which will give it smoothness.

-- Larry Hardin (New York)

In the Literature

Journal of the Audio Engineering Society, May 1977

*Floating-Point Encoding for Transcription of High-Fidelity Audio Signals: Based on psychoacoustic data of masking of noise, a floating-point representation for signal amplitude quantization is derived which yields 100 dB dynamic range with no preemphasis/deemphasis (p. 266).
*Tracing Distortion Correction: A technique is described which uses a variable delay analog shift register for reduction of tracing distortion (p. 273).
*A Low-Noise High-Output Capacitor Microphone System: An FET feedback amplifier has been designed for the Schoeps MKT-45 capsule to produce a 125 dB dynamic range above a 15 dB SPL noise floor (p. 278).
*A Novel Approach to Linear Phase Loudspeakers Using Passive Crossover Networks: The theory behind, and square wave tests of, the Bang & Olufsen "phase link" crossover design is presented (p. 284).
*Design Problems of High-Level Cone Loudspeakers: A review of the problems encountered and tradeoffs necessary in designing a speaker to produce high sound levels (p. 294).
*A Microprocessor-Controlled Digital Waveform Generator: Two developments are discussed which aid in the design of an audio generating device which acts under direct computer control without the need for an A/D conversion step (p. 299).

Journal of the Audio Engineering Society, June 1977

*The Application of Digital Techniques to the Measurement of Loudspeakers: A detailed discussion of the characterization of loudspeakers through digital processing of the impulse response (p. 370).
*The Dynamic Vibration Absorber Principle Applied to a High-Quality Phonograph Pickup: The design of a tuned damping element is described for reducing stilus resonant peaks beyond 20 kHz (p. 385).
*A New Set of Vented Loudspeaker Alignments: The theories of Thiele and Small are used to design a speaker with a single response peak at the low frequency cutoff ("boom box") (p. 391).
*An Almost Locked Oscillator for Electronic Music Synthesis: Warmth can be provided in electronic music by utilizing the low order beating of tone generators which are not strictly phase locked (p. 394).
*A Patchable Electronic Music Percussion, Synthesizer (p. 395).
*Criteria for Evaluating Surround-Sound Systems: The analysis is presented without the assumption of a specific number of channels (e.g., 4) (p. 400).

Audio, August 1977

*Behind the Scenes: Bert Whyte reports on interesting equipment and ideas encountered at the L. A. AES convention (p. 16).
*Tone Arms: In part three of his analysis of turntable design, Joe Grado shows how the design of the tone arm affects both its own resonance and its transmission of external vibrations to the pickup (p. 28).
*Power Amplifiers and the Speaker Load: An analysis of how loudspeaker reactance causes misbehavior or increased stress in power amp circuits; densely written but worth the struggle to read through (p. 32).
*The Search for an Optimum Transmission-Line Speaker: Beginning with a nice introduction to signal flow-graph analysis, W. Hoge applies the analysis to transmission-line speakers. After concluding that conventional transmission-line speakers are uncomfortably low in efficiency, he pulls out of the hat the surprising idea that a horn system is a high-efficiency transmission line (p. 44).
*Time Alignment in Loudspeakers: E. M. Long discusses the general idea but does not give specific details on his proprietary time-alignment technique for speakers (p. 58).
*Equipment Profiles: Favorable test reports on the Sony 6800 receiver, Tandberg 330 three-head cassette deck, AKG C451E condenser microphone, Pioneer 510A direct-drive turntable and Optonica 3535 cassette deck. Unfortunately, the C451E was tested only with its cardioid capsule, showing its characteristic bass rolloff; the potentially more interesting omni capsule was not tested (p. 66).

The Audio Critic, No. 3

*A short 22-page issue devoted solely to short reports and preliminary sonic impressions of audiophile products seen and heard at the Consumer Electronics Show in June. Favorable notes on ADC, Audionics (Berning), Beveridge, Breuer, Bryston, Cizek, DB Systems, DCM, Dunlap-Clarke, Hafler, Hegeman, Infinity, Koss, Linn, Levinson, Naim, Promethean, Pyramid, RAM, Rappaport, SAEC, Symdex, Symmetry, Van Alstine and Verion. Particular raves for Beveridge, Cizek and the Levinson HQD (a $20,000 system). Condemnations of Acoustique 3A, Bertagni and ESS/Heil. News items: Mike Wright has left Dayton-Wright to start Watson Labs, John Curl is with Symmetry and Pyramid is Richard Sequerra's new company.

db, June 1977

*Custom Disc Mastering: A look at a Nashville disc studio (p. 35).
*A Portable Oscillator: A single-frequency sine-wave oscillator; would be better if redesigned around the BAS oscillator circuit (p. 38).
*Test Report: A favorable report on the Shure M615 analyzer and SR107 equalizer for room equalization (p. 42). But the Ivie analyzer will be a better buy if it turns out to be accurate (p. 41).

db, July 1977

*The L. A. AES Convention: A look at new pro audio products (p. 27).
*A FET Audio Limiter: Design details and schematic (p. 32).
*Test Report: Technics 1500 open reel tape deck with Iso-loop transport; as discovered by other investigators, the transport is terrific but the frequency response is not as flat as it should be (p. 34).

Electronics, July 21, 1977

*The Right Gyrator Trims the Fat off Active Filters: In theory, LC filters have the lowest sensitivity to component variations. This shows how these filters can be built without the traditional problems associated with wire-coil inductors by using a gyrator and capacitor as an inductor substitute.

Gramophone (England), June 1977

*Sounds in Retrospect: Surveying the sound quality of some recent releases (p. 113).
*High Fidelity 77: Notes on new equipment at the Heathrow hi-fi show (p. 117).
*Equipment reviews: KEF Cantata (excellent), Sansui QSD-2 quad decoder (good) and the Stylift automatic end-of-side arm lift for manual turntables (interesting and effective) (p. 122).

High Fidelity, August 1977

*News and Views: A note on a $2000 impulse noise suppressor and a look at Andy Rappaport (p. 27).
*Equipment Reports: Mostly favorable reports on the Tandberg 330 three-head cassette deck, Koss ESP-10 headphone, Optonica 2050 cassette deck, Hegeman Input Probe and Hitachi D-800 cassette deck. The Hitachi's response curves are rather sloppy by today's standards; compare the Optonica, for instance (p. 39).
*Tape Recorder Tests: Background on how the tape deck lab reports are done (p. 48).
*Buyer's Guide: A list of most of the currently available tape decks (p. 52).
*Taped Talk: Notes on acquiring oral history via portable cassette recorders (p. 61).
*The Tape Deck: R. D. Darrell reports on the resurgence of good-quality pre-recorded open-reel tapes from Barclay-Crocker and The Reel Society (p. 96).
*Resurrecting the Beatles: How a lousy 1962 mono tape was turned into a good, commercially successful recording (p. 101).

Modern Recording, August 1977

*From Tape to Disc, Part I: A look at the mastering process (p. 24).

*Shelly Yakus: An interview with a pop recording engineer (p. 30).
*Session with the BeeGees: How a pop studio recording is made, with details on mike placement (p. 38).
*Ambient Sound: An essay on bi-amplification (p. 44).
*Lab Reports: Delta-Graph EQ-10 modular octave equalizer kit (performance very good, assembly instruction manual poor, with shock hazard noted). Spectro-Acoustics 210 octave equalizer (excellent). Advent 201A cassette deck (a very strange report: the data reveal excellent performance, but the report writers go out of their way to find things to complain about, even making some measurements with TDK Audua tape, despite Advent's clear instruction that Audua is not recommended for use in the 201A). Yamaha P-2200 power amp (excellent at low frequencies, not quite perfect at high frequencies). (p. 46).

Popular Electronics, August 1977

*Stereo Scene: Ralph Hodges discusses four developments which will have a major effect on the future of tape recorders: the Hall-effect head (needs less EQ, has less phase shift and less noise than conventional heads); Metafine iron-particle tape (with about 10 dB more headroom than current tapes but requiring more bias than present tape machines can deliver); the Iso-loop transport, used in the Technics 1500 deck; and PCM digital recording (p. 14).
*Test Reports: Julian Hirsch raves about the Heathkit AR-1515 receiver, Thorens 126 turntable with Isotrack low-mass arm, and Ortofon MC-20 moving-coil pickup with MCA-76 head amp (p. 29).
*Choosing Portable Tape Recorders: Basic buying guidance (p. 43).
*How to Handle MOS Devices without Destroying Them: Useful advice for the home-brew experimenter, now that MOS ICs are invading the audio field (p. 67).

Radio-Electronics, July 1977

*TIM Distortion - How It Affects your System: Len Feldman explains TIM and several ways of measuring it. He touts TIM as the long-sought explanation for the difference in sound between tube and solid state power amplifiers without mentioning the still active controversy among many audiophiles over the significance of TIM (p. 47).
*Test report on the Fisher RS-1080 stereo receiver (p. 50).

Radio-Electronics, August 1977

*Test reports on the Optonica ST-3535 tuner and Hitachi SR-903 stereo receiver (p. 54).

Stereo Review, August 1977

*Audio Q & A: Larry Klein talks about groove pre-echo and other matters (p. 24).
*Tape Talk: Craig Stark tells the exciting story of Metafine iron tape and the Hall-effect playback head (p. 28).
*Test Reports: Onkyo A-7 amplifier, Heath 1307 audio power meter kit, Koss ESP-10 headphone, Harman/Kardon Citation 16A power amp and SAE 2800 parametric equalizer. The Heath 1307 peak-reading power display unit looks especially attractive and has switching for speaker and amplifier comparisons (p. 30).
*Speaker Design: Straight talk on the compromises and design choices speaker engineers deal with (p. 62).
*Distortion in Loudspeakers: Surprising information, destined to be controversial, on the audibility thresholds of various kinds of distortion, based on experiments in which controlled amounts of distortion were injected into musical signals (p. 68).
*Speaker Kits: Collected data on the available kits (p. 74).

Wireless World, June 1977

*Purpose-built Matrix H Decoder: While the BBC surround-sound experimental transmissions may be received by modified commercial decoders designed for other systems, this decoder is specifically designed for matrix H (p. 34).
*Transient Intermodulation Distortion: The claim of a letter, that TIM is just a new name for distortion caused by slew-rate Iimiting, is refuted by M. Otala (p. 47).

*Interactions of Loudspeakers and Rooms: Basic discussion of how wall reflections modify the sound of loudspeakers and some techniques for optimizing speaker location.
*Broadcast Stereo Coder - 2: Continuation of construction article for a FM stereo coder useful as a test instrument for aligning FM tuners (Part 1 in April 1977 issue) (p. 75).

Wireless World, July 1977

*Multi-system Ambisonic Decoder: Part 1, in a series of articles giving the basic design philosophy behind a decoder capable of decoding all major existing and proposed two-channel surround-sound systems (p. 43).
*A letter describes a method of reducing TIM in power amplifiers by using a very low collector current in the first stage to limit its bandwidth.

-- Peter Mitchell
-- John Schlafer

July BAS Meeting

The July meeting of the BAS was held at the GTE facilities in Waltham. Jim Brinton opened the meeting with the following announcements: (a) persons interested in running for office in the BAS should submit their names; (b) we have been informed by the Discwasher people that samples of their tone arm damper, the Disctracker, will be sent to the BAS for evaluation; and (c) no word has yet been received from either Sabtronics (makers of the digital multimeter kit) or Delta-Graph (ten-band graphic equalizer) concerning our inquiry about a bulk purchase. Jim promised to give us an update on these inquiries at the next meeting.

Frank Van Alstine

Frank Van Alstine, modifier of Dynaco amplifiers and producer of the new Van Alstine Model One preamplifier, was the first speaker of the evening. He began by citing some key points of his audio "philosophy" and how they relate to the evaluating and modifying of electronic components. The first point he raised centered on the question: Are you sure your evaluations of audio products correctly describe what is actually taking place? Product evaluators of both the mass-market and esoteric audio magazines can be deceived by perceived changes in the sound of a system containing a new component. For example, a reviewer might take a system with which he is intimately familiar and substitute a new preamp for his old one in order to "test" it. Any differences he then hears, Van Alstine argued, will be attributed to the newly introduced preamp. The problem with this is that the new preamp may be so much better than the previous one that it reveals troubles elsewhere in the system. If the system with the new preamp sounds "brighter" and "grainy," it may be the case that one is hearing problems with the arm bearings or cartridge which the other preamp obscured. In this way, he argued, reviewers often end up evaluating the wrong component in a system. He also suggested that sonic differences between phono cartridges, especially those between samples of the same cartridge, are probably more often caused by differences in the adjustment of the tone arms (e. g. , vertical tracking angle, overhang, etc.) used to evaluate the cartridges than to the cartridges themselves. Van Alstine claimed that one can easily hear differences as small as two minutes of arc in vertical tracking angle, differences which could be mistakenly attributed to the cartridge involved.

He next offered a statement of what he regards as the purpose of an audio system: "to recreate the illusion of live music in your room in three dimensions. " The more accurately "the output follows the input the better the system is going to sound. " On this latter point he criticized Bose (though not by name) for building speaker systems that do not attempt to accurately reproduce the sonic equivalent of their input but instead add the "distortion" of artificial concert-hall reflections to the sound.

As to the proper way to evaluate audio components, Van Alstine remarked that he regards listening tests as the "objective" test of a component because the "object" of the component is to play music. Electronic measurements he argued, are "subjective" because they don't, at this time, correlate well with the sound reproduced.

The Van Alstine modifications of Dynaco components are expressions of this philosophy, he maintained. Dynaco products are well designed components which employed parts which often interfere with the function of the unit intended by the designer. Therefore, he said, he substitutes higher quality electronic parts for the originals to enable the units to behave as the designer intended. Among such changes he cited were the substitution of 1% metal film resistors for 5% carbon composition types, silver mica replacements for mylar capacitors, and the replacement of transistors with selected low noise units.

In addition to substituting higher quality parts, the Van Alstine modifications often include the addition of new circuitry, the most obvious being the outboard addition to the power supply of the Dynaco 400. This box, containing 100,000 uF of capacitance, increases the "pool" of B+ available to the amplifier, and therefore any draw on the B+ will result in less voltage drop and induced ripple than if the capacitive pool were smaller. The result, he claimed, is a better sounding amplifier. The ideal, in Van Alstine's view, would be to have a separate, high energy, low impedance power supply for each stage in an amplifier (inputs, pre-drivers, drivers, etc.) so that no stage could interact with any other through the power supply, which he claimed was a common, though generally unrecognized, problem.

Finally, Van Alstine offered the members some tips on how they might improve their own audio systems. Specifically, he suggested that one can damp the high frequency resonances in tone arms by pressing a small (1/4 inch diameter), 1/2 gram glob of modeling clay (the kind that does not harden) onto the rear underside of the headshell, just behind the cartridge. Also, he suggested, the hard dome in the center of many dynamic speakers can produce audible ringing, and that a disc of clay (1/2 inch for a midrange, 1 inch for a woofer) pressed onto the dome could eliminate this effect. For soft dome tweeters, he (rather tentatively) suggested poking a hole the size of a pencil lead into the dome. (Representatives of Allison Acoustics and Acoustic Research in the audience suggested yet another effect of poking holes in tweeters -- cancellation of warranties.) Van Alstine admitted that one could not tell in advance if this treatment would help his speakers and cited no satisfactory way to fill the hole in the tweeter dome if the treatment did not work.

A demonstration of the tone arm damping using modeling clay was then attempted with the following components: a Phillips turntable, ADC XLM cartridge, Van Alstine Model One preamp, Van Alstine modified Dynaco 400, and Cizek Model One speakers. After playing sections of the EMI pressing of Holst's Planets (Previn/LSO) both with and without the clay connected to the headshell, Van Alstine polled members as to whether or not they heard a difference (about half did) and whether the clay improved the sound (no clear consensus on this). Because of time pressures, great care could not be taken in this A/B test, and significant factors (e. g., tracking force) were only approximately equal. Recognizing this, Van Alstine suggested that each member of the audience try it on his own system and see if it works.

The ensuing question and answer session was lively and often heated, with Van Alstine being challenged on several of his assertions. Two of the most controversial points seemed to be his cavalier attitude toward electronic specifications (on the ad sheet for his preamp he cites as a damping factor "unit should not get wet") and his claim that substituting higher quality parts in a well designed amplifier would a priori result in a better sounding component. On the former, Van Alstine said that he really didn 't know the precise frequency response of his preamp but that it was essentially flat to within one dB "over several megahertz." He argued that this was of little importance anyway since, in his view, "raw differences in frequency response" were not a major factor in the differences between the sounds of preamps. This comment led to a heated exchange between Mark Davis of MIT and Van Alstine, Davis reporting that in tests done by BAS members, it was found that provided that noise was kept to a reasonable level, frequency response was the most significant factor in determining the sonic performance of a preamp. Further, Davis stated, once the relative frequency responses of various preamps were carefully matched, they became sonically indistinguishable. Van Alstine doubted that this was the case.

Davis also challenged Van Alstine's assertion that replacing 5% carbon composition resistors with 1% metal film units necessarily results in a better sounding unit. Davis argued that he could design an amplifier with a THD of . 0001% using only 5% carbon composition resistors and, since this amount of distortion was inaudible, he doubted the causal relationship Van Alstine was asserting existed between metal film resistors and sonic improvement. Van Alstine replied that this was a case of a distortion specification which, while applicable to sine waves, was largely irrelevant in the case of asymmetrical transients (music) and was therefore a specification which would not correlate well with how the unit sounded, and so on it went. None of these issues was resolved at the meeting, but the exchange of views did reveal with some clarity the wide range of assumptions, often quite contrary, which are held in the audiophile community. -- Richard Glidewell

[Note: Some of the material by Peter Mitchell was not presented at the meeting because of time limitations. For the sake of convenience, we are printing what he would have said had he had more time. Much of this material will appear in an upcoming article in High Fidelity. -- Ed.]

Peter W. Mitchell

If you walk into your local stereo emporium and confess that you are shopping for loudspeakers, you very probably will be offered a new Linear-Phase, Time-Aligned, Phase-Synchronized, Time-Shift Compensated, Phased-Array model featuring Coherent Phase and Improved Waveform Fidelity.

Manufacturers haven't agreed on what to call it, but phase linearity is the hottest topic in loudspeakers today. Is it spreading because it represents a genuine improvement in performance or value (analogous, say, to the general adoption of the acoustic suspension woofer after 1954 or the dome tweeter after 1959)? Or is it merely a fad intended to give some loudspeakers a sales advantage over the 250-odd competing brands?

Perhaps we should begin by dealing with the sticky problem of language. In engineering parlance, for example, a "minimum-phase" system is not one that has a minimal amount of phase shift; it is simply one in which the phase shift varies with frequency in a way that is predictable from the frequency response of the system. "Linear phase" is no less confusing. When High Fidelity's reviewers describe an amplifier as having linear frequency response they mean that its voltage response is uniformly equal at all audio frequencies. But a linear-phase system does not have uniformly equal phase shift at all frequencies (except in the special case when the phase shift is zero at all frequencies); rather, a linear-phase system is one in which the phase-shift is linearly proportional to frequency, so that when the frequency is doubled the phase shift also doubles.

The way to make sense out of this language is to remember that "phase"' is an engineer's way to describe the timing of the signal. Linear phase response means uniform time delay, and this is the clue to its audible significance for the music listener. A system with nonlinear phase shift alters the relative timing of the various frequency components in a signal, causing some frequencies to arrive at the ear earlier or later than other frequencies. Obviously if the signal frequencies are being dispersed in time, the signal is being altered. (Noting that THD, IM and other conventional distortions arise from amplitude nonlinearities in a circuit or system, D. Preis at Harvard has proposed that time-shift be defined as a "linear distortion. ") In a linear-phase system all of the frequencies in a transient signal emerge from the system at the same time, so the system may be described as "time-aligned*."

The terms "time-dispersing" and "time-aligned" are more useful ways of describing the property we wish to explore than are "nonlinear phase" and "linear phase. " The latter are misleading and inadequate. A time-based language gives an intuitively clear picture of what is physically happening; furthermore, as we shall see, in every case where "phase" turns out to be audibly significant in music reproduction, it is explicitly the time shift rather than the phase to which the ear is responding.

What, then, do we hear? Perhaps the best place to start is by examining the scientific evidence. There are three aspects of sound reproduction which have been claimed to be affected by phase or time shift: tone quality, transients and stereo imagery.

The Effect of Time Dispersion on Subjective Tone Quality

In recent years dozens of controlled experiments have been performed by researchers aimed at defining the phase sensitivity of the human ear. Typically these experiments are conducted with electrostatic headphones, or with loudspeakers in an anechoic chamber, in order to ensure

*N.B. "Time-Align," with capital letters, is a proprietary trademark of E. M. Long Associates.

that the cleanest possible signals are presented to the ear with the least interfering noise and the fewest extraneous factors to confuse the interpretation of the results.

An oft-repeated class of experiments involves the use of an "all-pass" filter, a circuit which produces phase shift in a signal but does not alter the frequency response. Using various complex continuous tones (either musical sounds or square waves), and with amounts of phase shift similar to or far exceeding that found in normal audio components, listeners have consistently reported that the all-pass filter produces no perceptible change in the quality of the tone. Mark Davis at M.I.T. has performed an extreme example of this experiment in public demonstrations to audiophiles: a square-wave signal, rich in harmonics spanning the audio spectrum, is sent through a General Radio 1925A 1/3-octave filter set consisting of thirty narrow-band filters wired in parallel covering the audible frequency range. Each narrow filter causes drastic phase shifts because of its sharply limited bandwidth (phase shifts much steeper than any conventional audio component would yield). In this test the GR filter set functions as an all-pass filter, so that the many harmonics of the square wave are preserved in amplitude but scrambled in timing, yielding a severely distorted waveform. In A/B comparisons between the undistorted square wave and the phase-scrambled version, listening audiophiles describe the difference in sound as either barely perceptible or entirely nonexistent.

Another dramatic experiment, which can be performed with simple equipment, involves playing a square wave and periodically phase-shifting the third harmonic by 180°. (A square wave contains a fundamental frequency with all odd-numbered harmonics; a 500 Hz square wave contains a third harmonic at 1500 Hz, a fifth harmonic at 2500 Hz, etc. Thus the 1500 Hz component would be alternately reversed in phase.) This drastic phase shift produces a radical alteration of the waveform, yet listeners find it extremely difficult to identify any change in the sound.

Many independent researchers have arrived at the same conclusion: in continuous tones, waveform fidelity is not relevant to human hearing. Physiologically, the ear does not operate as a waveform detector; amplitude information and timing (phase) information are detected separately in the ear.

Here is an experiment which has confused some audiophiles, because it seems at first to be similar but yields startlingly contrary results. Begin with a continuous tone at 1000 Hz, close to C on the musical scale, and let it be a square wave or distorted sine wave so that it has a harmonic overtone at 3000 Hz (and perhaps other harmonics as well). Add a second tone a musical fifth higher in pitch, at 1500 Hz (close to G on the scale), and let it also be a distorted sine wave with harmonics at 3000 Hz, 4500 Hz, etc. Now phase-shift the second tone at its source. The result is that a clear difference in the quality of the combined tone is heard. One could be tempted to conclude from this that the ear is sensitive to the phase of harmonically related frequencies. In fact, what the ear is responding to is <u>amplitude</u> -- the varying level of the 3000 Hz component. As the two tones vary in phase their 3000 Hz overtones mutually reinforce or cancel each other. As Tim Holl (Chief Engineer at Acoustic Research) has pointed out, the ear seems to be sensitive to the phase of distorted signals because the phase affects the amplitude of the distortion products. To put it simply, phase affects the audibility of the distortion itself. However this occurs only if the component tones are phase-shifted prior to being combined. In the more common case of an already distorted complex sound being played through a phase-shifting medium such as a loudspeaker, the phase shift has no effect on the audibility of the distortion.

Scientists eventually managed to find certain specialized continuous signals to which the ear genuinely appears to be phase-sensitive. The phenomenon requires complex tones containing closely spaced frequencies. For instance, if the ear is presented with 950 Hz, 1000 Hz, and 1050 Hz simultaneously, and if the phase of the 1000 Hz component is changed by as little as 30 degrees, the ear hears the shift as a change in the texture of this highly dissonant composite tone. Evidently the prime reason is that when the ear detects the loudness level of a tone, all sound in a "critical band" about 1/3 octave wide contributes to the sensed level. Changing the phase of the 1000 Hz component alters the composite "envelope" of the three-frequency complex and thus alters the subjective sensation. Even so, these effects are reported to be fairly delicate and can be masked by normal amounts of room reverberation, background noise or the presence of other frequencies in the sound. And these tests are hardly relevant to music listening. They require absurdly steep phase changes within small frequency intervals. And most musical sounds involve a variety of frequencies which (a) mask subtle phase effects and (b) are harmonically related, thus spaced more than 1/3 octave apart.

As a practical matter, then, we are left with the conclusion Helmholtz announced nearly a century ago: the subjective quality of a continuous tone depends only on its harmonic (i.e. , frequency) content and is unaffected by phase shifts. And since "phase, " properly speaking, is a concept that has meaning only for continuous tones, then the old question "Is phase audible?" must be answered "No."

The Effect of Time Dispersion on Transients

Helmholtz has sometimes been blamed for the widespread neglect of the importance of timing (phase) effects in sound reproduction, but in fact he was careful not to apply his conclusion to the beginnings and endings of sounds.

The requirements for good transient response are fairly easy to assess if we ask the right question. "Is phase shift audible on transients?" is a question which has led some writers to propose sophisticated and difficult scientific inquiries -- difficult partly because of the problem of devising an appropriate test signal, one sudden enough to qualify as a transient and still sufficiently repeatable that the phases of its frequency components could be reliably determined and manipulated.

But when we ask the proper question, "Is time shift audible on transients?" one answer is obvious. If the fundamental of a tone starts now and the overtones start later, we hear the change. If the bass frequencies of a Beethoven symphony emerge from the loudspeaker now, and the higher frequencies emerge after delays of several seconds or minutes, the music is distorted beyond recognition. So the question is not "Is it audible?" but simply "How much?" What is the threshold of audibility for time shift?

A simply way to find out is to create the sharpest, shortest transient sound possible -- an instantaneous pulse lasting for only a few millionths of a second. Alternate it with a second signal composed of two simultaneous smaller pulses whose combined strength equals the intensity of the previous pulse. The single pulse and the equally intense double pulse sound identical -- indeed they are identical until they are examined with millionth-of-a-second resolution. Now start spreading the two smaller pulses apart in time, and A/B their sound against that of the single pulse. When this is done it is found that they can be separated by as much as 0.001 second (one millisecond) and still sound identical to the single pulse. With larger time separations the character of the sound begins to change; the "tick" becomes a "thud." When the delay between the two pulses becomes large (thirty to sixty milliseconds, depending on the listener), the ear begins to distinguish the two separate pulses rather than hearing them as a single broadened pulse.

With the sharpest and shortest transient sounds generated by laboratory equipment, the audibility threshold for time smear is about one millisecond. With less demanding test signals (including spoken voice and most music) the threshold increases to several milliseconds. In the early 1930s, telephone engineers found that when voice frequencies were dispersed in time by ten to fifteen milliseconds (because of accumulated phase shifts in long-distance lines), speech began to become garbled; so time-shift compensation became standard in telephone systems to hold time smear to less than five milliseconds.

In 1935 motion-picture sound engineer John K. Hilliard found that recorded tap dancing had a flasely resonant quality when reproduced via the finest laboratory reference loudspeaker of the era. This giant two-way speaker system had a folded-horn woofer with an air column length of eleven feet, plus a horn tweeter three feet in length. MGM recording director Douglas Shearer, brother of actress Norma Shearer, suggested that the time-shift caused by the woofer's extra eight-foot depth might cause the echo. Realignment of the woofer and tweeter yielded accurate reproduction of the transients in Eleanor Powell's tap dancing.

Listening tests with a variety of music and special sound effects have indicated that a two millisecond delay between woofer and tweeter cannot be detected, and that limit was adopted in 1938 as the maximum permissible time spread in theater speakers. In recent years numerous experiments have confirmed that delays of one to two milliseconds between woofer and tweeter produce no clearly identifiable effect in monophonic sound reproduction. (In conventional two-way and three-way dynamic loudspeaker systems the maximum woofer/tweeter delay is about one millisecond. In a few large horn systems the woofer is two to three milliseconds behind the midrange and tweeter. Of course, in a bi-amplified system in which the woofer, midrange, and tweeter are mounted in separate cabinets in different locations in the room, larger amounts of time smear can occur.)

When a large amount of time dispersion is present in a system, its audible effect on transients is quite dramatic. For example, Mark Davis has broadcast a demonstration using the same G.R. 1925A filter set which earlier showed that gross waveform distortion of a continuous square wave is inaudible. When sharp pulses (containing many audio frequencies) are fed through the filter, their "tick" sound becomes a "thunk;" one actually can hear a rapidly descending frequency sweep caused by the time dispersion of the thirty narrow filters, the high frequencies emerging quickly and the low frequencies many milliseconds later. As for musical sound, synthesizer engineer Dennis Colin conducted a demonstration for the Boston Audio Society in which a piano recording was played through a Magnepan speaker. At intervals an all-pass filter with a steep phase shift at about 600 Hz was switched into the circuit. The sustained portions of the notes were unaffected, but the attack at the beginning of each note was strikingly softened and muddied. Subjectively the sound was described as "veiled."

Of course, these experiments involved steep phase shifts and-consequently large time dispersions. Because the maximum time dispersion of a conventional loudspeaker is typically about one millisecond, the subjective effect on the reproduction of transients would be expected to be at the threshold of audibility. An interesting opportunity to confirm this was provided this year by the "live versus recorded" demonstrations Acoustic Research conducted for audiences at the Consumer Electronics Show and for the Boston Audio Society. Under the technical supervision of Victor Campos, jazz-like improvisations were anechoically recorded by free-lance percussionist Neil Grover; these were then played back through AR-10π speakers in a normal-sized listening room and subjected to an A/B comparison with the live sound of the percussion instruments. The percussion set included the full standard array of drums and cymbals, played with considerable vigor, and it was found during the preliminary trials that some of the transients strained the capabilities of the finest recording equipment and the largest power amplifiers. Audiophiles who heard these live-versus-recorded comparisons report that most of the time it was-impossible to tell when Grover was playing and when he was faking it while the speakers played. In some of these demonstrations a few differences have been audible, notably in the tonality of the snare drum, attributable in part to performer fatigue and (as Grover showed) to incidental factors such as the choice of drumsticks or a change in humidity between the recording date and the playback date. The striking fact is that, although the AR-107 is not a conspicuously linear-phase speaker, the attack transients of the percussion instruments were accurately reproduced with no apparent softening or veiling, not even in the case of the brilliant bell-like cymbal included in the set.

The conclusion that time smear of one millisecond or less has no audible effect on the reproduction of musical transients is widely accepted by psychoacousticians and speaker designers. But some designers believe otherwise, and credit for this probably belongs to V. Hansen and E. R. Madsen of Bang & Olufsen in Denmark, who conducted a group of experiments about five years ago which suggested that substantially smaller time shifts can be heard. The key to this result is the discovery of a special test signal. It consists of an asymmetric sine wave (i.e., a sine wave with a DC offset) with every second cycle switched off. Note that as the sine wave is switched off and on, its DC offset is also switched off and on. The result can be regarded either as a peculiar continuous signal or as a rapidly repeated single-cycle pulse transient. If the sine-wave frequency is 1000 Hz, then each second of the test contains 500 one-millisecond segments of offset sine wave alternating with 500 one-millisecond gaps. Mathematical Fourier analysis showed this signal to be equivalent to three continuous sine waves at 500 Hz, 1000 Hz, and 1500 Hz misaligned in phase, with their phase angles proportional to the size of the DC offset. When listeners heard this signal they found that as the DC offset was increased they could hear a change in the timbre of the complex tone. By finding the threshold of audibility of the DC offset and calculating the corresponding phase angle, Hansen and Madsen concluded that the ear could hear phase shifts of less than ten degrees at 1000 Hz, corresponding to time shifts on the order of thirty millionths of a second. With higher-frequency test signals the threshold time shift was about the same, while at lower frequencies the permissible time shift increased rapidly.

This experiment has not been repeated or verified by other researchers, and it has been criticized by skeptics for its methodology, its interpretation and its use of a peculiar test signal unlike anything encountered in music or acoustics. But it has also been widely influential, its conclusions have been accepted by various designers, and it appears to have been one of the principal stimuli for the recent influx of linear phase loudspeaker designs.

The Effect of Time Dispersion on Stereo Imagery: A Hypothesis

In examining the psychoacoustic evidence regarding the effects of time dispersion on perceived tonal quality and on transient response, we have reached essentially negative conclusions. With continuous tones waveform fidelity appears to be irrelevant, even with square waves. The phase sensitivity of the ear is subtle at best and can be demonstrated only with very specialized test signals having little relevance to music. As for the reproduction of transients, only one controversial report contradicts the general view that the threshold for audibility of time smear is one to two milliseconds, which happens to be the maximum time dispersion found in most loudspeakers. So time dispersion can be viewed as a relatively minor problem among the many ills to which loudspeakers are heir.

However this conclusion is essentially applicable to monophonic reproduction. When we examine stereo reproduction time dispersion is not longer a trivial factor; it becomes a major issue. Everyone is familiar with the fact that when the phase of one of the loudspeakers in a stereo pair is shifted by 180 degrees (by reversing the leads to one speaker), stereo localization becomes completely vague and diffuse. Clearly the relative phase shift between the two speakers of a stereo pair must be tightly controlled if precise stereo localization is to be obtained.

Stereo localization by the human ear depends essentially on two factors: the relative intensity of signals arriving at the two ears and the relative timing of those signals. An imbalance of 0.5 dB in level produces a perceptible image shift. And the dependence on relative timing is amazingly critical. According to N. V. Franssen of Philips, trained listeners have detected image shifts caused by an interaural time difference as small as one microsecond under laboratory conditions. A more typical value for the average listener is thirty microseconds, corresponding to a spatial angular resolution of about three degrees of arc. So, although we noted in the previous section that it may be permissible for a loudspeaker to have a frequency-dependent time shift of up to one millisecond, the two loudspeakers in a stereo pair must be identical in time-shift to within a tolerance approximately thirty times smaller. If the stereo loudspeakers differ in their time-shift behavior by more than about thirty millionths of a second (or a finer tolerance, perhaps, for critical listeners), the stereo image will be perceptibly smeared. The two speakers must "speak" together at all frequencies if the subtlest details in the stereo field are to be preserved.

This, quite simply, may be the principal advantage to be gained from "linear-phase" or "time-corrected" loudspeakers. The manufacturers who are striving to reduce the time dispersion of loudspeakers to zero may also be ensuring that there will be no significant differences in signal propagation timing between the two speakers in a stereo pair. The delicate timing information in a stereo recording is thus accurately retained and is transmitted to the listener unaltered.

If this conclusion is accepted, then a useful corrollary may immediately be drawn. It is well known that stereo localization is primarily a mid-frequency and high-frequency phenomenon. The ear does not localize low-frequency sounds well. When something like a bass drum is heard, the ear uses the higher-frequency components in its attack transient to localize it. But when a pure low-frequency sound is heard (e.g., a 50 Hz tone from a sine-wave oscillator), its sound is nearly impossible to localize. Therefore it follows that in loudspeakers there is little advantage to be gained from eliminating time dispersion at low frequencies. If the woofer crossover is reasonably low in frequency (e.g., around 500 Hz or lower), essentially the full benefit of time-corrected operation can be realized by designing just the midrange and treble units for minimum time dispersion. This proposition was confirmed recently by Henning Moller of B & K, a leading proponent of linear-phase design. Speaking to the Boston Audio Society this year, Moller noted that in experimenting with staggered-driver speaker systems he found that midrange/tweeter alignment dramatically affects the sound but changes in woofer alignment are very hard to hear. So, as a practical matter, a manufacturer can produce a time-aligned loudspeaker without having to make it look like a pregnant kangaroo. Conventional cabinets can be used.

What Causes Time Dispersion?

If we are going to suggest that time-dispersion differences in loudspeakers should be minimized, it may be useful to review some of the sources of time shift in the reproduction process.

1. Departures from flat frequency response. In general, nearly every departure from flat response, in any audio component, has an associated time shift. Examples include the rolloffs at the low-frequency and high-frequency limits of microphones, tape recorders, phono pickups, tuners, amplifiers, and speakers, as well as all frequency-response alterations with tone controls, filters, and equalizers (including those heavily used in recording studios). To the extent that these are identical in both stereo channels, they are not expected to degrade the stereo imagery. In general, if a frequency response change in one component is compensated by equalization in another component, the time shift is also compensated.

2. Resonances. Any sharp resonance will yield substantial time-shifts within the octave band above and below the resonance frequency. Examples to consider include the fundamental woofer resonance, the midrange or tweeter resonance (which in many systems is less than an octave below the crossover frequency), the midrange resonances associated with woofer cone breakup, the resonant filter commonly used to suppress the 19 kHz FM stereo pilot tone in FM tuners and Dolby encoders, and the resonant circuit used for high-frequency equalization in tape recorders.

3. Crossovers. Loudspeaker crossovers are rolloff filters, so they cause time shifts in the region around the crossover frequency typically amounting to several hundred microseconds. Some manufacturers do not attempt to control the crossover tolerances very precisely; since the drivers put out substantial amounts of energy for a full octave on either side of the nominal crossover frequency, a 20% or 30% production variation in crossover frequency will have little effect on the response curve of the speaker. So it is not uncommon to find that two successive speakers off a production line may have crossover frequencies of 1500 Hz and 2000 Hz respectively. The resulting time-shifts will differ by about 100 microseconds in the crossover region, and to this may be added the differences in time shift produced by the typical 10% to 20% production tolerance on tweeter resonance frequency. If these two speakers are used as a stereo pair, ideal stereo imagery cannot be expected. On the other hand, a manufacturer who exerts tight control over crossover tolerances and driver resonance frequencies may be expected to produce pairs of speakers having unusually precise stereo imagery.

4. Axial driver geometry. This is the aspect of loudspeaker time compensation that has received the most conspicuous attention. The woofer's sound emerges behind that of the tweeter because of the depth of the woofer cone. Typically the time shift is a few hundred microseconds, comparable to that caused by the crossover. It is the desire to compensate for this woofer time offset that has led to the proliferation of speakers with tilted front panels and other cabinet shapes intended to make the effective acoustic positions of the woofer, midrange and tweeter equidistant from the listener. Incidentally, a bi-amplified system can be time-corrected using electronic delay circuits without requiring physical realignment of the woofer. Of course, if the woofer is flat rather than conical, it will have little geometrical time shift anyway; similarly, flat-panel radiators such as full-range electrostatics and planar magnetics have little woofer offset, though they may still require crossover correction.

5. Lateral driver geometry. If the drivers are side by side in the cabinet, then the images of voices or instruments usually are broadened in proportion to the spacing of the drivers. So the sharpest stereo image usually is obtained with the drivers aligned vertically in the cabinet rather than horizontally. (As noted earlier, it is the alignment of midrange and tweeter that counts most, as low frequencies contribute little to stereo localization.)

6. Multiple sources. A loudspeaker in which two or more laterally spaced drivers operate in a frequency range may not produce as sharp and detailed a stereo image as a system employing only one driver for each range. The system "speaks" twice with every input transient. The same problem occurs when a strong reflected image is produced by placing a single driver close to a reflecting surface.

Some conclusions are obvious from this listing. One is that a loudspeaker with staggered drivers in a funny-looking cabinet may not in fact be accurately time aligned. Elimination of time-shift effects requires close control over (and compensation for) crossover circuits, driver resonances, and lateral driver geometry. A second conclusion is that since these factors yield time-shift differences at least as large as (or larger than) the time offset due to woofer depth, and since low-frequency time shift is relatively unimportant anyway, it is likely that there are some conventional rectangular-box loudspeakers which are more accurately aligned in time than some staggered-driver systems. Support for this view is found in reports that certain conventional-looking speakers (for example the Cizek Model One and the KEF Model 104ab) produce stereo images of exceptional depth and resolution. It may be significant that Cizek employs a 2% tolerance on crossover parts.

Incidentally, a few claims have been made to the effect that elimination of loudspeaker time dispersion not only affects stereo imagery but also improves the monophonic sound of a speaker. The quoted evidence is the fact that when the axial alignment of drivers is varied, the system sound is noticeably changed. But such changes are equally attributable to other parameters which change when axial driver alignment is altered: the radiation pattern of the tweeter, reflections off cabinet surfaces, cancellations caused by interference between drivers, etc.

<u>Is It Worth the Bother?</u>

As with many issues in sound reproduction, it's up to you to decide whether the reduction of time dispersion in loudspeakers is worthwhile. If you try to do listening comparisons you will discover the unglamorous fact that a speaker's sound still depends mainly on its frequency response and its angular dispersion; by comparison time dispersion is a relatively minor problem. However, if you compare speakers having similarly good frequency response and angular dispersion characteristics, then you may find that loudspeakers can be grouped into four classes:

1. those in which no particular attention has been paid to time dispersion;

2. those in which the drivers have been staggered but in which no attention has been paid to other sources of time dispersion, such as crossover circuits;

3. those in which the drivers are not staggered but which nevertheless deliver precise stereo images thanks to careful control of crossovers and other details;

4. those which are fully time-corrected from top to bottom.

You will have to depend mainly on your ears, at least until standard lab tests for loudspeaker time smear (and for sample-to-sample uniformity in time behavior) are generally adopted.

To end this discussion, the principal advantages and disadvantages of time-corrected loudspeakers can be summarized.

<u>Advantages</u>

<u>1.</u> <u>Depth</u>. This may surprise some listeners when they first hear it, since many speakers (and records) elicit only a general left-to-right spread. But "stereo, " as originally conceived, implied a three-dimensional sound in which voices or instruments could be localized at different apparent distances from the listener as well as at various lateral positions. Listeners to time-aligned speakers consistently report hearing a stereo image with unusual depth.

<u>2.</u> <u>Resolution</u>. The stereo image is reproduced precisely, each voice or instrument having its proper place and width. In complex sound sources such as symphony orchestra, individual instruments can be resolved with unexpected clarity. In the old cliche, "I hear details I never knew were in the recording. " Some listeners have incorrectly attributed the improved resolution of detail to more accurate transient response, but the better definition of details is simply the result of the reduction of blending in the stereo image.

<u>3.</u> <u>Separation of ambience</u>. With loudspeakers whose stereo image is slightly blended because of time-smear, any hall ambience or reverberation in the recording tends to become slightly mixed with the instrumental sounds, causing coloration of those sounds. Consequently, with such speakers closely-microphoned recordings tend to sound better because of their distinctly

defined sound. But with time-corrected loudspeakers, the ambience is resolved as a separate sound, and larger amounts of hall ambience in recordings can be enjoyed.

Disadvantages

1. Restriction of listening position. In order to get the maximum benefit of a time-aligned system listeners must be located close to the stereo axis equidistant from both speakers. With some systems an optimum height is also specified. In the KEF 105, for instance, an alignment light is built-in; you and the speaker are optimally aligned only when you can see the light in the speaker.

2. Exaggerated depth. For a typical listener in a concert hall, the instruments on stage are at only slightly different distances. But when recording microphones are placed near the front of the orchestra, the relative distances of the instruments at the back of the orchestra are exaggerated, as is the relative prominence of various details such as the scrape of the bow and the clicking of oboe valves. Loudspeakers which reproduce this perspective may be accurate, but they do not provide a realistic representation of the sound of an orchestra in a concert hall. Thus many listeners will prefer loudspeakers whose time smear flattens the exaggerated depth.

3. Poor recordings ruthlessly exposed. A time-aligned speaker may be too analytical, revealing that many recordings do not contain a genuine stereo image. Instead, every instrument is close-miked, sounding as if it were recorded in a closet, and artificial reverb is then added. As designer Richard Sequerra noted in an interview, the majority of recordings actually sound more pleasant and more realistic with a less analytical speaker, especially a multi-directional speaker which adds its own blended spaciousness to the sound. (Of course, the overly dry sound of many recordings, as heard through time-corrected speakers, may be improved with the aid of a time-delay ambience-synthesis system, adding spaciousness via secondary speakers while the main speakers retain their precise, detailed stereo imagery.) -- Peter Mitchell

Bob Berkowitz

Bob Berkowitz of Acoustic Research observed that, though there can be little doubt that phase shift is an audible phenomenon, studies have shown that, even with carefully constructed test signals, subjects must be trained for months in order to detect a two millisecond time reversal. Other studies have arrived at figures of from two to five milliseconds. Berkowitz also reported that he and others at Acoustic Research had duplicated Hansen and Madsen's experiment and found many "objective and subjective discrepancies" which they are preparing to present in a letter to the AES. He asserted that Hansen and Madsen's quantitative results were in error by a factor of six, apparently because of a miscalculation in the phase shift of the test signal. Nor, he pointed out, do they separate the phase shift from an admittedly audible phenomena, polarity reversal, in their tests.

Berkowitz also reported that recent tests conducted at AR have also revealed that most front-firing loudspeakers have time delays of less than one millisecond in their direct radiation, with occasionally longer delays at specific frequencies because of reflections from cabinet edges, hardware, etc. And, he added, assuming one could hear such delays in an anechoic environment, even these effects would be rendered inaudible once the speaker were placed in a reverberant area, i.e., a listening room. His conclusion: most good speakers on the market today exhibit time delay effects well below the threshold of our ability to hear them.

Mark Davis

Mark Davis described a test he designed to determine the factors which influence imaging in loudspeakers. The test was conducted with three loudspeakers arranged in a stereo (with center channel) array. A "real" center image (a signal sent to the center speaker only) was compared with a "phantom" center image (identical signals sent to the flanking speakers only). In an ordinary listening room it was easy for listeners to distinguish between the two "images, " but when the same test was conducted in an anechoic chamber, it became very difficult, if not impossible, to tell them apart.

Davis also pointed out that human hearing cannot resolve signal phase above 1500 Hz, as the nerves of the inner ear require a millisecond between discharges and thus can only resolve an envelope of waveforms, rather than individual waveforms, as high frequencies.

Davis concluded by criticizing most of the psycho-physical tests concerning phase for failure to adequately control all relevant variables and cautioned that it is all too common for experimenters to conclude that effects they hear are phase effects when in fact other variables are responsible.

-- Richard Glidewell

[Mark will present a more detailed analysis in next month's Speaker. -- Ed.]

A Publication of the BAS

A Subject Guide to The Bas Speaker - Volume 4

John E. Gombos

This is a subject guide to The BAS Speaker for Volume 4: October 1975 through September 1976, using the system first described in my March 1976 article. Before I present the guide, I must list some limitations. These are due to the structure of the filing system.

1. Since this is a subject guide, in some cases article titles are not listed -- only the subject. In addition, space restrictions caused me to abbreviate a few titles. (These actions are not meant as criticism of the authors.)

2. Filing is partly subjective, so others will disagree with some of my classifications. Also my determination of category is sometimes based on a quick skimming of an article. This may have caused a misunderstanding on my part leading to a misclassification.

Special Instructions

1. Page numbers were assigned consecutively from the first page of each month's issue to the last page of the final publication of that issue.

2. In some cases, references are made to both the beginning and the middle of an article. This occurs when the article covers more than one subject.

3. The articles in each category are presented in subdivision order with the subdivision indicated on the left. I have used the following abbreviations to indicate subdivisions:

AC - Accessories
CM - Commercial
CN - Construction
DS - Design
IN - Installation
MN - Maintenance
OP - Operation
RP - Repair
SP - Specifications
TS - Testing
No abbreviation - General article

a

HI FI/STEREO GENERAL

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MICROPHONES

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