THE B.A.S. SPEAKER

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In This Issue

Hi-fi seems to be almost as complicated as love. It's as hard to keep head and ear in line in the one as head and heart in the other. If you consider carefully the articles, long and short, in this issue, you will find enough contradictions in facts and feelings to fuel a month-long nonstop debate. We have several items on equalization: a feature review of a Shure equalization test device and communications looking for suitable equalization standards and questioning the value of any form of equalization.

Another feature article reviews the Audio/Pulse Model One and Dave Ranada tells us all about pitch. Henning Moller of B&K offers a provocative viewpoint on what are the really important measurements in evaluating an audio system (in the meeting summary along with the Genesis speaker presentation). Then there's Victor Campos, who could form a one-man debating team. He offers a spirited defense of the SAE 5000 and comments on the "wire with gain" concept. Try squaring that point of view with that of our member who listens through three preamps, not to mention the equalization issues already mentioned.

We think everyone, especially our out-of-Massachusetts members, should read "The Shop Talk Bicycle." No, it's not an improved form of locomotion, but we think it's a fine way of spreading the "Shop Talk" word to the hinterlands and not at all controversial. See if you don't agree.

Finally, we call attention to "In the Literature," not only for its capsule review of where the good articles are, but also for an important announcement about its immediate future. Once again, The Speaker needs your help.

And with that, we invite you to touch gloves and come out reading.

-- Henry Belot

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

*Schoeps CMT-50 series condenser microphones, standard 48V phantom powering, current consumption approximately 0.7 mA, standard XLR connectors; one pair CMT-55U two-pattern (omni, cardioid) with switchable 10 dB attenuators, asking $700; one pair CMT-56U three-pattern (omni, bidirectional, cardioid), asking $800. The capsules are all of recent manufacture and correspond to the latest series of "Colette" capsules made by Schoeps. The two-pattern capsules have consecutive serial numbers, and the three-pattern capsules are a specially selected and matched pair. David Satz, 48 R.C. Kelley Street, Cambridge, MA 02138.
*Son of Ampzilla with transferrable warranty, mint condition, $340. Trevor Lees preamp, professionally wired on new PAS-3X chassis, used 4 to 6 hours for A-B comparisons. Includes Telefunken tubes and Trevor Lees faceplate, $275. Eugene Constant, (408) 373-4491.
*Advent 100A Dolby outboard, low-noise power supply, walnut case, $160; Pioneer SA-9100 integrated amp, 60 Watts/channel, still in warranty, $265; Dual 1218 with Shure V15-II, $80. Peter Bronk, (617) 734-9057, evenings.
*Magnepan MG11's, black. Better than new, correctly broken in, at full efficiency with proper imaging. All packing intact, warranties blank. Will demo. $550 includes delivery and set-up in E. Mass. William Juch, Dept. of Philosophy, Brandeis University, Waltham, MA 02154 or (617) 647-2654 and leave message. Wanted: Quad ELS's.
*New dbx 122, $200, new 117, $130, 50 TDK-SA C90 @ $3 ea. (in boxes), plus freight. Ray Auclair, 132 Bennett Street, Woonsocket, RI 02895, (401) 765-1093 or 769-4400.
*Dyna-Quad adaptor box, Pioneer stereo display SD1000, Levinson JC-2 preamp with moving-coil and regular cards and oak case, Audio General preamp, AR-11 speakers, Marantz 250 amplifier. Selling to best offers. (617) 492-1043 or (617) 868-4191.
*Advent 202 cassette playback with Dolby, excellent condition, $50. Call (201) 731-5925 after 4:30 and ask for Robert.

Elections

The BAS Treasurer's position will open up this year at the September Business Meeting. Those interested in running for the job should know that your name should be published in the August issue of <u>The Speaker</u> (consult your Constitution and By Laws). To make this issue, we must have your announcement by mid-July at the latest. You therefore have less than two months to decide to run.

Anyone interested in discussing the duties of treasurer, please call Harry Zwicker at 862-5500, ext. 5811 (days).

-- H. Zwicker (Massachusetts)

(We can't run the BAS without officers, so please don't be shy. -- Ed.)

The "Shop Talk" Bicycle

For those members who have not heard about it, "Shop Talk" is an informal, 90-minute radio program about audio matters, broadcast every Saturday morning, 9:30 to 11, on Boston University's WBUR-FM. The program is hosted by BAS members Peter Mitchell and Richard Goldwater. About half the shows feature a live or recorded conversation with an audio engineer or manufacturer, and many include question-and-answer sessions with phone calls from the audience. In the last several months interviews have included Julian Hirsch, Jake Rabinow of Rabco, Peter Pritchard of ADC and Sonus, Bruce Maier (Discwasher), Roy Allison, Oscar Heil, Dunlap and Clarke, Tom Holman (Advent), Rene Jeager (dbx), Mark Davis of MIT (the show that <u>Audio Critic</u> lambasted), Bob Carver of Phase Linear, producer Andrew Kazdin (Columbia Records), and a moving exploration of music and human psychology with conductor Colin Davis. The. program is non-commercial, so participants are free to criticize and recommend products by name. Needless to say, Boston-area audiophiles reserve Saturday mornings for "Shop Talk.

In fact, the fame of "Shop Talk" has spread to other cities and aroused the curiosity of audio hobbyists who would like to hear the show. One of these would-be listeners is Dean Slindee of La Crosse, Wisconsin. He has hatched a plan for "sharing the wealth" with those not fortunate enough to live within range of WBUR's signal. The plan is very similar to a method of film and broadcast distribution known as "bicycling." When a film or program is bicycled, the distributor makes a small number of original copies and sends them not just to a single user, but to lists of users. The first customer on the list uses the tape or film and then mails it, at his own expense, to the second name on the list. That customer mails the show to the third name on the list, and so on until the program comes full circle and returns to the distributor.

Anyway, Slindee's proposal intrigued Peter Mitchell, who is uniquely qualified to serve as the "distributor." So here in detail is Dean Slindee's plan for making "Shop Talk" available to interested BAS members, incorporating a few modifications suggested by Peter:

1. Interested members write to Dean Slindee, who will serve as coordinator for the project.

2. If there is sufficient interest to establish the service, each member of the group subscribes $25, or about $.50 a show. Peter estimates that this fee would be sufficient to cover the ongoing expenses he would incur in producing the dubs and distributing them: labor, machine wear, postage and occasional purchases of fresh cassettes as tapes wear out or are lost in transit.

3. Each member of a bicycle group would receive a list of member names, addresses and phone numbers, so he could keep in touch with the members before and after him in the circulation chain.

4. Each week Peter would start a new program on the bicycle. Sometimes "Shop Talk" is devoted to a topic of only local interest. On such occasions, Peter would draw upon his stock of permanent file copies of past shows to circulate a program of general interest. If you reread the first paragraph, you'll see what treasures that file contains.

5. Each member would be permitted to hold a tape one week before mailing it on to the next member. Peter proposes to make enough dubs to keep the total circulation time down to ten to twelve weeks. For example, if 32 people were to subscribe, they would be sorted into three lists, and Peter would provide three copies of each program.

All of this is conditional on sufficient interest from the membership. Although Peter Mitchell has volunteered to do the dubbing, the coordinator for this project is Dean Slindee. If you're interested, write to him, not Peter. The address is: Dean A. Slindee, 613 South 8th Street, La Crosse, WI 54601. Let him know the extent of your interest, whether your expected playback format is mono or stereo, whether or not you want Dolby encoding, and give him any suggestions you have regarding the plan.

Incidentally, it may also be possible to provide subscribers with tapes of certain BAS meetings. If that's of interest to you, mention it in your letter to Dean. The Speaker will keep you posted on further developments, but, again, if there are to be further developments, you must write to Dean Slindee.

Ark Records

This is just a reminder that orders are still being taken for Ark Records; for prices and availability, see page 3 of the March issue. (One disc, the 1972 Robbinsdale Band record, 5112-S, is now out of print.)

The first 42 discs have been shipped quickly from Fulton Electronics -- within two weeks after order -- and quite safely by UPS. These have been sent to members via "Book Rate" mail, so let us know if you have not received your order within six weeks after mailing.

Note that the BAS is in no way implying any recommendation for or against these discs; we are simply making them available at a discount. Please send us your listening impressions as soon as possible.

-- H. Zwicker (Massachusetts)

Ark Comments

I have three of the Fulton records: "Organ Music from Westminster," 10251-S, "Choral Music from Westminster," 2123-S, and "Carmina Burana," 4773-S. Both the organ and the choral records are very well done, with good, clean, natural sound and good performances. The "Carmina Burana" recording is, however, just the opposite. The work is too much for the performers, for the conductor and for the engineers. The performance sounds amateurish. Tempos are pulled out of shape by inadequate performers. The recorded sound is primitive, not at all capturing the sound of a real performance in a real hall. Occasionally the chorus may sound natural, but on the whole the record could best be characterized as "strange. " Don't buy it!

Below, I offer my ranking of the extant "Carmina Burana" recordings in terms of both sound and performance:

1. Kegel-Liepzig Chorus and Orchestra, Philips 9500040
2. Previn-LSO/LSO Chorus, HMV-ASD-3117
3. deBurgos -New Philharmonia, Electrola-C-065-00-053
4. Tilson-Thomas-Cleveland Orchestra, Columbia M33172
5. Jochum-German Opera Orchestra, DGG 139-362

All of these are much better than the Ark "Carmina Burana. " -- L. P. McGovern (New York)

For Your Library ...

Here are three product-review books researched and printed in England, which member John M. Tooley is making available to interested BAS members:

<u>Hi-Fi Choice: Cassette Decks & Tapes</u> contains reviews of 55 decks including such names as: Aiwa, Akai, BASF, Dual, Harman-Kardon, Hitachi, JVC, Nakamichi, Neal, Toshiba, Kenwood (Trio), Uher, and Yamaha. Also 50 tapes including Ampex, Capitol, BASF, EMI, Fuji, Hitachi, Maxell, Memorex, Philips, Scotch, Sony, TDK', et al.

<u>Hi-Fi Choice: Receivers</u> covers names such as: Aiwa, Akai, Alpha, Armstrong, Bang & Olufsen, Harman-Kardon, Hitachi, JVC, Luxman, Marantz, Nikko, Pioneer, Revox, Rotel, Sansui, Sanyo, Sonab, Sony, Tandberg, Technics, Kenwood (Trio), Yamaha, and others.

<u>Hi-Fi Choice: Loudspeakers</u> includes reviews of: AR, B&O, Radford, Bose, Bowers & Wilkins (B&W), Celestion, Electrovoice, IMF, JBL, KEF, KLH, Marantz, Ortofon, Philips, Pioneer, Quad, Sansui, Spendor, Tandberg, Tannoy, Technics, Venturi, Videotone, Wharfdale, and Yamaha.

In addition to the product comparisons, all three books include long and interesting introductory chapters. They sell in England for about 1.50 pounds each. Reader Tooley will supply copies for $5 each for the loudspeaker and receiver volumes and for $6 for the cassette volume. The books will be sent directly to the member's address via airmail from England. A recent order for eight copies took two weeks from the time the check was posted to reach these shores.

To order, send your check to: John M. Tooley, RD2 Box 120E, Milton, DE 19968, or for more information, you may phone him weekends or after 5 p. m. weekdays at (302) 684-3443.

And For Your SME .. .

Tooley is also interested in hearing from those who might want to buy the new SME damping adaptor (see Hi-Fi News & Record Review, April 1977, p. 63), which (suspiciously) resembles Bob Graham's device. He will contact you if SME will make a good price on a group purchase. The address is as above.

The Record Sleeve Habit

Anyone attending the recent BAS meeting at which Dr. Bruce Maier discussed the incredible array of hideous weapons nature has amassed in her unending struggle to spoil our records may want to contact the Andrews Nunnery - a uniquely named company that manufactures record sleeves.

As you may recall, Dr. Maier suggested that the vinyl-lined record sleeves that some of us use to protect our precious discs are not of the highest quality. The plasticizers used in the vinyl may actually migrate to the record and harm its surface. Therefore a plain white paper sleeve (without the hole in the middle that lets dust get in) is preferable. But try to find one! I tried all over the Boston area and even contacted Columbia records. They suggested their supplier, the aforementioned Andrews Nunnery. The Nunnery sells the sleeves at $25/1000 plus $3 shipping. They also sell a variety of other sleeves, with and without holes or vinyl.

As you probably have surmised, their minimum order is 1000 sleeves, as they are not a retail outlet. And yes, I did buy them and can now attend to the music in quiet repose. If you want to buy a thousand or more, the address is Andrews Nunnery, 74 Alpha Plaza, Hicksville, NY 11801. If you want to buy a smaller quantity, I have a limited supply which I will make available at BAS meetings or through Box 7 at 3¢ apiece. This offer is good only until I've exhausted the excess supply on hand. -- Elliott Berger (Massachusetts)

Chemistry Lesson

In the November 1976 issue of the Speaker, I noted a confident remark by David Sherwood, which remark represents that Sound Guard bears the generic chemical denomination "trichlorofluoroethane," aka DuPont's "Freon 113."

Well, I contacted DuPont, and I quote their response: "Huh!!? Do you mean 'trichlorotrifluoroethane?' " Well, being a lawyer and not a chemist by trade, I answered artfully: "Huh!!?"

To make a long story longer, with a lot of "could be this's," or "could be that's," or "this is probably it's," I have ended up with a free sample of Miller-Stephenson Chemical Co.'s "MS-122 Fluorocarbon, Release Agent/Dry Lubricant." This 16-ounce aerosol can containing DuPont Freon manifests the legend, "ACTIVE INGREDIENTS: Tetrafluoroethylene Telomer Solids, with the specification, "MIL-L-60326 (MU) AMEND 1 TYPE 1." So, smart guys, since Ball Corp. doesn't find it financially wise to disclose such information on its instructions or labels, which one of you might tell me and the rest of the readership whether I am holding a "true ersatz" Sound Guard?

I await a learned response while my MS-122 teeters between my record collection and the trash masher. -- Richard Benjamin Grant (California)

How to Keep a Discwasher Damp

Those of you who use the Discwasher plush-pile "brush" to keep your discs reasonably free of dust may have cursed the fact that, unlike the old standby Preener, the Discwasher dries out between every use. If you don't care to use their special fluid, you will find that there is a very simple way to keep the thing damp for months at a time. This technique also works very well with the Preener and does not crush down the pile the way the standard Preener tube does.

Obtain a small jar that is large enough to place the Discwasher in and is sized such that you can easily get the thing out when the unit is standing on end in the jar. I have found that a number of jelly jars will work just fine. Now, place something on the bottom of the jar that will make a platform to hold the Discwasher about 1/4" off the bottom of the jar. I used a little piece of plexiglas, but almost anything will do so long as it is waterproof. Fill the jar about 1/8" deep with water. When your Discwasher is slightly damp (as it should be when properly used), place it in the jar with one end on the platform so it is not touching the water. The top of the Discwasher can simply rest against the side of the jar. Screw the lid on tightly so that the-jar is airtight. The best lids are those that open with only a fraction of a turn.

With the lid on the jar and a little water in the bottom the Discwasher will stay moist forever. If you wish, you can use the Discwasher fluid (the water will tend to reduce substantially your use of this fluid). The Discwasher must not be allowed to become more than very slightly damp. Never leave it actually wet, or it will grow moldy. Once every few months you will need to add a bit of water to the bottom of the jar to adjust for evaporation. If tap water is used, the bottom of the jar will gradually accumulate a layer of sediment. This is no problem, as the sediment does not get on the Discwasher, but you will find that distilled water is much better if you want everything to look clean.

-- Roger Sanders (California)

Tone Arm Alignment in Four Easy Steps

Several people have been wondering if their tone arms are adjusted for lowest tracking angle error. Both the articles by Banjamin Bauer (Electronics, March 1945) and book by Edgar Villchur (Handbook of Sound Reproduction, pp., 187-92, 1957) give figures incorrect for modern LP records. Bauer in 1947 was concerned with 12-inch 78's and assumes a 3 3/4-inch inner groove diameter. Villchur in 1957 uses the same figure and does not give the equations from which his constants were derived. The equations may be found in Radiotron Designer's Handbook, pp. 725-26. If 4 3/4 inch is substituted for the inner diameter, these equations give geometrically correct information. Curiously, the AR turntable arms have the correct angle for their length, only the original overhang gauges were wrong.

Offset angle is reasonably critical, and manufacturing tolerances may be more the problem than designers' inability to apply high-school geometry. (At least one tone arm designer, Peter Pritchard, admits to deliberately sacrificing tracking angle accuracy in order to reduce skating force, and told us at a meeting that he might very well do so again. And Jake Rabinow in his talk, said that the tracking angle error advantage of a straight line arm was trivial. -- HB) In any event, the use of an alignment-critical Shibata stylus makes even a well-made arm an approximation. Most cartridge mounting systems have sufficient play in the hardware to allow for perfect set-up using the procedures Mitchell Cotter describes below.

-- Scott Kent (Massachusetts)

Procedure

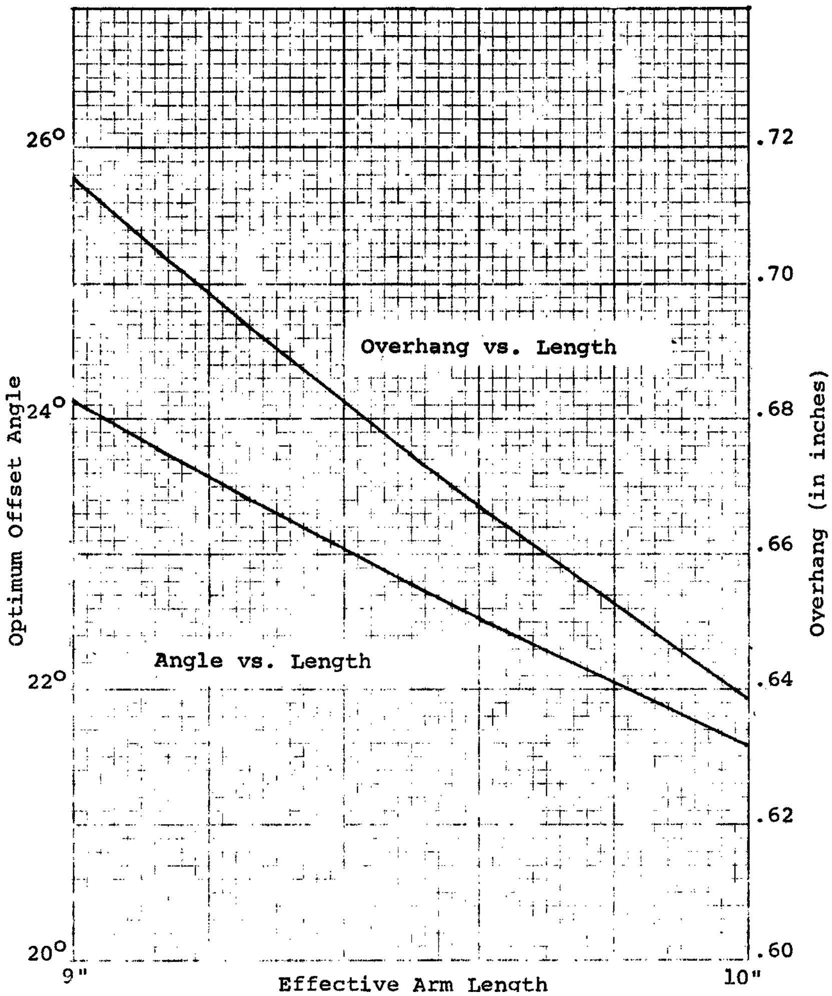
Your tone arm is not aligned properly. If you haven't taken informed measures to align it yourself, ignoring the manufacturer's instructions, that statement is almost certainly true. I have looked at a number of well-respected arms from many manufacturers, and none of them were aligned correctly (as constructed) by the manufacturer. More than that, the instructions provided with the arms didn't yield precise alignment either.

But chances are you can correct the situation easily. Most good arms do provide a way to adjust the overhang, and it's not very hard to twist the cartridge in the headshell to obtain the correct offset angle. And so with an accurate machinist's scale and a five-by-seven-inch index card you can adjust your arm for optimum performance using the simple instructions which follow.

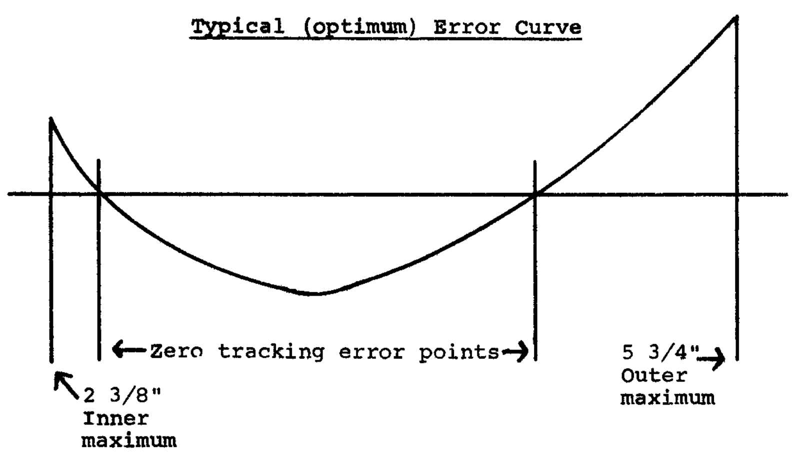
1. Mount the cartridge loosely in the headshell and measure the effective arm length -- the distance between the tip of the stylus and the pivot point. Right now, this measurement will be an approximation, but in the end, it is to your advantage to establish this distance as precisely as possible in order to obtain the proper offset angle.

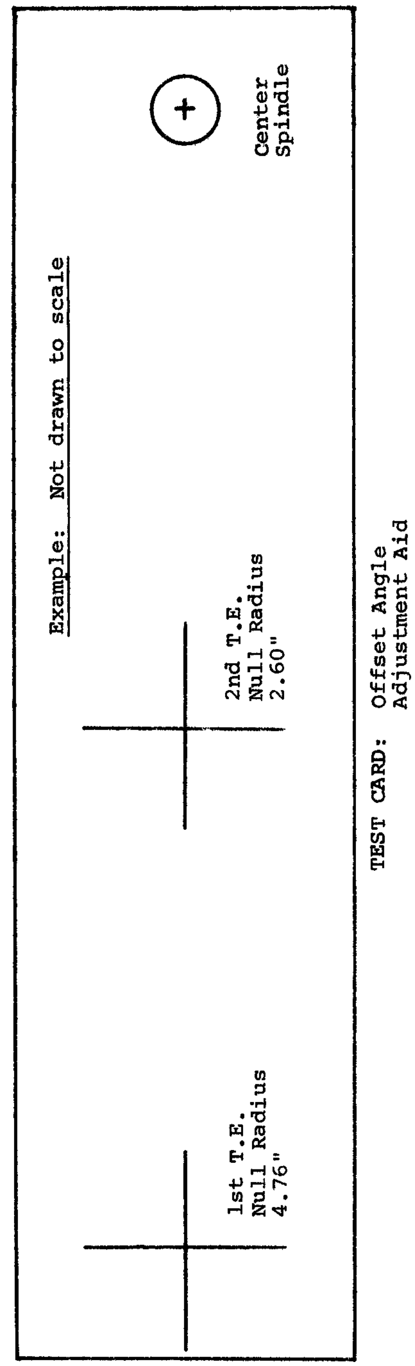
Use a machinist's scale calibrated in tenths or hundredths of inches. Lay it on the turntable, set the stylus on the scale and measure back to the pivot. If the actual pivot point on your arm is difficult to reach, try either dividing the thickness of the mount by two or taking its circumference with tape and dividing by 2π. Add the result to the measured length.

2. Consult the accompanying graph to find the optimum overhang for that length arm. Measure the overhang of your arm by swinging the arm over the spindle and measuring from the stylus



Typical (optimum) Error Curve





to the edge of the spindle. Add .14-inch (half the diameter of the spindle) to the measurement you obtain and refer to the graph to see how closely the value you've obtained matches the value indicated for your length arm. Correct your overhang value if necessary. Because changing the overhang also changes the effective arm length, you will have to repeat this sequence until you have narrowed down the differences after which the overhang and effective arm length will coincide on the graph.

Notice on the graph that although the overhang changes very little with changes in effective arm length, the offset angle changes significantly. It is for this reason, that you should continue this trial-and-error procedure until you have obtained the optimum values.

3. Now construct an offset angle adjustment aid. Take a five-by-seven-inch index card and start a hole with a sharp instrument near and in the middle of one of the narrow ends of the card. Press this hole over the spindle of the turntable. Remove the card, and with a straight edge, extend a radius from the center of the hole toward the further end of the card. With an accurate protractor or triangle or by construction with a compass, draw two lines perpendicular to the radius, one 2.60 inches from the center of the hole, the other 4.76 inches from the hole. These two perpendiculars mark the points on the radius at which tracking error should be zero on the record regardless of the length of arm employed.

Place the adjustment aid on the turntable as though it were a record and set the arm over the outer perpendicular. You can now sight along the line to the stylus to check that tracking error actually is zero. Better yet, if the front face of the cartridge is flat as, for example, the Denon 103 is, you can use the face of the cartridge as a "gun sight." The face should line up with a point beyond the spindle by the same distance as the face of the cartridge exceeds the tip of the stylus. Otherwise sight along the line through the stylus tip to the center of the spindle.

If the stylus is delicate, it is wise not to set the stylus on the card, but rather use the cueing lever or the guard supplied with the cartridge to protect it from harm if the turntable should move.

After adjusting the cartridge mounting on the outer perpendicular, check it with the inner reference mark. Cross checking between the two will help you develop the skill of sighting accurately.

When correctly set up the worst-case error angle occurs at the outer record diameter and each of the other two maximum errors is progressively less (see the illustration). The distortion produced by these errors increases as the groove speed decreases; it is directly proportional to the angle divided by the groove speed. The degrees-error-per-inch-of-groove-radius is equal at the three maximum error points, but is minimized for the available arm length.

4. To adjust the antiskating force, play a low-level recorded passage and raise and lower the stylus from the groove with the cueing lever. Watch the stylus carefully to be sure it remains in the same lateral position in and out of the groove. Adjust the antiskating bias until the stylus remains centered.

-- Mitchell A. Cotter (New York)

SAE 5000: Snap, Crackle and Pop

Since I was the first to call attention to the SAE 5000 (on "Shop Talk"), I feel obligated to respond to the comments of David Satz and Ward Stevenson in the March 1977 Speaker.

The SAE 5000 will not remove pops and clicks from records because the circuitry cannot detect the decay of a signal and thereby differentiate between music waveforms and noise. It's unfortunate that the presentation of the SAE 5000 in ads tends to mislead. The technology and hardware is certainly available to make a de-popper and de-clicker, but not for $200. The cost would certainly be large enough to discourage most purchases. (I'm speaking of costs over $1,000, not just a few hundred.)

What the SAE 5000 can do -- and this is how I presented it -- is to lower the annoyance level of bad scratches, thereby making otherwise unlistenable records enjoyable or at least passable again. To be detected as a scratch, the signal must meet three criteria: (1), it must have a very fast rise time -- less than 100 microseconds; (2) its amplitude must be greater than that of the music signal immediately preceding it; (3) there must be differences in the signal between the two channels. Music signals require two to four milliseconds to reach comparable amplitudes, and it is most unlikely that a scratch would cause identical signals in both channels. And thus the 5000 differentiates between music and some types of noise.

Adjusting the device to try to eliminate pops and clicks, as Satz and Stevenson did, will "confuse" the circuit, and it will try to block music waveforms because the amplitude of the music signal will be very close to that of the pops. Nevertheless, properly adjusted, you can at least play bad records your friends have walked on.

One last thing: as "Adventures in Sound" I think has proven, the more devices, bopmagilvies, amps, etc. you put in a reproduction chain, the less pure the sound will be. A fundamental rule that Audiophilism seems to ignore is that you don't gain something without losing something else. The straight wire with gain is a very misunderstood and mistaken concept. As your speakers and amplifiers know, you lose even when you have only a straight wire without gain. That is, something is lost even when a simple electron flow is created in a copper conductor. So the more you try to do, the more you'll lose. -- Victor Campos (Massachusetts)

Equalization: A Debate with Three Sides

Flat or Rolled Off?

I am baffled by the discrepancies that exist between two schools of thought on room equalization. On the one hand we have Dan Shanefield telling us in the August/September 1976 Speaker and in the May 1976 Stereo Review that we should set our equalizer controls at positions that will render the flattest frequency response as recorded on a strip-chart recorder. That would seem to make sense to me; otherwise, what is the point of striving so hard to get the flattest possible frequency response from our amps, speakers, and so on?

But on the other hand we have the argument advanced by John Puccio in the December 1976 Speaker and by others who say that in order to achieve natural sound and to avoid unrealistic brightness, we should roll off the response by about 3 dB per octave above 1 kHz. This would result in a curve that would be down 12 dB at 16 kHz. In a way this makes sense to me because in live music high frequencies don't have the same energy as lower ones. Therefore, in the actual setting of an auditorium they are not as loud, especially in the back rows.

I regret that I cannot shed any light on this question, but I hope that my observation will trigger a discussion in The Speaker among those who can. - Reynaldo Puesan (Connecticut)

A Case Against Equalization

If equalizers did no more than alter the tonal balance of a signal, I would have no argument against them. However, this does not appear to be the case. I gather from friends who use them that even the best add colorations of their own. And I find it difficult to believe that equalizers do not affect those most elusive of qualities: the openness, airiness, depth and three-dimensionality only the finest of preamps and amplifiers can reproduce. Whenever I switch in my Dayton-Wright SG Mk. II equalizer, even when it's set for flat response (Few equalizers are really flat at their center settings. --Ed.), these qualities deteriorate, especially in the upper frequencies, Mr. Puccio's comments in the November Speaker notwithstanding. This loss is not just a volume effect, because increasing the level of the equalized sound during the comparison does not help.

Some time back a friend with a Soundcraftsmen and I compared equalizers directly, and they seemed to perform identically (perhaps the Dayton-Wright was even cleaner), so I guess that the Soundcraftsmen would also have detrimental effects on sound quality. I gather that the Soundcraftsmen is one of the best, which leads me to suppose that all readily available equalizers would add similar colorations. This conjecture seems to be confirmed by the experiences of friends who own equalizers.

The question then becomes one of how much coloration one is willing to accept to achieve improved tonal balance. Personally, I would prefer the degree of imbalance found in the usual unequalized system to sacrificing airiness, openness and three-dimensionality, but tastes could differ here.

One should keep in mind that most equalizers are not precise enough to properly equalize a room. I understand that one really needs a one-third octave band equalizer to do a proper job, but such devices are very expensive (and difficult to set up -- Ed.). I assume that the best one can do with the more common octave equalizers is to reduce large, wide-band deviations to small ones. So, is only a limited restoration of tonal balance worth the added coloration? I do not have much direct experience with equalizers, so the above may reflect just my own degree of ignorance, but I do believe people who plan to equalize should consider these questions.

-- Collins Beagle (Virginia)

Audio Systems More Perfect than Live Music?

I would like to add a comment to John Puccio's statements (on page 6 of the December 1976 issue) on "live" versus "perfect" sound. I agree completely with Mr. Puccio that a live musical performance in many ways is not equal to what an audiophile's system produces, but I believe that Mr. Puccio has placed the emphasis for this state of affairs at the wrong end of the reproduction chain. It is not the speakers which are the primary cause for this but the microphone technique used to make the recording. Recording engineers can and do place microphones in places where the human ear cannot go during a performance (when was the last time you listened to a piano recital from inside of the piano?), which results in high-definition recordings that are unnatural in terms of a live listening experience.

I became much more conscious of recording engineers' inclination to a high definition approach because of experiences with a recording engineer friend of mine. Primarily because of my desire to recreate a natural, ambient sound space (but also because of a limited budget) I use only two microphones in a stereo configuration when I make recordings, regardless of whether I am recording a solo folk singer or a full choir with chamber ensemble.

Trying to aid in my efforts, my friend has offered advice which usually sacrifices a coherent sound space for high definition. He would have me record a musician who sings and plays guitar by placing one microphone a few inches from the guitar strings and the other a foot or so from the singer's mouth. By this means I will get a very clear recording.

My response is that in the process one is losing any possibility of capturing a natural sound space and that in a live situation one would not hear such transients and definition anyway. I guess our instincts and aesthetic goals are just different. Judging by most recordings, my friend's view seems to be the majority one among recording engineers.

So do not lose a proper perspective on home playback equipment. Each component in the chain should pass as accurate an analog of the signal it is fed as possible. If one wants to reproduce in the home an experience closer to an actual performance, then one has to choose recordings engineered with this goal in mind (such as Fulton's records). -- Collins Beagle (Virginia)

Is a Pre-Pre-Preamp for You?

I use three preamps. I like their combined sound better than that of just one or two. But I don't know why.

It all started with my purchase of a Davis-Brinton phono preamplifier. As most people know, the output from the Davis-Brinton must usually be routed into the aux input of a control preamp. In my case, I used my Soundcraftsmen PE 2217. But during the course of some experimental switching of components, 1 substituted my Citation 11 for the PE 2217. Strangely, I liked the clarity and realism of the sound much better than before, although I missed the deeper bass the Citation's meager equalization facilities could not provide.

So one day I plugged all three together, both to regain the equalization and to see what would happen. Lo, the sound was the best yet gutty softness of massed strings, clean bite of brass, the whole bit.

I might add that the Davis-Brinton made all this possible -- when in the circuit it cleaned up the bottom and imparted a clarity and delineation to the midrange superior to that of any other combination I have heard. Yet the whole is better with both of the other preamps in the circuit than with either alone. What I'd like to know is why.

It's not my imagination, because after being away and hearing the New York Philharmonic in its new hall and the London Philharmonic in Royal Albert Hall, I returned to my rig and used one preamp first, then the other, always with the Davis-Brinton, only to find the three together gave that last measure of realism.

For the record, other components are a Thorens TD 125B with the Rabco SL8E and Decca Mk. V export model, the Citation 12 power amplifier, and four speaker systems: in front a pair of AR1W/Janszen 130's and in the rear a pair of KLH 7s fed through a Dynaquad. A dbx 119 is in the tape loop of the PE 2217. Incidentally, my speaker cables are 12-guage solid copper wire, an idea from The Speaker that dramatically improves the output of the power amplifier.

-- William S. Vincent (Connecticut)

KEF 104aB Speakers

I agree with Collins Beagle's review (March 1977 Speaker) of the KEF 104, particularly his remarks about depth imaging, definition and dynamic range. These features are particularly evident listening to "Unrehearsed Experiment" (side 1, band 2).

However, compared to Dahlquists, they sound dull and not as open -- not as alive as the Dahlquists. Voice sounds more natural on the Dahlquists, and ambience is captured better with the Dahlquists. The Dahlquists project a larger stereo image than the KEFs; bass reproduction is subjectively on par. The Dahlquists seem to interact with our room more than the KEFs do, especially as regards stereo imaging, with the result that the KEFs image slightly better.

For anyone spending $700 on new speakers, I think the KEFs would be a good choice.

-- Roger Foster (Manitoba)

"Rough Trade Live" Direct-to-Disc Album

This album, cut by a popular group from Toronto, has been mentioned with praise in at least two equipment reviews in Audio. Indeed, there is some deep, well defined bass and good dynamics cut in this disc (e.g., Birds of a Feather" and "Take Me"). Focus is good and imaging three-dimensional, but vocals don't sound very natural on the four systems I have used to reproduce this recording.

The lyrics and music are not everyone's cup of tea.

- - Roger Foster (Manitoba)

Two Views of the Snell Loudspeaker

For the past few months, it has been the reviewer's (and his spouse's) pleasure to live with a new loudspeaker so good, as to defy determination of its total potential for sound reproduction. In attempting to expound upon this wonderful discovery for audioneurosis, however, the reviewer is caused to contradict his own cynical premise: viz, it is impossible to write a meaningful (to others) loudspeaker review.

Subjective reviews, the lifeblood of the cognoscenti, are, because of the existential nature of the listening experience, relevant to nothing. (Acid test: spend an evening with the BSO at Symphony Hall. Rush right home, and play the same music on your megadollar hi-fi system.) Objective reviews of speakers tend to be impossible because of the absence of the ultimate comparison, with the live sound source (and of other, more practical considerations, such as the lack of valid psychoacoustic specifications). Thus, with the admission that it is virtually impossible to write a meaningful loudspeaker review, the reviewer proceeds, urged on by his own Carteresque sense of duty.

The Snell loudspeaker is the end result of three years' intensive development by the designer, Peter Snell. A fundamental design criterion was flat response, both frequency and power into the listening room. To this end, the configuration of the speaker's three drivers is a vertical line with best dispersion directed toward the other speaker of the mirror-image pair. The midrange and tweeter are placed on a rounded baffle to minimize diffraction effects and are specially loaded to increase dispersion. The woofer is also specially loaded, and the speaker is designed to be placed at a floor-wall intersection, to minimize room boundary effects. Visually, the loudspeaker has excellent proportion and appears to be a single walnut cabinet with black grillcloth. In fact, it is divided into two modules, the woofer section and the midrange tweeter section, which is positioned above it. The modular nature of the speaker allows for easy handling and effectively eliminates any chance of woofer interaction with the other drivers. The manufacturer states typical room response to be +2 dB from 36 Hz to 18 kHz at up to 45° off-axis, and the reviewer has no reason to doubt it.

In addition, great care has obviously been lavished on the cabinet work, a rare treat. Even greater care seems to have been placed on the innards, as each crossover network is hand tailored to the drivers, and each driver separately fused. The quality throughout is impressive indeed, as the designer is a perfectionist in his desire to assure the consistency and longevity of his product.

The reviewer, in attempting to find some evidence of sonic flaw, has almost given up in exasperation. Any sonic aberrations have ultimately been found attributable to other sources, such as the program material or other components in the stereo system. Perhaps the only flaw (?) of the Snell speaker is the merciless revelation of what comes before it. (The reviewer remains slightly suspect of its very top end, but not when playing master tapes.) Trial and error have led the reviewer to the following observations:

1. The Snell does not sound like a loudspeaker. It sounds like the program material colored by the other audio components.
2. Quality control on the speaker is outstanding -- every Snell sounds like every other Snell. (Try A-Bing your own speakers with each other.)
3. The speaker sounds fine in any size room.
4. The speaker induces no sense of listening fatigue. Its sound is a welcome relief after auditioning other speakers.
5. The general reaction of new listeners is one of amazement. Typically, their response is "where are the musicians?"
6. The imaging characteristics of the stereo pair and the depth perspective of the image are most realistic on all forms of program material.
7. The speaker does not favor any portion of the frequency spectrum. Hearing a dynamic woofer not exaggerate bass is unique.
8. Sound at low volume remains clear, and the speaker can play painfully loudly in a large room.

One listener remarked that the speaker could not be as good as it sounded. The reason put forth was that it was not esoteric enough -- any speaker of that quality just had to be an electrostatic! The only true problem for most audiophiles will be their inability to audition the Snells because of the currently limited dealer network.

For the cognoscenti, the following ancillary components were used: cartridges: Shure V15-111C (w/Formula 4 arm), Denon 103C (w/Stax arm), Decca Gold (w/Formula 4 arm), Sonus Blue Label (w/Grace 707 arm), AKG (w/Grace 707 arm); preamps: db Systems, Levinson JC-2, Yamaha C-2, Rappaport Pre-1, AGI, Trevor Lees, Dunlap Clarke 10; power amps: Epicure, Yamaha

B-2, Son of Ampzilla, Dunlap Clarke 500; master tapes recorded and played on modified KLH 40; other loudspeakers compared: Rogers BBC, Beveridge, Dahlquist, Allison 1.

-- Joel Disend (Massachusetts)

I have been enjoying a pair of Snell Acoustics speakers for the past four months, and I would like to take this opportunity to share my impressions of these transducers with members of the BAS. Before buying the Snell Acoustics speakers, I had become increasingly dissatisfied with my KEF 104's. The KEF 104's are basically accurate transducers, but, in my opinion, they suffer from several flaws. To wit, they lack truly deep bass, sound a trifle too warm and resonant in some portions of the mid bass, and have a tendency to sound a bit boxy. In addition, the KEF's have superior imaging characteristics, but only if you're sitting more or less directly between the two speakers. I mention these properties because the KEF 104 is a fairly well known and highly regarded speaker, and I hope to give the reader some idea of my tastes and biases, so that my comments about the Snell speakers will be more meaningful.

Snell Acoustics is a small speaker manufacturing and development company in Newburyport, Mass. The founder, Peter Snell, has expended considerable effort to design and build a speaker system of exceptional quality and sound character. The Snell Acoustics speaker is basically a three-way, floor-standing system, which uses conventional dynamic drivers. However, the acoustic signature of each driver is highly modified in the final product. Each speaker's character in the final system is a result of its having been modified either structurally or by the way that the driver, is mounted in the cabinet or both. In addition, Snells' skillful crossover design achieves a homogeneity of sound which is rare today. The tweeter and midrange are mounted on a sculptured baffle in an enclosure which is separate from the bass enclosure, to avoid diffraction effects and driver interaction. The bass unit is an acoustic suspension design, which is unusual in that the woofer faces downwards towards the floor. The enclosures are very substantial and require a great deal of care in their construction.

The reproduction from these speakers is truly remarkable. First of all, these speakers have one quality which I find in almost no other speaker: namely, extraordinarily wide dispersion combined with excellent imaging ability (depth reproduction is unusually fine). The speaker has very good deep bass reproduction, which is rivaled only by a separate bass commode, such as the Bottom End. The mid-bass is tight and articulate without a trace of heaviness or resonance. The midrange is very smooth and beautifully open sounding, without any hint of the hollow "cupping" type of coloration so common in conventional, bookshelf speakers. A simple test is to listen to applause reproduction on the Snell Acoustic speakers. Most other speakers sound absurd by comparison. The high end reproduction sounds similar to that of electrostatics but without the electrostatics' tendency to beam. The high-frequency sound remains clear and well delineated, even when the speakers are reproducing complex orchestral passages or the exaggerated high ends of most pop records. In short, these speakers are exceptional in their ability to present a highly focused but very open sounding sonic image which is both delicate and uncluttered at all listening levels.

The Snell is a three-way dynamic loudspeaker system, $1,370 per mirror-image pair. Drivers: 10-inch woofer, 4-inch midrange, 1-inch dome tweeter. Crossovers: 300 and 2500 Hz. Impedance: 4 ohms. Minimum recommended power: 40 Watts. Dimensions: 46 1/2" H x 23 3/4" W x 13" D. Weight: 97 lbs. Information on dealer locations can be obtained by writing directly to the factory at 10 Prince Place, Newburyport, MA 01950.

-- Gary Rancourt (Massachusetts)

Beveridge Speakers: A Minority Report

I would like to offer some comments on the Beveridge speaker, which to date seems to be getting fairly unanimous praise. I have had the opportunity to hear them on three occasions for a total of several hours, and my own reactions are not so positive. Originally I listened to them for about an hour and a half in a leisurely manner in a home environment (General Acoustics, Silver Spring, MD). Associated components included an AGI 511 preamp and a Denon conical cartridge with a modified Denon AU-320 transformer. During this first experience T thought I was listening to the finest system I had heard to date. In retrospect, after relistening to my own Dayton Wright

Mk. 3's, I decided that though the Beveridges were excellent in the highs, I was not so sure that they were superior to the D. W.'s in the midrange down.

Then I had the opportunity to hear the speakers again at Audio Arts, Richmond, VA in another home environment for a leisurely three hours. Here the electronics setup was most curious, with a Fidelity Research Mk. 3 cartridge feeding a Quatre pre-preamp, which fed an Accuphase C-200 preamp, which in turn fed the amp section of a Stax SRA-12S preamp.

This time I started to develop reservations about the speakers because of their unnatural sound field and poor focus (here I am ignoring the bright -- to the point of harshness -- quality of the system, which I assume is caused by the Fidelity Research cartridge and Quatre pre-preamp -- not the speakers.

Then I finally heard them again at Audio Arts, when Harold Beveridge (a most easy and enjoyable person with whom to speak) was making a presentation. Though activities lasted over several hours, I only listened seriously for about an hour. The setup was the same as before except that the Accuphase preamp had been replaced by an SAE IXB.

During this session I developed further reservations about the speakers and finalized my own views. On the positive side, the speakers will fill an area with sound, presenting a stable stereo image from most points in a room. A friend of mine who went with me was most impressed by this quality. But one pays the price that the sound field is as wide as the speakers are apart. Consequently, a small jazz combo is spread as far apart as a full orchestra. I am not sure if it is a related problem, but the speakers have rather poor focus, with sound sources creating a vague image. The speakers appear to have good depth-imaging, but their unnaturally wide sound field creates a more shallow overall perspective.

In terms of tonal balance, the speakers appear to be good except for a rolloff in the mid- to deep-bass region. The upper frequencies are reproduced with crystalline clarity (this is the quality which most attracted me to the speakers). But as one descends in frequency, definition, transient ability, detailing, and focus all deteriorate. Also, this speaker seems to act as a compressor on dynamic peaks, especially in the lower frequency range. The friend who was with me the third time put his finger on the Beveridge's problem (and he holds a higher opinion of it than I do): one never has the sense that a solid sound source is present. We were listening to a somewhat spectacular display of musical phantoms. The speakers failed to have that ultimate sense of detailing, air and three-dimensionality. So, for me the Beveridge is not one of the state-of-the-art speaker systems around. However, this is a matter of personal taste, and listeners must decide for themselves.

-- Collins Beagle (Virginia)

Discs for Drum Sounds

For the type of audiophile who enjoys sound itself (in addition to music), I would like to report the discovery of a little sonic gem: a series of sharp, clear drum beats which are recorded at very high volume. They occur from 80% to 95% into Band 4, Side B on "Gladys Knight & The Pips - 2nd Anniversary, " Buddah BDS-5639.

By comparison, the drum sounds on other disc recordings that I've heard have much less sock-it-to-itiveness. There are plenty of others just as loud but which are (to my ears) unpleasantly bloopy, even with equalization. An example of such bloopers is at the beginning of Band 3, Side B, of "The Best of Gladys Knight & the Pips," Buddah BDS-5653. Another one is the famous "From Menaggio to Bellagio" on Sheffield LAB-1. There are a few recorded drums which are slightly more realistic, such as those at 15% of Band 2, Side 2 of "Percussion Music, " Nonesuch H-71291, but these are not loud enough to have much punch, in comparison to the other instruments.

So I recommend BDS-5639 to all (of us) punchdrunk drum nuts. There is occasional boomy electric bass in the background, but you can't let that bother you. Also, the music is pretty good, too, in a popsy rhythm-and-blues vein.

-- Dan Shanefield (New Jersey

PCM Records, Chapter XXV

Since this serial has been going on for a long time now, perhaps an update is in order. The hero (or villain) of our tale is Odyssey 33200, a solo flute recording featuring Jean Pierre Rampal. It was made by Nippon Columbia using a digital audio recorder. The plot hinges on the question of whether this record advances the state of the recording art. Several correspondents have offered opinions on this issue, among them Collins Beagle and Scott Kent. In August, Beagle asserted there's so much traffic and room noise on this recording there's no way of telling. He also suggested the hall ambience is excessive. Scott Kent responds to these points in our November issue, asserting that the object of the recording was not to prove how quiet a PCM disc could be, but to demonstrate that a digital recorder could cope with the challenge of a predominantly sinusoidal sound, such as that produced by a flute, without introducing noticeable anomalies attributable to the step-function (digital) process. The recording, he said, meets the challenge.

And with that for background here's this month's episode:

I would like to add one more comment to the PCM dialogue. If I understand Scott Kent's point correctly, he is suggesting that this recording challenged PCM at its possible weak points, and the PCM system came through with flying colors. This is all well and good and nice to know, but it is somewhat beside the point. Great, the PCM method does not demonstrate inherent problems, but is it superior to conventional analog techniques? This was the question that was originally raised and that we have been discussing.

I didn't say that I dislike ambient information. Rather, I said I do not care for overdoing this sort of thing. I agree completely with Mr. Kent on the desirability of capturing natural ambience while making a recording (and this is one of my chief concerns when recording local performances). Along with him, I would certainly object to later processing to create ambient effects. But I become concerned when the ambience is so prominent that it draws attention to itself, distracting the listener from the performance. Even when the intentions are the best, this situation can arise if the recording locale is, for whatever reason, unsuitable for the performance. I believe such is the case here. It comes down to personal taste, but to me what Kent calls a "large quiet church or well-designed classical studio" is not the most natural environment for a solo flute performance. Thus I find the ambience of the PCM disc a bit disconcerting. -- Collins Beagle (Virginia)

(Note, though, the accursed ambience is a function of the recording mike location. PCM or analog would have reflected this ambience in about the same relation to the direct sound. --Ed.)

Review of Two Four-Channel Tapes

I just received my first two discrete, Dolbyized, four-channel tapes from Barclay-Crocker, and I am highly pleased with them. They are:

1. ISAO TOMITA - FIREBIRD * STRAVINSKY
   Firebird Suite: Introduction & Dance of the Firebird/Round of the Princesses (Khorvoid)/Infernal Dance of King Lastchel/Berceuse & Finale/DEBUSSY: Prelude to the Afternoon of a Fawn/MOUSSORGSKY: A Night on Bald Mountain.
   7 1/2 ips. Quad, 50 min. Dolby
   RCAQ 1312 Price: $11.95
2. JEFFERSON STARSHIP: Red Octupus
   Contains all songs on album, notably the hit "Miracles."
   7 1/2 ips. Quad, 42 min. Dolby
   GRUQ 0999 Price: $11.95

Both tapes came wound smoothly on see-through plastic reels, to each of which was attached a sticker indicating total time, format, and selections. The front and back of each album is readably reproduced on the respective sides of the box, much the same way as is done with SQ records. Short but adequate Dolby calibration tones are at the beginning of each tape. The tape surface appears to be relatively highly polished and quality appears uniform throughout. The tape comes wound so it must be rewound before playing (which is mentioned on a card inside the box). The tapes are distributed by Stereotape, and a card to join their mailing list (different from Barclay-Crocker's?) is also enclosed.

To my ear, the quality of these tapes in fidelity alone is far better than that of the corresponding records (thanks to Dolby), and justifies what some might consider a slightly high cost. Particularly, in the case of the dynamic ending of the "Firebird Suite," the clarity and definition of the tape makes it infinitely preferable to the disc, where the finale is crammed into those infamous inner grooves, causing audible distortion after only a few playings.

I am eagerly awaiting the next issue of Barclay-Crocker's bimonthly supplement to their (open reel only) catalogue, to see what other additions they will have. These tapes are both relatively new releases, and because turnover time from record to quad availability seems only a matter of months, I won't mind the wait if it means eliminating double inventory, i.e., having both record and tape.

As regards the discreteness and the added dimension of having two rear channels, this, too, makes both these recordings a worthwhile purchase -- as much, if not more, than their sonic quality. The Tomita tape stood out in the former respect, but this is partially for reasons having no direct connection with the quality of the recording.

Firstly, the Tomita rendition of the "Firebird Suite," being totally electronic music (in case you didn't know), is likely to be quite a novelty to the symphonic-concert-going listener. The attack and decay of musical sounds is generally much more controllable in electronic music than in its purely acoustic analog, and can, if the performer is not careful, sound unnatural, or at least different from what the listener (or even at times the performer himself) expected or intended. The point is that electronic synthesizers (digital and analog) are capable of such precise control over every parameter of a musical event (attack, decay, sustain, release time, to mention a few) relative to "conventional" instruments that things may sound a bit bizarre to the naive listener. And although these, along with the timbres and voicings chosen in a particular musical context, can be purely a matter of taste, they are nevertheless startling to the ear when first perceived, not only because they are changes from the "normal" musical palette, but also because they are themselves capable of rapid change -- which is what the ear is most sensitive to. Thus, when the impact of all of these factors is doubled by the quadriphonic recording, even if its fidelity and discreteness are not noteworthy, the net effect is, I contend, greater to the (electronically inexperienced) listener simply because he is dealing with electronic music, as opposed to comparable symphonic liberation from stereo to quad.

Secondly, a more open sounding work is bound to result from a quad mix, as the instruments are not so buried in the acoustical field as they would be in a stereo mix.

All of the above (fidelity, discreteness, and the still-novel sound of electronic synthesizers) together with some admittedly gimmicky mixing (pan-potting around the four-channel field), result in "sheer sonic sorcery:" speeding multiple-octave transpositions and what have to be sequenced (as opposed to manually-executed) chromatic lines applied to oscillators and audibly sophisticated speech synthesizing related circuitry (even though Stravinsky didn't score any vocal effects), make this tape worth owning just to show off your system, if nothing else.

From the point of view of electronic music technology, this tape demonstrates one individual's use of what approaches current state-of-the-art synthesis. I am especially impressed by the subtle choral and vocal effects which are to my (admittedly untrained) ears almost indistinguishable from what real voices might have sounded like. (There is one rather eerie sequence in which a high frequency tone with slight vibrato, i.e., slight modulation by a low-frequency sine wave, gradually descends in pitch, transforming into a female soprano with a tremelo.) So, even if you don't agree with Tomita's interpretation of Stravinsky or Stravinsky himself, for that matter, this recording at least deserves praise from a purely technological point of view in terms of both fidelity and of the use of electronic music synthesis techniques.

What more can I say except that I wish I had just some of Tomita's Moog modules instead of my now measly ARP, organ, and piano, and that you should listen to this recording just to hear the kinds of things that can and are being done with electronic music synthesis.

The Jefferson Starship tape is also thoroughly enjoyable. The romantic qualities of ira- cles" are particularly enhanced by quad. However, not being electronic music, my pet project, the Starship must simply take second place. Apologies to lead singer Marty Balin.

A note about delivery: Sent special handling, 50¢ for one to two reels, 70¢ for three or more, it only took about eleven days for me to receive the tapes. Ironically, I got the acknowledgement saying that Barclay-Crocker had sent the tapes (initialled by John Crocker himself) after I received the tapes.

I am interested in hearing from other BAS members who share my enthusiasm for electronic music. I can be reached at Rm. 359, Freefer Hall, Tufts University, Medford, MA 02155, (617) 625-5532.

-- Robert J. DiCamillo (Massachusetts)

Tips, Quibbles and Queries

I have a little tip for my fellow members. As soon as I receive my issue of the Speaker, I give it two clips along the left edge and remove the top corner clip. This gives the magazine more firmness and the advantage of pages that turn like those in a book. This way you don't have to depend on that little corner for the support of all those pages.

On "Phono" and "Tape"

In "A Test For Preamplifier Audio Quality" (June 1976 Speaker), Alvin Foster says that Soundcraftsmen has incorporated in newer units of the PE-2217 a modification to raise the phono overload level. In a prompt answer to my letter, the Soundcraftsmen people cleared my mind and correct Mr. Foster in what seems to be an assumption he made:

"First of all, the magazine article was wrong in its statement that we have incorporated this modification in our units. We have not. We are enclosing an instruction sheet for performing this modification. It is quite easily accomplished and requires only four 47 kOhm 1/2 Watt 5% carbon resistors. (Quite common and should be available from a local parts supply house.) You in no way affect the warranty of your unit by performing this change. "

Your article on tape machines for stereo field recording by Cary Lu (October 1976 Speaker) was very helpful and partly determined my decision to buy the Sony TC-1535D cassette portable deck. Soon I found it was incompatible with my Sony TC-177SD. It would have to be rebiased and re-equalized to the 70 us deemphasis, and the Dolby would have to be recalibrated for use with TDK SA. Does anybody out there know if this can be done to the 153 (or 152, which is the one I was planning to buy until this problem arrived) without detriment to the machine's operation or quality? When asking Sony for a service manual for the 177 I told them about my problem. I have my manual but not a word on how to modify model 152 or 153, which appear to differ only in their bias switch. The 153 has the added alternative of ferrichrome, but with normal equalization.

—Eduardo Valzaquez (Puerto Rico)

On Amplifiers and Speakers

Robert Graham's article on the McIntosh amps was very interesting and informative. I doubt any of the "glossies, " or even the "undergrounds" would publish something as well researched and knowledgeable. Congratulations. This, by the way, restores my faith in McIntosh, which I had thought to be durable, beautiful, mediocre equipment. Now it seems to be durable, beautiful, excellent and innovative.

This amplifier theme reminds me of a thought culled from the latest issue of The Absolute Sound. They printed a comment denigrating any amp of less than super-power. The question that I would like answered is whether 75 Watts/channel or less is considered insufficient for true high fidelity. I am somewhat disconcerted, because I have heard a pair of IMF Studios driven by a Dynaco Stereo 150 with rather good results. And those speakers seem not to be very efficient. Perhaps an article on amplifier-power-needs by Bob Graham would be suitable.

Also, I used to own AR-1w's with AR-3t's (the equivalent of a pair of AR-3's, but with two enclosures a side -- one for the woofer, another small one for the mid-hi). After having lived with them for about three years I was lent a pair of Janszen 130's. By listening only, I could tell that the AR-1w's slope at its high end and the 130's slope at its low end do not match; there's a hole left between the two (moderate, but still a hole). This is in comparison with the 3t. I definitely preferred the all-AR sound to that of the AR/Janszen combination.

-- Carlos E. Bauza (Puerto Rico)

Low Capacitance Cable ...

Those who favor low capacitance shielded cable for component interconnection may want to investigate Dearborn 6058F. This is an RG58/U coaxial cable with a foam cellular polyethylene dielectric. The center conductor is 20-guage solid wire, and the shield is braided copper. Rated capacitance is 17 pF per foot. This cable is compatible with Switchcraft phono plugs and offers better shielding than Belden 8421 and lower capacitance than Belden 8417 or other coaxial cable of similar diameter (6058 O.D. = 0.195").

... and Record Rumble

Recently I have bought several chamber music recordings, each of which has received critical acclaim, in which the musical performance is marred by low-frequency interference resembling the sound of traffic noise or rumble. Specifically, the records were Phillips 6747139 (Beethoven, Op. 59 Quartets), 6500241 (Mozart, Quartets 20 and 21), 6500620 (Mozart, Quintets in G and C), and RCA ARL1 1569 (Chopin Recital). Both Phillips and RCA have courteously offered either to replace my defective copies or to provide substitutes (RCA acknowledged that early copies of ARL1 1569 were pressed from defective stampers, whereas Phillips stated that the master tape for 6747139 was at fault, and that the Mozart recordings were being returned to Holland for analysis).

Though it is gratifying that major companies such as Phillips and RCA are willing to promptly and directly exchange defective records, it is annoying, to say the least, that record critics make no mention of faults such as recorded rumble when performances are reviewed. As my loudspeakers (DQ-10s) are not noted for extraordinary bass response, it is unlikely that I have uncovered defects which went unnoticed by the majority of professional reviewers. Occasionally an American reviewer will report poor surfaces, but rarely more than that, and the hapless collector is left on his own with regard to a host of other potential problems, many of which may be much more troublesome than a few ticks and pops. The BAS could perform a valuable service if, in addition to publishing reviews of outstanding records, it also were to list from time to time records marred by endemic faults, such as rumble, which may afflict all available copies.

-- Ronald Kalil (Wisconsin)

Replies

I' m glad Carlos enjoyed the review of the McIntosh amps. Hopefully, others got something out of it too.

In reference to the AR-1w and the Janszen, it would be good to know how old the janszen unit was. The crossover points of the various models have ranged between 500 Hz (claimed) to somewhat over 1 kHz. As I stated in my review of these units, the AR-1w cannot make it by itself to the Janszen crossover, and a mid-bass driver should certainly be used. When thus applied, I think it's a superior system.

-- Bob Graham (Massachusetts)

Power doth not fidelity make. My ST-150 is the most powerful amplifier I've ever owned. I'd like to have twice the power, but I could live with half -- 3 dB either way. If it sounds okay to you, why worry?

-- Michael Riggs (Massachusetts)

In the Literature

Before you get any further down the page to locate the best circuit designs of the month or to figure out where you can save money on magazines, a few words about this column:

If this were one of those slick-paper magazines on the newstand, we'd have Dana Craig's picture right up here next to the head. You see, in theory this column is a joint effort featuring contributions from everyone who's found something interesting "In the Literature." In practice though, at least 80 percent of the contributions come from Dana Craig, and so far as we know, Dana isn't a speed reader. If you scan the list of articles below, you'll appreciate the depth of coverage Dana gives the periodical scene and the amount of time he must give to this job each month.

Well, Dana deserves a break and he's taking one. He'll be away during the summer and that means an 80 percent reduction in the size of this column unless some of you out there pitch in. That doesn't mean one person sending in three pages of notes on articles -- not that we want to discourage anyone. But rather, if a half dozen interested members -- especially those of you who have access to the less widely circulated journals -- sent in a few lines on two or three magazines (or even one), we'd be able to fill in the gap. So hop to it! Perhaps you too will be rewarded by a phone call from our sleepy-voiced editor saying: "Thank you. Your contribution was much appreciated, really."

-- Henry Belot

Audio, May 1977

*What You Don't Know, Hurts!: Koss VP discusses sound perception. (p. 32)
*Build a Low-Noise Preamp with Weighting Filters. (p. 38)
*Thirty Years of Audio: Norman Eisenberg reminisces on Audio's 30th anniversary. (p. 48)
*How We Hear: By psychology professor. (p. 54)
*Test reports on Pioneer TX-9500 II tuner, DB Systems DB-1 preamp and DB-4 pre-preamp, and Technics RS-1500 US open-reel tape recorder. (p. 66)

AES Journal, March 1977

*Fast Room Acoustic Analyzer (FBA) Using Public Telephone Line and Computer: Sony engineers can now analyze your room by remote control. (p. 82)
*A Dynamic Noise Reducer for Sum-Difference Multiplex Systems: Channel separation is traded off for improved S/N. (p. 95)
*The RCA Quadulator: New CD-4 modulation system. (p. 99)
*Transistors Can Sound Better than Tubes: Includes design for high-voltage preamp. (p. 116)
*Am I Too Loud?: A Symposium on Rock Music and Noise-Induced Hearing Loss. (p. 126)

Audioscene Canada, March 1977

Equipment reviews of four speakers -- Dual 470, McIntosh XR-5, Marantz HD88 and Tandberg Fassett -- plus the Sansui 9090 receiver. This month's technical paper by Dr. Floyd Toole is his first article on controlled listening tests, with data to back his conclusions. "Audio Trends" section notes that the first Cleveland Orchestra direct-to-disc recording was supervised by Bert Whyte; and a second direct-to-disc recording "under" the Umbrella label, called "Ragtime," is now available.

Datamation, March 1977

*He's Making Caruso Sound Like Caruso: A look at the techniques Dr. Thomas Stockham of Soundstream, Inc. uses to "restore" Enrico Caruso recordings made prior to electronic recording (1926). The article discusses the many problems in the original recording process and the "deconvolution" computer processing used to detect and eliminate them. (pp. 168-71)

db: The Sound Engineering Magazine, April 1977

*Glossaria Interruptus: Humorous definitions of audio engineering terms. (p. 18)

*Control Room Acoustics. (p. 26)
*Transducer Power Handling: By JBL engineer. (p. 31)

Electronics, March 31, 1977

*Automatic Gain Control Has 60-decibel Range: Circuit uses 741 op-amp. (p. 107)
*MOS Op-Amps Form "Pink Noise" Source: Circuit uses two CA3130 op-amps. (p. 118)

Electronic Engineering Times, March 21, 1977

*Log Adaptive Delta Modulator: Similar to the Audio-Pulse modulator in technology, this and similar circuits using D/A convertors appear in pages 48-50 of this issue.

High Fidelity, May 1977

*Brief review of SAE 5000 noise reducer is favorable. (p. 30)
*Test reports on Yamaha CR-820 receiver, Kenwood KD-550 turntable, Sony TA-5650 V-FET amplifier, Dynaco SE-10 octave equalizer, and Shure M-615AS Equalization Analyzer System. (p. 35)
*New Products at Half Time: Report on the Winter Consumer Electronics Show, which occurs in the retail industry's midseason. (p. 60)
*Quality Portables for Vacation Entertainment: Hi-fi you can carry along. (p. 66)

Mr. Audio's Bimonthly, Vol. 1, No. 1

This is mostly a vehicle for Trevor Lees, an Australian engineering student whose tube preamp has caused a stir on the west coast. It's not a review magazine. Most of the material concerns design philosophy (discussed rather vaguely) and equipment modifications. Perhaps with good reason, the publication takes itself less seriously than do many others. It's a relief. Sometimes, though, one gets the feeling he's paying for Trevor's advertising. $12 for six issues from Mr. Audio's Bimonthly, Box 77225X, San Francisco, CA 94107. (Hold off on subscribing, though. The editor has resigned, and the magazine may be in limbo.)

Popular Electronics, May 1977

*Stereo Scene: The Decontamination Squad: Discusses phonograph cleaning. (p. 18)
*Julian Hirsch tells how headphones are tested and evaluates the Sennheiser HDI434 infrared headphones and Teac PC-10 portable cassette recorder. (p. 26)
*Introducing Speechlab -- The First Hobbyist Vocal Interface for a Computer: This example of future shock could be used to control the operation of a stereo system through voice command. (p. 43)
*How to Match Hi-Fi Components: Unsigned article tells you how to apportion your money. (p. 66)
*VMOS-MOSFETS with Muscle: Including circuit for 40-Watt amplifier. (p. 76)

Radio-Electronics, May 1977

*New Way to Room Equalization: Len Feldman evaluates Shure system. (p. 42)
*Test reports on Nikko TRM-750 amplifier, and Heath AP-1615 preamp. (p . 56)

Tape Deck Quarterly, Winter 1977

*Cassette Tape Roundup. (p. 4)
*Taping in the Field. (p. 10)
*Profiles in Tape: The Capitol Story. (p. 12)
*Feldman Lab Reports on Teac A-400 cassette deck, Dual C-919 cassette deck, JVC MI-E60/B 6-channel mixer, Sansui SC-2000 cassette deck, Nikko Beta I preamp, and Sanyo STD-2000 cassette deck. (p. 24)

-- Dana Craig, Harry Zwicker, Roger Foster and Michael Riggs

We really do need someone to take over this column. For once, by the way, we have a job that can be done by an out-of-stater. Get in touch with me through Box 7. It's a big plus if you have access to engineering journals and the like.

-- Mike Riggs

April BAS Meeting

The April meeting of the BAS was held once again at Boston University's Sherman Union. Jim Brinton opened the meeting with several announcements, including the traditional requests for speaker-stuffers and articles for the Speaker. There was also a request for a volunteer "Speaker stamp-Ticker." The current "stamp-licker," after several years of faithful service, announced his retirement (presumably with a bad taste in his mouth). Al Foster mentioned a two-record set of the B.U. Symphony Orchestra, which is both musically and sonically excellent and costs only $8.70 postpaid from: Boston University, School for The Arts, Room 202, Boston, MA. Local members may obtain it directly from the B.U. bookstore for $7.95. Other items for sale at the meeting were announced, and Brian Leeming requested overseas record orders to be submitted by May 1 for Order #2 of this year. Henceforth overseas orders will be placed quarterly rather than monthly. Eliot Berger of E.A.R. , Inc., requested that those who received the vibration isolators last month send him their comments. They were not totally free gifts: the recipients have an obligation to discharge and should send their feedback to Berger, c/o P.O. Box 7.

Meeting Feature #1

John Naab, Director of R&D, Genesis Physics Corp., displayed to the BAS the three loudspeaker systems in the Genesis line. All three use an 8-inch woofer (with a bright green surround) and a 1-inch tweeter. Starting with the Model I, having one woofer and one tweeter, Naab outlined the design considerations for the two basic drivers in the Genesis systems. The key premise in determining the basic driver components was to obtain good midrange performance with a two-driver system. An 8-inch woofer (6 1/2-inch effective piston) with a 3/4-inch throw and high magnetic damping was chosen. A larger cone would offer better bass performance, but at the expense of midrange linearity because of mass limitation. The suspension of the woofer is linear over a 1/2-inch excursion and its mechanical damping is controlled by the compliance of the spider. The voice coil form is 3/4 inch long, allowing the entire, slightly shorter, voice-coil winding to remain in the gap for the entire linear excursion of the cone. The surround is substantially more compliant, and thus the driver compliance is controlled by a single mechanical parameter, making its performance more predictable. A rubber washer lies at the bottom of the voice coil gap, preventing clatter when the cone is driven to the safe limits of its suspension and improving the longevity of the voice coil. Naab stated that this attention to detail results in a woofer with useable performance to 2 kHz, eliminating the need for crossover in the most critical part of the midrange spectrum.

Cut-away models of the woofer were passed around, and Naab explained the tweeter, a 1-inch concave dome unit with a voice-coil gap filled with ferrofluid. This ferrofluid is the same type of compound used by Allison, AR, etc. and consists of small magnetic particles molecularly bonded to a heavy oil (it is not a colloidal suspension). Thus the oil will not run out of the gap and turn the cone to mush, while leaving iron filings in the gap. In fact, the material has such affinity for the magnetic field in the gap that an eyedropper-full of ferrofluid deposited on the center pole of the magnet assembly (in production) will be pulled into the gap with no residue. The fluid correctly centers the voice coil in the gap, for assembly, and tends to maintain centering through excursion, as well as improving dissipating heat in operation. The improved heat dissipation allows the use of a lower-mass voice coil and benefits high-frequency linearity. The tweeter's concave (rather than the usual convex) dome allows for closer positioning of the grill material and thus less cabinet-edge obstruction of the tweeter's sound field.

Naab made one caution regarding speakers using ferrofluid. When it gets cold the ferrofluid becomes stiff, and below freezing the voice coil may refuse to move at all. This probably rules out Genesis speakers for winter P. A. work or for those energy-conscious folks who keep their thermostats at 32°. Naab recounted the case of a rep who arrived at a Wisconsin dealer's one morning this past winter. Removing the speakers from his car's trunk, where they had spent the night at 5° below zero, and setting up the demo, they found serious loss of highs. Two spare tweeters from his warmer briefcase solved the problem, and the "defective" units were quite operative by the time they were returned to the factory. Naab took this opportunity to mention that all drivers are checked for uniformity against a standard, so that dealers can stock replacement drivers and not be concerned about a substitute's altering performance. All products are covered by a lifetime, non-transferrable warranty. (Lifetime of the manufacturer.)

The Model I two-way system is a 0.9 cubic-foot acoustic suspension design with a crossover frequency of 1800 Hz. The tweeter is connected through a 12 dB/octave high-pass network; the woofer's mechanical rolloff acts as its low-pass filter. The design is critically damped, having a Q of 0.7, and efficiency is 89 dB at 1 meter for 1 Watt input. The drivers are in phase at crossover, and the tweeter is balanced for flat energy response.

The Model II system is larger in size, uses the same woofer and tweeter, but contains in addition a 10-inch passive radiator. Thus, the acoustic suspension design is transformed into a tuned enclosure, allowing a substantial improvement in low-bass performance, without reducing efficiency from that of the original Model I design by adding mass to the woofer. The passive radiator has both a spider and the necessary edge suspension, or surround. Some passive radiators are designed without a spider, but this can result in twisting of the cone and concurrent nonlinearity. As with the woofer, the spider controls the radiator's mechanical damping. The excursions of the passive radiator and of the woofer are equal at 45 Hz. Below this frequency, with maximum excursion at 32 Hz, the passive radiator moves more air. As with all tuned systems, the woofer is not heavily damped by air loading at subsonic frequencies, and a low-frequency filter a la the Davis -Brinton preamp is desirable to prevent excessive excursions.

Naab then detailed the features of the Genesis III, their most recent development. This three-way system uses the Model II's components, and the even larger cabinet encompasses a separate, well damped sub-enclosure housing a 3-inch cone midrange driver and the 1-inch dome tweeter. The tweeter is enclosed at the rear but is placed near the midrange driver on the front panel to insure good dispersion and imaging. Both of these drivers are mounted flush with the cabinet edges to avoid reflections. Naab mentioned that in all the Genesis systems the drivers are placed relatively close to each other and are only slightly asymmetrical about the vertical centerline of the enclosure. Hence, the Genesis enclosures should be placed vertically for best dispersion and least interference and phase error between drivers at the crossover points. The Model III's crossover frequencies are 750 Hz and 3200 Hz, and both Models II and III have provisions for altering mid- and high-frequency levels with two three-position switches.

Following Mr. Naab's thorough coverage of Genesis design concepts and a brief break, members were treated to a short demonstration of the qualities of the Genesis Model III, or depending on one's point of view, of the unfortunate acoustics of the Sherman Union Conference Room. After a healthy round of applause for John Naab, the stage was set for:

<u>Meeting Feature #2</u>

Henning Moller, Electro-Acoustic Engineer of Bruel & Kjaer discussing the importance of phase distortion. Moller began his discussion by stating that there are many ways of defining good sound. One way is personal opinion -- whatever sounds best to a particular person. In addition to being subjective, this approach may be somewhat too inclusive: it tries to cover too much area with only one "measurement. " A better way of defining good sound is to examine some specific characteristics of sound and perhaps choose certain characteristics one can measure with instruments. One cannot choose only one or two particular parameters to measure and base all judgments of merit thereupon. Moller used the example of harmonic distortion as a parameter. If the harmonic distortion is 0.0001%, must the sound be good? Might designing this one parameter to be good make some other parameter poor? Of course, no single measurement tells the entire story, and for different devices an appropriate group of parameters must be measured, and evaluated relative to one another, before a claim can be made that a device is good or bad. Another example would be designing solely for amplitude response; phase response might be very poor. Conversely, designing for perfect phase response might result in terrible amplitude response.

Moller went on to mention that many parameters, such as phase and TIM (TID, SID, rate limiting), have been known and measured for years, but are periodically forgotten and then rediscovered by later generations. He stated also that many phenomena can be heard only under ideal conditions, i.e., with "good" source material and a "good" listening environment. With "poor" conditions, these same phenomena may not be noticed. Moller mentioned that many tests have been run in recent years to determine the degree of phase-error audibility. He tried to explain the results of some of these tests and to state what tests he felt to be important and how these tests, including those regarding phase, are related and should be evaluated. At this point a member raised the question of what degree of phase error is claimed to be audible. For example, is a phase error representing a time of less than two milliseconds audible? Moller replied that he feels an error of 15 microseconds is audible with good loudspeakers and good source material under some conditions.

He suggested dividing the audio band into different ranges of frequencies, with some tests being more important in one range than in another. He suggested the following categories:

1. One-third-octave frequency response in room -- ±20 to 2000 Hz
2. Phase response - 0 to 20 kHz -- and free-field amplitude response -- 2 kHz to 20 kHz
3. Gating response and impulse response -- +20 Hz to 20 kHz (with the transient performance to be measured at a distance from the transducer of one-quarter wavelength, depending on frequency)
4. Distortion: DF -3, DF -2 2 to 20 kHz; IM +2 -20 to 2000 Hz
5. TIM distortion
6. Resonances from 2 Hz to 2000 Hz

An interesting aspect of Moller's outline is that the entire system is considered important in all measurements -- from initial microphone to final transducer. He referred to this as a "global" approach as opposed to a "local" approach. A member mentioned that both the source and playback environments create far more errors to alter sound than does any of the equipment. Moller agreed that certainly many records are poor sources of good sound, as they were not correctly 'recorded with two microphones. Multi-miking creates differing times of arrival for the same bit of information, leading to smeared transients. Near field miking creates distortion both of frequency balance and of perspective.

Starting with one-third-octave frequency response using pink noise, Moller stated that he feels this is the most useful measurement, as it averages out the effects of standing waves in the room. Above 2000 Hz, because the standing waves increase in proportion to the square of the frequency, free-field response is a quite accurate reflection of sonic quality, and above 10 kHz very accurate as all sound received by a measuring microphone is direct information. Other information is absorbed by walls, etc. Phase response can be measured using a number of B&K instruments, each costing several thousand dollars. The complexity of the system and technique was not covered, given the time required. Technical papers explaining the technique are available from B&K. Numbers are: 15-098, 15-075, 15-089, 15-090 and 15-107. Gating response measurements are made using tone bursts with a synchronized gate sampling the information before room reflections become a factor. Conversely, one can also measure the room response by evaluating the reflections, and one can measure the speed with which the transducer responds and ceases to respond to impulses.

Having covered the "linear" responses of systems, with the emphasis on transducers, Moller went on to define "non-linear" responses, or distortion. He feels that two-tone distortion tests are most informative, in that music is never a single frequency. He suggested that DF -2 and DF -3 should be measured for frequencies between 2 and 20 kHz. Using two higher-frequency tones 100 Hz apart, DF -2 would be the 100-Hz difference product, and DF -3 would be the lower tone less 100 Hz. For lower frequencies he suggested IM +2. The distortion products are approximately the same, with IM +2 representing the sum of the two frequencies used and IM implying the SMPTE test format but including sweeping the higher frequency through the range from 2 kHz to 20 kHz. Some members raised questions at this point as to whether single-tone tests do not reveal the same differences of quality with somewhat lower absolute number values or percentages. Moller used magnetic tape non-linearities as an example of a gross discrepancy between harmonic distortion at a single frequency versus the difference-tone DF -3 distortion product. When pressed further, he stated that some amplifiers when fed 30 kHz and 31 kHz reproduce an audible 1 kHz distortion product. Another member countered that neither of these frequencies is found in normal program material and suggested also the use of a filter. Moller suggested that the ultimate filter would be to turn off the amplifier; this would solve all problems which might be generated by transients containing ultrasonic information. These comments place him solidly in the pro-TIM camp, although he feels that response in excess of 40 kHz probably is not necessary. He mentioned that 40 kHz is also the limit of response of the B&K 4133 measurement microphone.

Moller completed his definition of six categories by mentioning resonances, including tone arm resonances, record warps, surface irregularities and off-center spindle holes. These low-frequency problems, though not audible in themselves, can lead to amplitude- or frequency-modulation effects in the audible range. Room-wall and speaker-cabinet resonances were also mentioned.

Moller also discussed some papers and presentations by Moir in England regarding phase audibility. Moir's first demonstration used a sine wave with third and fifth harmonics added. When the phase of the harmonics was changed, the waveform on the oscilloscope changed, but not the sound. However, with continuous tones, the greatest relative phase shift created was only about 60° and was marked by phase shifts inherent in the equipment used. Moller mentioned that in Moir's second demonstration of the inaudibility of phase, the monitoring system and program material had phase error sufficient to hide the phase shift being demonstrated, which was only a few hundred degrees. The recording process and microphone techniques used for the recordings, introduced several hundred degrees phase shift..

At this point there were some questions regarding phase errors in average loudspeakers. Moller suggested that several thousand degrees is common but that several thousand degree phase shifts are not uncommon in multi-mike/multi-track recordings. He discussed the extreme phase errors obtainable by overdubbing and by having different microphones picking up the same impulse at different times.

Moller then described Moir's most recent experiment, this year, which consisted of generating a transient by bashing two rocks together. The principal difficulty in hearing the phase shift this time was that the electrical network used to create the phase shift shifted the entire body of information. No other frequencies remained as a reference, because a 4-kHz high-pass filter was used between the "click" and the monitor system. The phase error could not be heard as a time delay until it reached approximately 3 milliseconds, which corresponds to several thousand degrees of phase shift. The (presumably erroneous) assumption was made that several thousand degrees of phase shift is required for it to become audible. A member asked whether "correct" relative phase is audible only because the human ear is non-linear in terms of its response to compression versus rarefication. He questioned whether the ear can distinguish qualitatively although tests have indicated one can hear a change on an A-B comparison, neither condition sounding "correct." Moller mentioned other tests which have shown the audibility of phase changes as small as 20° with asymmetrical waveforms. He did not explain these tests in detail, however, as time was running short.

Moller ended his presentation by describing a "guaranteed" test to show the audibility of phase error. This procedure (shaggy-dog story) consists of moving the strings of an orchestra further and further away, while increasing their level to maintain balance. When the strings are 100 kilometers away, the conductor is signaled to rest. The fact that one can hear the strings playing for another five minutes shows the amount of phase shift inherent in the experiment.

Mixed groans and applause indicated the end of another BAS meeting.

-- Scott Kent (Massachusetts)

A B.A.S. User's Report

The Audio/Pulse Model 1 Digital Delay System

Gerald Larsen

No sooner had I seen an Audio/Pulse on my dealer's shelf than, like the proverbial Pavlov's dog, saliva started running, and I had to have one. This meant obtaining much additional equipment, but the Audio/Pulse was an idea whose time had come.

It's been slightly over two months since I installed the unit, fair time in which to appraise its various features and to offer some thoughts and comments to other BAS members.

My Audio/Pulse is hooked into an all-McIntosh system. The input comes from a C28 preamplifier; the primary output goes to an MC2125 amplifier; the secondary output goes to an MC2105 amplifier. The four speakers used are all the new McIntosh XR5's. Speaker placement is according to Audio/Pulse suggestion. The two main speakers face the main portion of my listening room, while the secondary speakers are at either end of the room, facing the center. Because the secondary speakers are used primarily for effect, they needn't be of the size or quality of the primary speakers, nor need they be driven by an amplifier whose power is as great as that of the primary amplifier. In fact, Audio/Pulse suggests a secondary amplifier whose power capabilities are 25-50 percent of the primary amplifier's (given speakers of equal efficiency).

Audio/Pulse does suggest that the secondary speakers should have the same "coloration" as the primaries to achieve the "best" sound. It is hard to evaluate this without trying multiple arrangements, but my listening tests reveal that an Audio/Pulse could be conveniently attached to a secondary system using relatively inexpensive speakers and a low-powered amplifier. The secondary circuits sharply roll off frequencies above 8 kHz, so super response seems uncalled for in the secondaries. This can be an advantage in that it makes it less costly to get into a time delay configuration than you might first imagine.

After two months of listening to the Audio/Pulse and fiddling with its controls, I can unequivocally claim great pleasure from the unit. The ambience effect is nothing short of magnificent and can substantially enhance almost anyone's listening enjoyment. The unit is well-made, and the lengthy, detailed and well-written instruction manual is a good example of what manufacturers ought to do when they produce an expensive ($600) audio product. My listening tastes run to classical, folk and rock music. All three sound materially enhanced and more enjoyable when played through the Audio/Pulse.

Few dealers properly demonstrate the Audio/Pulse, but once you set one up yourself, it is remarkably easy to demonstrate to any visitor the pronounced positive effect of ambience generation. Everyone who listens to my system is genuinely excited, and many friends are eagerly saving their pennies for an Audio/Pulse.

Using the Audio/Pulse, you can create almost any sort of ambience effect, from a small listening area to a cavernous cathedral. The latter seems very unnatural, and though it makes an interesting demonstration, it is not pleasant or reasonable for normal listening. The adjustment controls on the Audio/Pulse let you set the delays for specific types of music according to what sounds best to you. An interesting phenomenon is that, after using the Audio/Pulse for a while, one finds its absence makes the music seem strikingly dull and flat. When used properly, the ambience effect is barely noticeable, and the secondary speakers seem not to be playing at all. This in accordance with Audio/Pulse instructions.

The comments which follow notwithstanding, the Audio/Pulse Model 1 is an outstanding unit, one which I highly recommend. For those who purchase one, and (perhaps) for the Audio/Pulse people, the following are some comments on Audio/Pulse operation which, to my mind, could stand substantial improvement:

1. For the Audio/Pulse to perform properly, the input signal level must be adjusted using a set of "level-match" buttons located on the front of the unit. The input signal level is matched to the Audio/Pulse circuitry by observing an LED display and adjusting the buttons accordingly. Although I realize there are several different ways of connecting the Audio/Pulse into a stereo system, it seems to me that this manual level matching could have easily been accomplished by transparent circuitry. If the levels can be matched by an LED display, then the signal levels can certainly be sensed and automatically adjusted internally. The buttons are annoying and seem superfluous, particularly as they need readjustment each time the loudness control on the preamplifier is moved. (A change in preamplifier loudness control alters the output signal level and requires a commensurate readjustment of the level match on the Audio/Pulse.) I object to this readjustment in part because the push button switches, when used, tend to inject their own peculiar brand of thumping and noise into the system. If we must have level adjustment buttons, they should have more sophisticated circuitry to eliminate this noise. A final comment on these push buttons: the range of level matching possible on the Audio/Pulse is insufficient to permit really loud music to be played through the unit without overloading it. Even at the highest attenuation level, I can overload the Audio/Pulse when trying to play loud rock music. As I am not using underpowered equipment, this seems an abvious deficiency in Audio/Pulse design thinking.

2. The manual is insufficiently cautious about the way in which the Audio/Pulse must be hooked into a system. Whatever you do, <u>don't hook an Audio/Pulse into a switched outlet</u>. The attendant turn-on and turn-off transients can damage your amplifiers or your speakers. Audio/Pulse mentions that the unit should be hooked into an unswitched outlet, but doesn't give enough emphasis to this important point. Even allowing for this, insufficient caution and perhaps insufficient circuitry are available to handle certain amplifier characteristics. For example, when I switch the system off, the Audio/Pulse is left in an on. If I wait several seconds, then switch off the Audio/Pulse, a rather large thumping sound is heard through the secondary speakers. This thumping can be eliminated by moving the secondary slider controls on the Audio/Pulse down to zero and is a consequence of the design of the McIntosh 2105 amplifier and of the lack of protection in the Audio/Pulse. When I first hooked up the unit, I managed to blow out a transistor on the MC2105 with a transient, so I suggest caution. I hope that in the future Audio/Pulse will do a better job of putting time delays and other cut-out circuitry into their unit. (I wrote a letter to Audio/Pulse mentioning this particular feature of their unit but have never received a response.) My suggestion is to be extremely cautious and to experiment with the effects of turn-on and turn-off transients when using the Audio/Pulse. The manufacturer suggests that perhaps you should just leave their unit on all the time. This seems wasteful and silly -- a last-ditch effort to get around a defect in their own circuit design. Apart from the potential for damage to speakers and amplifiers, a system should be capable of being turned on without a long, formal, start-up procedure.

3. It's obvious that the digital circuitry in the Audio/Pulse allows many signal activities to be detected and acted upon. It seems possible, therefore, that this same circuitry could also be used to detect clicks and pops in incoming signals and act effectively to remove them. The addition of a pop and click eliminator would enhance the overall utility of the Audio/Pulse.

Apart from the deficiencies I've noted, the Audio/Pulse is an outstanding unit, worth the money, which can add to your listening pleasure tremendously. The product is well-made, the manual is excellent, the physical design of the unit is tasteful, and the ambience effect is clearly noticeable and very pleasant.

A B.A. S. User's Report

Shure M615AS Equalization Analyzer

David Glasser

The 615 is an intriguing tool, providing real-time octave-band frequency response analysis in a low cost (under $500), compact unit. It is meant to be used in conjunction with an octave equalizer.

The 615 provides a stable pink noise source at mic (lo or hi Z) or line level which is fed into the sound system. A calibrated microphone is placed in the listening area, in a location free from annoying standing waves. The mike, which is compensated for flat response by a switchable filter in the 615, feeds the analyzer section. The frequency response is read in octave bands on ISO preferred center frequencies from 31.5 Hz to 16 kHz. Two rows of LEDs indicate frequency aberrations above and below a variable measuring window, which is gradually closed from 12 dB (±6 dB) to 2 dB (±1 dB) as the filters on the user's octave band equalizer are varied to extinguish the LEDs, indicating flat response within the tolerance of the measuring window. When all the LEDs are extinguished in the narrowest measuring window position, the system should be flat ±1 dB from 31.5 Hz to 16 kHz in octave bands.

With a little practice, operation of the 615 system is quite simple. As Shure warns in the excellent service and instruction manual, one should avoid feeding excessive levels of wide band pink noise through high fidelity loudspeakers. Excessive boost at the frequency extremes should be avoided as well. What Shure doesn't mention is the terrific headache several hours of exposure to pink noise can induce. Wear hearing protectors or sealed cavity headphones when tuning your room.

How does it perform? Tests were conducted with a Soundcraftsmen 20-12A and Large Advents. The 20-12A has octave centers very close to the filter centers of the analyzer. (The 615 will not perform its task with equalizers whose center frequencies do not coincide with the ISO standards.) My listening room is far from typical, with a 14' vaulted ceiling, sloping walls and a loft that functions as a very efficient bass trap. (Mid-frequency RT60, using the Fitzroy equation, was calculated at 1.10 seconds. My subjective impressions tend to confirm this.) A total of twelve curves per channel were made -- two each from three mike positions (near field, reverberant field, and at Dc) and with two speaker positions (on the floor, three feet from the floor/wall boundary and six inches off the floor at the same location). In all cases, the curves were quite similar except for some minor differences in the region from 125 to 250 Hz, which was to be expected. The resultant eq settings were dramatically different from settings I had arrived at subjectively. I was about to pronounce the room tunings a failure until I used the analyzer to tune an associate's system, also with Large Advents but with an MXR equalizer. This system, owned by Steven Colby, is in a small, fairly bright room, and both speakers are on bookcases, angled in toward the listening area. The equalizer settings showed the same trends toward attenuation of the bands centered at 63 and 250 Hz and emphasis of the two highest octaves. Using the settings obtained using the 615, Steve's system sounded excellent, without any indiction of mid-bass boominess or unnatural sibilance. Resetting my own equalizer to the curves obtained with the 165, I concluded that it was my own prejudice toward a more familiar sound that caused my initial rejection of the 615 tuning. Having lived for a while with the equalization obtained with the 615, 1 am extremely satisfied and would recommend the Shure analyzer to anyone presently using an octave equalizer in his system.

In the two rooms in which the 615 was used, the analyzer display called for attenuation more often than boost. I therefore found it necessary to deviate from Shure's instructions and to begin with all equalizer bands at around +8 dB. (This is not flat response elevated by 8 dB.) When the room tunings were completed, most of the octave bands were centered around 0 dB. Beginning with all bands at 0 dB, the equalizer did not have sufficient gain to compensate for the attenuation the analyzer called for.

I offer the following comments on equalization in general and on the Shure analyzer in particular.

1. If the peaks and dips in the system fall between the equalizer's frequency band centers they will not necessarily register on the analyzer's display. Don Davis reports in the October 1976 Syn-Aud-Con Newsletter (Vol. 4, No. 1, p. 11) that in one instance, when using the Shure analyzer, the 1/3-octave band response was ±5 dB even though the octave band response was flat ±1 dB.

2. The 615 does not provide a graphic representation of the frequency response. Without knowing the characteristics of the analyzer filters (and Shure does not provide this information) one does not know the nature of the frequency response aberrations or whether they are correctable by equalization.

3. The principal advantage of using the 615 instead of a sound pressure meter is that of having the entire passband of the system analyzed in real time. Using the 615, one can readily see the interactions of the equalizer bands as their settings are varied. When the 615's measuring window is only 2 dB wide, a slight adjustment of an equalizer band can have a pronounced effect on the frequency response up to several octaves away. This is a desirable occurrence. A well designed equalizer will exhibit so-called "combining" action, mitigating the need for extreme equalization settings of a single band when the combined action of several adjacent bands will accomplish the same result. By taking advantage of this property, extremes in equalization can be minimized, along with resultant phase shifts and group delay effects. For more information on equalization, I recommend Sound System Engineering by Don and Carolyn Davis and the Syn-Aud-Con Sound Engineering Seminars conducted by the Davis's. They should be back in the Boston area with their usual truckload of test gear and fascinating anecdotes around October.

4. If your system includes a dynamic range enhancer, such as the dbx 119, it should precede the equalizer in the signal path. The expander's discriminator circuits could be confused by the eq settings.

If anyone has had any experience with the new Ivie hand held octave-band analyzer with the LED matrix display and self-contained condenser microphone, I'd be interested in his impressions.

A Word of Caution

There is some dispute as to how a loudspeaker should measure in a room. The controversy includes the type of microphone (omni versus a more directional pattern), power response, and what the ideal curve should look like. Not knowing the position Shure has taken on this subject, I suggest that you move with caution. Find out before you spend your cash. For example, Robert B. Schulein, in AES Preprint No. 863 (E-4), suggests that a speaker's frequency response should be 10 dB down at 10 kHz when measured on axis, in a room, with an omni-directional mike and pink noise. (At the time of writing, Schulein worked for Shure.) Henning Miller, in a Bruel & Kjaer Application Note, Relevant Loudspeaker Tests in Studios, in Hi-Fi Dealers' Demo Rooms, in the Home, Etc., holds that -5 dB at 10 kHz is appropriate. Both conclude that a speaker should measure flat on axis only in an anechoic chamber, but they totally disagree as to what is the proper frequency response below 1 kHz. Miller feels a bass rise is subjectively more pleasing, while Schulein opts for a flat low end.

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A Publication of the B. A. S.

Musical Frequencies

David Ranada

The following is a table relating musical pitch to frequency based on A4 = 440 Hz. It was done mainly as an exercise in computer programming and text editing, but I have decided that it is interesting enough to warrant publication. But before you start drawing conclusions from it, several caveats are in order.

None of the printed pitches is necessarily what you will hear. They were calculated using equal temperament, in which musical pitches are separated by the ratio of the 12th root of 2. Few instruments are designed, much less played, in equal temperament. Even in the case of the piano, that paradigmatically temperamental of all instruments, the strings are not tuned to exact equal temperament frequencies. The inharmonicity of the strings moves the generated harmonics away from true integral multiples of the fundamental. Good tuners (instrument tuners, that is) account for this in the process of "stretching" a keyboard: adjusting the low notes downward and the high notes upward to make the piano "seem" in tune. It should also be remembered that equal temperament is not the only musically valid method of compromising musical necessity with the laws of physics. Various older "meantone" temperaments, for example, may be "off" the equal temperament frequencies by as much as a few Hz in the middle frequencies (A4-A5).

This chart also does not take into account the rise of pitch through history. It is generally known that the standard pitch in the 18th century for non-church music was about 1/2 tone below that of today. This means that instead of A4 equalling 440, it would have equalled about 415 Hz. Many of the ensembles using "authentic" instruments tune to below modern standard pitch. What is less well known is that the pitch in the 19th century sometimes got up to 1/4 tone above "normal" in places. A4 would have been about 452 in these cases. Even today there is a push to raise the standard frequency; the Boston Symphony plays at A4 = 442 Hz, for example. When dealing with organs the pitch situation becomes even more complex since organs have traditionally been made (until the 20th century) to sound above the prevailing normal pitch, sometimes more than a halftone high. It should be obvious by now that the absolute values of the calculated pitches should be taken with grains of salt of varying sizes.

Another blatant omission is any mention of wind and brass instruments. This is because the variety of keys and ranges of these instruments is so great that it precludes anything but the most cursory mention of them. Another omission is the entire percussion family. Most of the members of this group have such ill-defined pitches that any attempt to specify them would be so misleading as to be false. Suffice it to say that any piece with a bass drum contains frequencies well below 40 Hz, and any piece with high metallic instruments (bells, cymbals, glockenspiels) would have considerable energy at high (> 10 kHz) frequencies. Hall resonances, some at frequencies so low that they approach the barometric, have also not been listed.

What can one learn from a chart such as this? Firstly, one can generally assume that anything in the music signal above 5 kHz is a harmonic of a lower pitch. This is not true of organ stops, however. The short pipes of mixture stops, for example, are tuned to extremely high pitches only to reinforce some harmonic of a particular lower note. These mixture pipes also emit their own harmonics. But though most of their harmonics are out of the audible range, such high frequencies might be significant to equipment because of intermodulation effects.

Most of the conclusions can be drawn about the low frequencies. We can see that for 90% of the orchestral music written before about 1880 (without bass drums, tubas or organs) all of the music lies above 40 Hz. This implies that a sharp cutoff highpass filter can have its -3 dB frequency up to about 35 Hz with theoretically no possible degradation of the musical signal. After about 1880 the low C1 on the double bass became more popular with players and composers. For the Bach Sonatas & Partitas for solo violin the -3 dB frequency can be moved up to at least 150 Hz; there are simply no musical frequencies which fall that low. As usual, the organ has both the highest and lowest fundamentals available to the musician, not counting electronic instruments.

Note also that the degree of precision here is far from necessary. Even the best instrument tuners cannot tune to better than the nearest 0.05 Hz at mid frequencies (A4-A5).

The pitch names used here are the standard used in acoustical measurements. Unfortunately, most music texts use differing, illogical and confusing systems. This one is the most logical and the easiest to learn.

Pitch	Frequency	
C 0	16.3515	< Lowest 32-ft. organ stop pedal note
C# 0	17.3239	
D 0	18.3540	Approaching the "integration time" of the ear
D# 0	19.4454	
E 0	20.6017	
F 0	21.8267	
F# 0	23.1246	
G 0	24.4997	
G# 0	25.9565	
A 0	27.5	< Lowest note on the modern piano Debussy: L'Isle Joyeuse, last note
A# 0	29.1352	< Lowest contrabassoon note
B 0	30.8677	
C 1	32.7031	< Lowest (normally tuned) modern double bass note Mahler: Das Lied von der Erde, Abschied, start Lowest 16-ft. organ stop pedal note Strauss: Also sprach Zarathustra, start
C# 1	34.6478	
D 1	36.7080	
D# 1	38.8908	
E 1	41.2034	< Lowest (normally tuned) 18th and 19th century double bass note and therefore the lowest possible fundamental of 90% of all orchestral music written before about 1880
F 1	43.6535	< Lowest large 18th century French harpsichord note
F# 1	46.2493	
G 1	48.9994	
G# 1	51.9130	
A 1	55.0	
A# 1	58.2704	< Lowest bassoon note
B 1	61.7354	
C 2	65.4063	< Lowest (normally tuned) cello note Lowest 8-ft. organ stop pedal note Lowest bass trombone note
C# 2	69.2956	
D 2	73.4161	
D# 2	77.7817	
E 2	82.4068	
F 2	87.3070	
F# 2	92.4986	
G 2	97.9988	
G# 2	103.8261	
A 2	110.0	
A# 2	116.5409	
B 2	123.4708	
C 3	130. 8127	< Lowest (normally tuned) viola note

Pitch	Frequency	
C# 3	138.5913	
D 3	146.8323	< Lowest clarinet in B-flat note
D# 3	155.5634	
E 3	164.8137	< Lowest English horn note Lowest B-flat trumpet note
F 3	174.6141	< Lowest easily obtainable French horn in F note
F# 3	184.9972	
G 3	195.9977	< Lowest (normally tuned) violin note
G# 3	207.6523	
A 3	220.0	
A# 3	233.0818	< Lowest standard oboe note
B 3	246.9416	
C 4	261.6255	< "MIDDLE C" Lowest standard flute note
C# 4	277.1826	
D 4	293.6647	
D# 4	311.1269	
E 4	329.6275	
F 4	349.2282	
F# 4	369.9944	
G 4	391.9954	
G# 4	415.3046	
A 4	440.0	< Standard pitch for instrument and orchestra tuning
A# 4	466.1637	
B 4	493.8833	
C 5	523.2511	< "High C" for a tenor
C# 5	554.3652	
D 5	587.3295	
D# 5	622.2539	
E 5	659.2551	
F 5	698.4564	
F# 5	739.9888	
G 5	783.9908	
G# 5	830.6093	
A 5	880.0	
A# 5	932.3275	
B 5	987.7666	
C 6	1046.5022	< "High C" for a soprano
C# 6	1108.7305	
D 6	1174.6590	
D# 6	1244.5079	
E 6	1318.5102	
F 6	1396.9129	< Highest note for the Queen of the Night in Mozart's Magic Flute
F# 6	1479.9776	
G 6	1567. 9817	< Highest 8-ft. harpsichord stop note
G# 6	1661.2187	
A 6	1760.0	
A# 6	1864.6550	
B 6	1975.5332	
C 7	2093.0045	
C# 7	2217.4610	
D 7	2349.3181	
D# 7	2489.0158	
E 7	2637.0204	
F 7	2793.8258	
F# 7	2959.9553	
G 7	3135.9634	
G# 7	3322.4375	
A 7	3520.0	

C

Pitch	Frequency	
A# 7	3729.3100	
B 7	3951.0664	
C 8	4186.0090	‹ Highest note on the modern piano; 99.9% of all written orchestral instrument notes lie below this frequency; therefore most musical signals above this frequency are harmonics of lower pitches
C# 8	4434.9220	
D 8	4698.6362	
D# 8	4978.0317	
E 8	5274.0409	
F 8	5587.6517	
F# 8	5919.9107	
G 8	6271.9269	
G# 8	6644.8751	
A 8	7040.0	
A# 8	7458.6201	
B 8	7902.1328	
C 9	8372.0180	
C# 9	8869.8441	
D 9	9397.2725	
D# 9	9956.0634	
E 9	10548.0818	
F 9	11175.3034	
F# 9	11839.8215	
G 9	12543.8539	
G# 9	13289.7503	
A 9	14080.0	
A# 9	14917.2403	Approaching the limits of audibility
B 9	15804.2656	
C 10	16744.0361	Shortest pipes in organ mixture stops produce pitches in this area
C#10	1773 9.6883	
D 10	18794.5451	
D#10	19912.1269	
E 10	21096.1636	