THE BAS SPEAKER

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Volume 17 Number 1

Open Forum 2
"Am I Current?" 2
Audiophiles—Go Figure!?!?! 2
CD Review 3
A Statement from Clark Johnsen 3

September 1988 BAS Meeting 4
Open Forum 4
Meeting Feature: Keith Johnson and Precise Acoustics 5

October 1988 BAS meeting 12
Open Forum 12
Meeting Feature: John S. Allen 13

Advertisement 16
For Sale 16

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Open Forum

"Am I Current?"

The publication you hold in your hands is the first issue of Volume 17 of *The BAS Speaker.* If the top line of the mailing label says ***PLEASE RENEW—LAST ISSUE,*** we have not received (or processed) your renewal for this volume, and this is the last issue you will receive unless you renew. If you have renewed, the code line should contain a 17.

Audiophiles—Go Figure!?!?!

It is amazing what we will do for good sound and/or music.

On a recent Tuesday night I received a call from Ken Rudnick, a Boston-area resident who had been BAS Membership Secretary for several years. Within the previous few days he had heard both the Boston and Worcester concerts of the Munich Philharmonic Orchestra under the baton of Sergiu Celibidache (which Ken tells me is pronounced chell'-i-bi-da'-cheh or chell'-i-bi-da'-chee, depending on the reference). Ken said that if I could get tickets to the Washington, D.C., performance, he would fly down to attend! I did, and he did. Ken said, tongue-in-cheek, that he couldn't believe what he had heard at the previous two concerts, and wanted to attend this one for confirmation.

After an excellent dinner in the Rooftop Terrace Restaurant at the Kennedy Center, we went downstairs to our Concert Hall seats, which were Orchestra, Row X Right, #14 and #16. These are one-sixth the width of the hall from the right wall and about three-fourths the length of the hall back from the stage. They are not under an overhang, as the balconies are quite shallow. I think the hall, including the three balconies, seats around 2800. The stage has no shell and the top of the stage is continuous with the ceiling of the hall. There are no BBN 'clouds', either.

The orchestra used three levels of risers, with the timpani and bass drum on the top tier at the center rear, a most unusual placement. Of course, the skins of the bass drum were essentially parallel to the sides of the hall. The harps were where the timpani usually are placed—stage right rear. Otherwise, instrument positions were rather traditional.

I have never enjoyed the sound in this hall, and this experience was no exception. Even with 11 double-basses during the Mussorgsky-Ravel *Pictures at an Exhibition,* the bottom end was marginal. The bass drum was present but not strong, and the timpani were insubstantial. There are several resonances in the hall, but the degree to which they stood out surprised me. One was a particular string on the harp. Another came from the bassoon. Positively speaking, the sound of fortissimo orchestra was never constrained (as though in too small a box). Also, the hall is pretty quiet, and when a conductor actually gets the musicians to play pianissimo, one can hear the music clearly if the audience is quiet enough.

MUSIC! The reason for this expedition. I do not think this orchestra can be ranked as one of the world's finest symphonic assemblages, but under Maestro Celibidache's conducting, the MUSIC is amazing. I don't ever recall hearing, as a member of a concert audience, a string section produce such a silken sheen and melodious tone throughout the entire concert! This string section bows together as though a single player seen through a fly's eye. The brasses were excellent, with few burbles even on low-volume entrances, and displayed a smoothness and trueness of tone which was remarkable. The rest of the orchestra played quite well, but these two groups stood out, both solo and en masse. Ensemble, if not perfect, was as good as the finest I have experienced.

I prefer different interpretations, but I give Maestro Celibidache credit for making a good case for his choices and keeping to his decisions throughout individual works and the entire concert. The interpretations were extremely lyrical and romantic, but seemed drawn out to expose every nuance. I marveled at the inner detail that I heard in all three pieces (Ravel *Rapsodie espagnole,* Strauss *Don Juan,* and the Mussorgsky-Ravel); this is a concept I have often heard about but never before experienced so clearly. Every instrument seemed distinctly audible, even in the midst of full orchestra. Secondary themes and melodies were apparent. With few exceptions, transitions held together, and didn't sound as though the conductor got lost for a while (which is my usual perception). Although the tempi selected seemed extremely slow, the Mussorgsky took only 38 minutes, and the *Don Juan* was about 22 minutes long.

As always with a DC audience, which seems more intent on seeing and being seen than knowing what music is about, they applauded briefly, and embarrassedly, when the orchestra paused after the first section of the *Rapsodie espagnole.* Celibidache is an artist at milking applause. He stepped down among the first violins to take the applause, then began singling out musicians for solo bows; then select groups, depending on the piece; then the whole orchestra. When he was given a bouquet at the end of the concert, he started handing out individual flowers from it to select musicians. He didn't move quickly, sort of shuffling to and from the stage, and got a helping hand from a violinist in mounting the podium. His movements seemed commensurate with the apparent pace of his music.

This concert taught me a great deal about this music and about music-making. I enjoyed it immensely and hope many of you got the opportunity to hear them during their tour. I am quite grateful for Ken Rudnick's impetuous call, and glad that I could, at the same time, help him out. He spent quite a sum just to attend this concert. But after all, one cannot cost-justify one's "toys"!

David J. Weinberg [Tobias Audio]
(Maryland)

CD Review

[Equipment used: Magnavox FD1010 CD player; TEAC AN-400 turntable, Souther arm, Shure V15-V cartridge; dbx 224 decoder; Apt/Holman preamp; Apt One power amp; Stax SR-X Mk.3 electrostatic headphones; Allison CD9 loudspeakers; 39-year-old ears (2).]

The Leroy Anderson Collection (Leroy Anderson, conductor). MCAD2-9815-A/B (two CDs). AAD. 59:25 and 73:18.

Leroy Anderson's music is familiar to anyone who has ever watched "The Late Show" in New York *(Syncopated Clock),* entered a department store or listened to the radio during the Christmas shopping season *(Sleigh Ride),* or heard a Pops concert. It has become popular in both the best and worst senses of the word, falling periodically into and out of favor with people who like to hum.

I grew to like Anderson's music when I came to Boston, in the late '60s. The Boston Pops, after all, was Anderson's "home base," the orchestra and conductor (Fiedler) for whom he wrote most of his pieces. The Pops sound, in turn, was greatly influenced by Anderson's skill as an arranger, and subsequent arrangers for Pops have all copied his style to some degree.

In 1967, Maurice Abravanel and the Utah Symphony recorded 15 Anderson pieces for Vanguard (Cardinal VCS-10016), and *Fiddle-Faddle* from this excellent-sounding recording was used by Advent Corporation as part of the demo tape for their 201 cassette deck. I hauled out my copy of this LP (I have the dbx-encoded version) to compare to Anderson's own performances on these MCA CDs, because I remembered Abravanel's performances as quite lively and idiomatic.

Wow! What a difference. Anderson's experience as Director of the Harvard University Band is clearly the dominant influence in his performances of his own music. In the pieces common to both sets, all but one of his tempi are brisker than Abravanel's. The crisp playing, steady pulse, and no-nonsense phrasing all set off the wit and light heart of this music. Other performances I've heard drag by comparison.

The recordings (made for Decca) date from 1952 through 1962, with the earliest (8 tracks out of 47) in mono. The sound is occasionally congested, and obvious limiting occurs in spots. However, most of the sound is quite good, the tapes having held up well over 30-odd years. And Anderson gets very good playing indeed from his musicians (selected from New York's Local #802, according to G. W. Briggs Jr.), who seem fully up to the demands of the writing and the tempi.

An example: *Bugler's Holiday,* with a trio of triplet-playing trumpeters, takes 2:41 in Abravanel's hands; Anderson runs through it in 2:17, with nary a trip from the trumpets. The result is thrilling.

It is not possible (for me, anyway) to listen to all of these pieces at one sitting. Each short moment by itself is a well-crafted and, yes, hummable gem. Too many become too much rather rapidly, however, and the effect is compounded by the relentless energy Anderson brings to the playing. (I don't think Decca had it in mind, originally, to issue a 133-minute album.) But these are clearly the performances to have, and the sound is good enough that they might even be the only performances to have.

Just take them one by one.

— *Mark P. Fishman (Massachusetts)*

A Statement from Clark Johnsen

A negative review [Editor's note: two, actually, as well as a generally positive (I thought) one—mpf] of my first and only book, *The Wood Effect: Unaccounted Contributor to Error and Confusion in Acoustics and Audio,* recently appeared herein. If I may be allowed a rebuttal, I should say at the outset that I hold no grudge against the writer: he's a clever fellow and a genial correspondent. Nevertheless, did he tell you I refer to him as "Dangerous Dr. Dan Shanefield"?... Or that I accuse him by name of "misconstruing" facts?... Or that I refer to his "tragic flaw"?... He did not! Instead he penned what he was pleased to call a review and circulated it as an impartial criticism, which leads me to believe he's fallen prey to a severe case of something I'll identify later. At any rate, he apparently cannot hear Absolute Polarity.

Other reviewers claim I have written a radical manifesto ("Chairman Clark's Little Red Book") directed against both the Underground and the Audio Establishment and its media lackeys, employing as a Philosophers' Stone of Sound the audibility of Absolute Polarity. Indeed, the book's scientific evidence utterly demolishes the hear-nothingism of D. D. D. Shanefield, the Joint Chiefs of AES, the Hirsch Organization et al., and in a most "informative and entertaining" manner. Had he not been among the many brunts of my gentle humor, even the good Professor at Rutgers [Editor's note: Shanefield—mpf] might have gotten the point. Yet simply because he cannot hear polarity (some do not, I have learned), he seems to think everyone else must be faking it. His chill attitude is manifest in these unpleasant remarks:

> Maybe we should look again at polarity, in case there's something really useful hiding there. But so far as this book is concerned, if you can believe that the Japanese record companies would bother to do that every-other-band thing, I guess you can believe the rest of it.

I ask you, is that an outstanding example of scientific inquiry, or what?

So another 'Jersey bites the dust. Despite DDD, and hard to believe as it may be, most Japanese record companies do indeed alternate polarity on sequential bands of their vinyl records. [Publisher's note: can you prove it?—drm] (Conveniently he omits my corollary finding, that their CDs do *not.)* Poor chap, if he could but hear the phenomenon he reviews, he would be forced to agree. Instead the erstwhile Bell Labs doctor chooses to bludgeon an unknown, unlettered author who speaks the Holy Truth. Well, he should have picked a harder cudgel, because his has shattered badly.

Proof: In the September 1982 issue of the *Journal of the Audio Engineering Society,* Stanley Lipshitz, Professor of Applied Mathematics and current President of the AES, reports on the *double-blind tests* he administered to members of the SWMWTMBBD&O (I hope I have that right, they'll be reading this) [Editor's note: Johnsen is probably referring to the Southern Michigan Woofer & Tweeter Marching Society—mpf], in which he states:

> The authors have demonstrated the two-tone experiment described above to numerous people on different systems. No one has ever failed to hear the timbral change with phase, or discern the polarity reversal on the signal. Indeed, in a double-blind demonstration, the accuracy score was 100% on the summed 200Hz and 400Hz tones over loudspeakers, and over-all, including musical excerpts, the results on the audibility of the polarity inversion of both loudspeakers represented a confidence of more than 99% in the thesis that acoustic polarity reversal is audible.

For all the good that did in Jersey! DDD demands more! Very well: more proof. The September 1970 *IEEE Transactions on Audio and Electroacoustics* includes a long (and brilliant) article by David Stoldowsky entitled "The Standardization of Monaural Phase," wherein he reports, as summarized in the abstract

> A recent experiment evaluates Absolute Phase (polarity error) with reference to amplitude distortion. It is concluded that at high sound pressure levels [at least], Absolute Phase error is more detectable than 11.5 percent intermodulation distortion.

Isn't that special?

It's all in my book, but those two outstanding voices of reason and one hundred others who have spoken up to defend Absolute Polarity are disregarded by Dr. S. in his transparent attempt to diminish yours truly and an honorable thesis. Read for yourself and see whether you don't agree, his "review" bears scant resemblance to the reality. Thus I suggest that poor D. Shanefield must be suffering from, for lack of a better phrase, *intellectual bankruptcy*. [Editor's note: Clark, this strident *ad hominem* attack is beneath you and does your case no good at all.—mpf]

The Wood Effect, a "fascinating *tour de force"* [Editor's note: according to John Atkinson—mpf], may be ordered directly from the publisher [Editor's note: i.e., the author—mpf] for $7.50 postpaid. Address: Modem Audio Association, 23 Stillings Street, Boston, MA 02210.

— *Clark Johnsen (Massachusetts)*

[Publisher's note: Really, Clark Johnsen has little business accusing anyone of intellectual bankruptcy. The scientific error, wild claims, savagery, and deaf grammar in his screed all proclaim that his ear (among other qualities) is hardly to be trusted.--drm]

September 1988 BAS Meeting

This meeting was held jointly with the Boston section of the AES. Ira Leonard gave a brief summary of the BAS's finances to start off, saying that the costs of Volume 16 are expected to be about $5000. This was broken down into $3000 for printing and mailing, $500 for honoraria, $1200 for meeting reports, and $400 for ads, publicity, printing expenses, and miscellaneous. Members present decided to hold dues at $22 for the Volume. [See Treasurer's report in Volume 16 #6.]

Open Forum

The BAS was saddened to learn of the passing of two of its members. Ed Farnsworth and his wife died in a light-plane crash, and Bob Fulton (of Fulton Audio) died of cancer.

Allison Acoustics has been refinanced and reorganized. Bob Barr, Ron Falkenstein, and some others from Acoustic Research have bought equity in the company; Roy Allison continues to own part of the firm, but the rest of the original management team has gone. We expect the company will be repositioned in a search for better sales. With Roy still there the speakers should retain their familiar quality. [Editor's note: Allison has shown the new AL series of loudspeakers to the BAS at a recent meeting (report in a future issue)—all of this is true.—mpf]

Stereophile had an editorial by John Atkinson suggesting that Dan Shanefield, David Clark, and Doug Self are wrong about the nature of proof, and praising Clark Johnsen as a defender of the importance of things that are only slightly audible (if at all) to many people but appear to be crucial to others. This editorial also referred to David Moran's quoting of *IAR* to the effect that "clarinets would suck instead of blow" if loudspeakers had improper polarity. Moran said that this ought to be meant as a joke: sucking on a clarinet would not produce music. Moran also said that he had checked himself for the audibility of absolute polarity at great length using non-blind, single-blind and double-blind tests and had not been able to hear a difference. He characterized difference in polarity as like being in front of the timpani versus in back of them.

Scott Kent pointed out that the common term "absolute phase" was not correct and that the correct form is "absolute polarity." In a paper in the *Journal of the Audio Engineering Society* (1978) Lipshitz and Vanderkooy proved that absolute polarity is audible for some musical signals and for some speakers. They used electrostatic loudspeakers pulled well away from the wall. They could not differentiate with some other speakers nor with some other speaker placements, however.

Kent pointed out that possibly audible differences in absolute polarity do occur in some points in audio. For example, the control signal of Dolby B is derived from a half-wave rectifier, so that Dolby tracking could conceivably differ depending on signal polarity and the change in Dolby tracking could be audible by changing broad-band frequency response.

There were also a series of tests reported in the BAS *Speaker,* done primarily by Al Foster, which showed that a change in polarity was audibly detectable (although listeners could not tell which way was the "correct" one).

Kent also noted that the detection of polarity depends on the recording, with coincident recording being more sensitive to it. On multi-mike recordings it is not possible to tell. It is also more easily detectable on vocal music than on piano. Because he currently has a large number of students at the University of Lowell, he can say that this also varies with the individual. About a third of his students could hear absolute polarity as an obvious effect, while the rest could not. He was not sure of the effect of training on this sensitivity.

A confounding problem is that he was not sure if the whole experiment had ever been set up correctly with the entire chain correctly phased from the microphone through the tape recorder and to the speakers.

He suggested that given a choice an engineer would pick the tape with the "most flattering depth" [assuming the engineer wanted or even could detect depth] as the "correctly phased" tape.

Clark Johnsen mentioned that "monophonic phase" effects (i.e., absolute polarity or single-channel phase shifts, even of less than 180 degrees) were studied at Harvard in 1952 and were reported in 1970 in the *IEEE Transactions on Audio.* These effects were audible using both headphones and speakers, he reported.

Micha Schattner commented that some phase effects are easier to hear over loudspeakers than through headphones because the reflection from the nearest wall is a constant phase shift against which you can compare the change.

In support of this observation, Kent noted that Dolby B mistracking is more obvious over speakers than using headphones.

It may be easier to hear radical differences with changes in polarity using speakers because a speaker is a non-linear device: an input pulse causes distortion which may be different pushing and pulling (due to mechanical asymmetries). In this case there may be a difference but it is an artifact caused by the speaker's not being ideal (i.e., linear).

For anyone interested, the tape out of the DB Systems preamp is polarity-inverting, so it can be used to test the audibility of the effect by connecting tape out to the tape in and switching from source to tape. Also, the Adcom CD player has a polarity switch on the front panel which inverts the signal while it is still in the digital domain. The analog stages are exactly the same for both polarities, making the test even purer.

Meeting Feature: Keith Johnson and Precise Acoustics

The evening's main speaker was Keith Johnson, perhaps best known for his analog tape recorder, which he has used for several years on recording sessions done for *Reference Recordings.* He was also part of the *Sheffield Records* Moscow recording team.

Johnson has a BSEE from Stanford with minors in Physics and Music. He is the principal designer of the Precise Acoustics loudspeaker line, and has done filtration work on Spectral's high-end CD player.

As a recording engineer, he thinks that recordings deal with people (musicians, producers, and listeners) and thus involve complex human values. Music is animal, ethereal, and spiritual. As music is many-dimensional, so there is more than one optimum way of recording and playing it back.

A recording is "a trendy time capsule," and the concept of what is desired in recording and playback has changed over the years. The physics and aesthetics involved both prevent an ideal system from being used. For example, the first acoustic phonographs were thought to be nearly perfect because people could listen to a Caruso recording and identify the voice as belonging to Caruso. This is a situation where what is thought to be possible, what has not been done before, is also thought to be both perfect and aesthetically acceptable, not to say ideal.

In the period between Caruso and digital cannon, there have been major changes both in the presentation of the sound and in the desired effect. In early recordings the point was to be able to identify the voice, while the cymbals sounded more like two loaves of bread being slammed together. Today, recordings try to present the sound of the voice in a particular acoustic space, and cymbals sometimes sound like a spray can of flammable solvent being lit off. Another way of expressing the aesthetic change is that early recordings tried to place the artist in your room, and modern recordings try to transport you to the event as it occurred.

Johnson proposed that the ideal system would have "speed" and dynamic range that "just don't quit" and directionality characteristics that change with the instrument being reproduced. It is possible to create only an illusion with two static channels, he said. It was not immediately clear to some listeners why the directional characteristics should be dynamic, however, since (if the current aesthetic is taken as a given) we are trying to reproduce the soundfield of a hall whose "directional" characteristics are independent of the originating instrument.

In Johnson's opinion, "sonic nirvana" requires imaging, transparency, spatiality, glow, power, jump factor, tightness, dynamic range, roundness, depth and so on. The recording and the reproduction must be matched to create a sensory illusion.

A pattern in the audio industry, Johnson asserted, is to decide that certain performance parameters are important because of their influence on audible sound quality, then to decide that if a component reaches certain standards in those parameters, it must sound good. This process starts with the real goal—good sound—and then steps back to an intermediate goal—having good measurements.

The problems really become apparent when new technologies are involved. An example of this is digital audio: Johnson feels that there were glaring flaws at first, but it was quiet and seemed to have low distortion, so it measured well on the tests devised for analog equipment. Only recently has there been exploration of the limits of the new technology.

In a practical home system (but not a mass-market system or a true audiophile system), Johnson thinks that the sound can be quite good, good enough to ruthlessly expose the problems in digital and in studio production practices. Home systems, however, can be designed to cover up or reveal problems. Loudspeakers are the primary place this choice is made.

Much of the equipment being used by recording studios is not as revealing as the best home equipment. Studio technology is in a rapid evolution right now. The current studio design goals are: (1) reliability, (2) power, (3) more signal processing, and (4) digital recording, storage, and editing.

"Power" involves the use of "rock 'n' roll" speakers that have almost no relationship to home speakers. The moving parts are heavy and take a long time to settle. There is also a "tissue paper tradition," in which engineers put tissue paper over the tweeter to cut excessive treble.

Digital processing is silent, tight and convenient. In some studios, a good deal of the analog equipment never evolved beyond what was available in the late '50s and early-to-mid '60s, so the digital equipment is a godsend. Johnson prefers digital to the 1950s state-of-the-art found in many recording studios. However, he feels that his special analog recorder is much better. The S/N is better both quantitatively (i.e., it measures quieter) and qualitatively (i.e., the noise sounds more like room noise or benign hiss than does the noise floor on digital systems). The tape he uses does not appear to deteriorate, even after many years [Editor's note: the recording of Red Norvo's band on *Reference Recordings* was made in the 50s, and still sounds fine today. Many commercial tapes have deteriorated drastically in storage over the same period.—mpf]. In contrast, the tape used in some studios will begin to deteriorate after one or two months, Johnson claimed.

Using analog-to-digital converters requires lowpass filters to prevent aliasing. Because of the sharp cutoff required, these have to have high-Q elements, so large circulating currents are introduced in the filter. This can lead to TIM or phase dispersion of harmonics through either slew-rate limiting or magnetic effects. [Editor's note: you will get some phase dispersion near the cutoff frequency of the filter simply due to the filtering process, without regard to implementation.—mpf] The variations in group delay can be corrected at playback.

There is also a problem with the sample-and-hold amplifiers. Because the A/D converters do not work instantaneously, sample-and-hold amplifiers are used to hold the input constant until the A/D converter can do a complete conversion. These units should have an accuracy of 20-40 picoseconds (which is not currently commercially available) in when they acquire and freeze the samples of the musical signal or else, says Johnson, the time-base jitter will cause slew, aperture, and TIM distortion, which are among the worst-sounding types of distortion.

[Editor's note: A small amount of timing jitter will manifest itself as an amplitude error in each sample. I would think that it would have no more or less effect on distortion than a similar error in the converter. However, because the amount of error will depend on the rate of change of the input signal, it will be larger at high frequencies than at low for any given amount of jitter. This frequency-dependent character could well be audibly different from simple converter nonlinearity.—mpf]

A/D converters are also not as good as they should be. Common commercial units typically have only a 14-bit monotonicity. Johnson wanted 1/2-LSB accuracy over the entire temperature range for his recordings, and he could find only one mil-spec converter that was this good.

Johnson hears "grunge" wherever the A/D chain does not work well. This is like crossover distortion in improperly adjusted amplifiers, and just as distracting.

He said that, in his opinion, a 16-bit standard suffers a bit shortage. The available dynamic range is not the same as the S/N ratio. There have to be enough accurate bits to reproduce the dynamic range with an adequate S/N ratio in the quiet passages. If there are only 16 bits, then 10-bit encoding is the amount usually used in practice because most of the dynamic range of music is -26 to -40dB relative to the peak. Johnson claims that 10 bits are not enough.

[Author's note: This is hard to compare with an analog tape recorder because of the extra headroom available with analog tape. Digital recording runs into a stone wall when all the bits go to 1, and this digital clipping is very distressing to hear. In contrast, analog tape will distort the signal "softly" by compressing it when it nears the maximum output limits. The net result is that many analog recordings are distorted in their loud passages, although I think this goes mostly unnoticed. Do BAS members regularly hear differences between loud passages recorded in analog versus digital or between a digital original and an analog copy? Perhaps all digital recorders should have dbx compressors acting when the upper 2 bits are reached.—cd]

Whatever is going on, the result seems analytical to Johnson; there is a sense that only the bones remain, with all the meat picked off.

But in the studio there is a symbiotic relationship, with digital giving a tight, clean, crisp recording which is what artists and producers seem to want. Then the final mix is also digital and the overall digital artifacts communicate something which was not there at the performance. The same thing happened with older technologies. This goes back to the trendy time capsule. What we are producing now is desirable now just as Caruso's voice was the perfect voice for his era. Just as the Philco open-back enclosure and the acoustic bass were symbiotic, with the reproducer stressing the strengths of the instrument. There is a "jump factor" and a sense of being there which is better than the same musical pieces heard on modern systems. Looked at in this way, studios use digital as a special effect.

[Author's note: I always thought that the "clean" sound was the lack of flutter, and that it was more realistic.—cd]

A really advanced audiophile system runs rings around all this. Properly done, the speakers will disappear and mono should sound like one (phantom) speaker in the middle. If there is a phase difference between the two speakers then there will be a clear alteration, with the image appearing somewhere other than between the speakers.

Speaker testing and hearing

Johnson is interested in active specifications. By this he means that he takes the equipment, finds where it gets into trouble in normal use, and determines how much and what kinds of distortion are present there. He considers harmonic distortion, frequency-response inaccuracy, and dynamic-range changes to be the three major types of distortion in analog equipment. These do not completely describe the problems that he hears with digital.

For example, Richard Heyser devised a hypothetical device that has good frequency response, no noise and no distortion but speech passed through the box is unintelligible. In the black box is a computer that recognizes test tones and if the test tones are not present, the output is intermittently shorted to ground.

[Author's note: Of course, in the real world this is just an argument for more elaborate instrumentation. This type of box would be clearly detected using a filtered noise source and either a spectrum analyzer or even a simple ac meter. This points out not that our current test specifications are wrong but that they are not complete.—cd]

Something similar is true for loudspeakers. A loudspeaker is more complex than just its frequency response. [Publisher's note: Yes; radiation pattern, distortion, and phase response, as well as amplitude response. Anything else?—drm] The diaphragm that should be moving like a piston in reality could move like a bowl of Jell-o with complex behavior and random motion although the total energy output may be uniform. Because imaging is affected by the radiation pattern at various frequencies, he is concerned with this non-pistonic behavior.

For this reason, Johnson has developed Differential Stress Mode Analysis. This uses two tiny (roughly the size of two pinheads) pressure-gradient microphones which are turned sideways. This gives a figure-8 response pattern with a sharp null. If identical signals are present on each side there will be equal outputs with opposite phase so the sum will be zero. But if there is aberrant cone behavior there will no longer be a null. This can be used to map the behavior of the speaker by slowly moving the microphone pair over the surface of the driver while a test signal is being played.

The overall result is similar to the Celestion laser doppler analysis. The laser analysis is more precise, but the paired-microphone method can use a range of frequencies simultaneously and is much simpler to implement as no computer analysis is needed. Some of the analysis can even be done by listening, as a trained ear can recognize characteristic driver behavior from the microphone outputs.

Johnson reported some developments in hearing research done by the Stanford Research Institute and the U. S. Army. The highest frequency that the cochlear hair cells can respond to is in the range of 90kHz to 120kHz. The tympanic membrane will only pass 20kHz as the upper end but the higher-frequency response of the cochlear cells may be picking up the distortion products resulting from excessive sound-pressure levels. This is a dynamic, non-linear system that appears to detect loud transients to protect itself. He postulated that since the way digital rolls off the high end is different from analog, this mechanism could explain why digital is fatiguing to listen to.

Johnson appeared to be saying that this means that the 20kHz limit of audibility does not hold, but the levels are very different between music and gunshots. Also, the extreme high-frequency response must be excited by distortions generated within the ear, if the tympanic membrane in fact does not pass anything above about 20kHz.

There are now microphones which have very thin (1.4 micron or less) diaphragms. These sound better and "resolve inner detail" better although the frequency response is unchanged. Johnson feels that you can hear something related to mass.

Similarly, tweeters (ribbons, electrostatic, or domes) are lighter-weight, which gives better high-frequency resolution than the heavy horns often used in recording studios. These lightweight-membrane drivers, such as those used in his Precise speakers, can easily go out to 20-25kHz and have not only the advantage of being able to produce higher frequencies (which is unlikely to be what one hears) but also "faster acceleration." [Editor's note: meaning less stored energy?—mpfl

He also said that paper was a better driver material than polypropylene because paper is lighter and stiffer.

Another area of development of psychoacoustics is eigensonics, which is the characteristic sound of things we already know but do not want to hear while playing recordings. We recognize the sounds of objects and with experience can detect them when they are present in extraordinarily tiny amounts.

A chemical spectrograph ionizes chemicals, usually in a flame or arc, and collects the light emitted. The emitted light is separated by wavelength and may be collected on a photographic plate. Johnson said that a trained operator could just look at the plate and tell from the spectrum what trace elements were present. (I would note that this would require considerable expertise with the particular class of compounds involved.)

Similarly, with experience a listener can hear trace coloration from speakers, even if it is a discreet coloration that probably has not grossly changed the frequency response. Trained ears are more sensitive than a computer. On a good speaker system, somebody spent a lot of time moving drivers around on the baffle to get an optimal sound. When this is done the speakers sound better but the average frequency response probably has not changed. [Author's note: this implies a very simplistic view of loudspeaker frequency response for someone who has stressed radiation pattern.—cd]

What do mikes sound like?

Johnson briefly described the effects of various microphone techniques commonly used for commercial recordings. Multi-miking, he said, produces a sound which is all mixed up. This is a spoon-fed recording, the bottom of the line. It has perfect balance but the sound is constricted and the spread is all wrong, with only a left/right balance. There is no magic or aura and there is no feeling of space.

A pair of crossed cardioids (i.e., coincident) is mono-compatible and useful when the recording *must* be usable. This method yields the effect of a clothesline between left and right speakers, with the instruments hanging on it. It gives an accurate sound spread but front-to-back images are formed by sound bouncing from the stage.

The MX system is a cardioid facing forward and a figure-8 microphone sideways. The left and right channels are unmixed by a matrix. This gives a sense that the sound stage is wider than the speakers but the center image may be disconnected from the rest of the space because it is produced by the cardioid alone.

Blumlein crossed-figure-8 pattern, if properly set up, will give a sense of a large curved screen behind the speakers and if done in a proper hall the sound will appear to go beyond the speakers as well. However, this requires a great deal of care to implement correctly.

There is the problem with all the single-point mike techniques that the important center information is off the microphone axis. This puts the most important part of the sonic image into the roughest part of the microphones' frequency response. Only the very thin membrane types are really good for single-point miking.

Spaced omnis give "sound balls" around the speaker. The sound is much fuller, but it is harder to stabilize the center image.

Johnson's own miking does not fall into these categories. While he is a minimalist, he is not a purist. Most of his recordings have 6 mikes (or more) but one pair is dominant and the rest are just accent mikes to get the correct balance.

Many famous recordings have been made similarly. For example, Decca records with a tree of mikes, with the center based on either crossed cardioids or Blumlein and outriggers that are usually omni. This gives a fill image and has time information which is not just reflections from the side. Pure single-point miking only has time information from hall reflections. Listeners can hear time cues and a recording engineer can use these to generate L/R shifts by changing delay rather than amplitude.

Creative multi-miking can create a phantom sound field that never existed to surround the listener; this is still a valid way to create a surround sound system. It is a powerful method and produces good results with speakers in the $200-300/pair price range.

A binaural system using microphones in a dummy head can be one of the most realistic miking methods if good microphones and headphones are used. However, it does not work very well if the recording is played back with speakers because there is very little time difference information to provide spatial clues.

When asked how he got artists to record, he replied that it was usually very hard. An exception was Nojima ("Nojima plays Liszt," *Reference Recordings RR-25*), who came to Johnson because, as an artist, he was unhappy with his recordings. Usually the best musicians go after the big money. Johnson has also found that many musicians (now) cannot play in an acoustic space/environment. They tend to play at a mezzo-forte continuously and expect the recording engineer to change levels as appropriate.

While discussing the Moscow recording sessions (done with Sheffield), he said that Melodiya uses a tonmeister system with multiple mikes and that the musicians were used to performing under this system. He used a minimal mike setup like that used by Decca-London. They did an initial runthrough to set the basic recording levels, then held a public concert as a warmup.

The peak levels in the public performance were +20dB higher and the softest passages somewhat quieter than those of the initial recording session. The peak levels increased enough that the TV film crew blew a monitor speaker. Ultimately, the greater dynamic range was used in the final recordings.

[Author's note: I wonder if the constriction in the dynamic range reflects the tonmeister system or the restrictions in recording for LPs. As far as I know, most of the Russian output is LPs rather than tape or CD.]

[Editor's note: The term "tonmeister" is merely the German designation for a record producer, and for the schools that teach the craft of recording. Mr. Johnson seems to equate "tonmeister" with "multiple mikes," which is not necessarily correct or fair.—sho]

The Keith Johnson tape recorder

Johnson's recording system is all class A, discrete circuitry with largely balanced differential input and output. There is a large inductor in the line amp to compensate for changes in the microphone diaphragm, as the 1.5-micron materials are hygroscopic.

His recorder is still essentially the same (3-track) as it was in 1965. It has a 3.5MHz bias oscillator and uses a time-constant of zero in the playback equalization. The deck has a 75dB signal-to-noise ratio but the perceived noise is quite natural. The tape noise "sounds like room noise" rather than the usual hiss and is buried in the noise of the recording environment.

The same machine is used to cut records with an additional head block placed on the left hand side for the preview signal (used to control the pitch of the lathe). Because the recording and mastering are done with the same deck, alignment is better than if different tape recorders were used.

In a picture of his recording setup there was a pair of crossed cardioids with an omni mounted high up. There were 2 other mikes placed in the back of the hall to create a surround sound. Normally the cardioids are pointed inward. Depending on the recording situation, Johnson will change the mike spacing; these were spaced about the same as human ears to get some time-difference information.

Differential-stress-mode testing

His lab mikes are tiny figure-8 units mounted on a servo-motor-controlled beam. The cone profile is drawn on paper and tracked with photocells so that the mikes are kept a constant distance from the average cone position (remember the driver is playing during testing). The test uses only small-scale excursion, at the high-milliwatt level at most.

The speaker is in free space, without any enclosure (although most tests do use some enclosure). The bounce-back through the cone is attenuated enough that it is not a problem. This is not a terrifically high-accuracy test so that a 20dB attenuation of the signal is enough experimentally.

The loudspeaker faces upward. There is acoustic foam on the ceiling to absorb bounce-back. To overcome the effect of gravity on cone position and restore the symmetry of movement within the magnetic gap, Johnson applies a dc offset.

This method is designed for small loudspeakers and standing-wave behavior. With larger speakers the crossovers are low enough that the speaker can be put in a cabinet and individual parts probed directly.

In response to a question about the use of higher frequencies, he replied that the smallest speaker he can test this way is a one-inch dome. With a one-inch speaker he can plot the center of the dome, the voice coil attachment point, and the first breakup mode. These all have different frequency response and very different phase relationships among them.

A member asked about testing only at low amplitudes. He replied that this did not pose a problem unless polypropylene drivers were used, as these have significant hysteresis and will play back differently at high and low levels. This would be a plastic mode rather than an elastic mode.

Johnson feels that this method represents a simple way to determine the problems in loudspeakers so that now he can make small changes that dramatically improve driver behavior and hence the frequency response. And these improvements translate into something that is practical to build.

In a closeup of the scanner, we saw that the microphone grouping has a pivot point that follows the contour of the cone closely (generally < 1/8 inch). With greater distance the output starts dropping very fast. (Sensitivity of such a dipole falls off with the cube of the distance.) This method requires carefully balanced mikes which have to have equal amplitude and phase response. Johnson started with 25 Panasonic capsules and ended up with 1 pair that worked well; then he added electronic equalization to get the perfect balance required.

Scanning the driver produces a series of 3D plots depending on input frequency and location (radius) vs. amplitude (dB). These are differential (bending) stresses. Usually at about 2/3 of the cone radius there is a beat or cancellation pattern between energy radiating from the voice coil and reflected from the surround. This is "cone cry" and is very audible in a speaker. With an 8" driver this occurs in the middle of the voice range.

The flexure points uncovered by this technique represent deviations from pistonic motion of the driver, and are responsible for differences in radiation 'pattern between drivers with similar dimensions and total energy output.

Soft-dome tweeters can have flat frequency response but still have lots of energy jumping back and forth within the dome because of out-of-phase motion of the center of the dome relative to the surround. To make improvements in the drivers used by Precise, Johnson used simulated production runs and redid the measurements until he had a set of drivers that had reproducible characteristics in production. These modifications included little holes, little dots, and little cut-away areas. Such modifications are certainly no less producible than other custom modifications of tweeters using glues and felt dots.

Foster asked how consistent and reproducible the scanning method was. Johnson replied that the method was not perfectly reproducible in terms of absolute level but that he got the same shapes and modes on repeated scans although the levels were different. These data also varied with time and temperature but the position of the nodes and their associated distortions had been very reproducible.

Johnson demonstrated some of the characteristic sonic effects of driver resonances and stress modes using a set of Precise speakers and a portable Technics DAT deck.

The playback obviously used the whole driver although the recordings were made by miking parts of either the radiation pattern (with a movable, conventionally near-field microphone) or the diaphragm itself (with the differential technique).

1) Pink noise. There was a cyclic pumping which was an artifact of the numeric reset in the digital IC used as a random-number generator.

2) An omni-mike pickup of the random pink noise reproduced through a speaker while moving the microphone off axis. There was a change in the frequency balance and degree of coloration of the sound as the mike moved off axis.

3) The output of a single probe microphone as it moved across the surface of the driver. The levels varied across the speaker and the character of the sound changed with the microphone location (similar to #2).

4) Transient probe doublet (differential mode). This was full of resonances and changes in amplitude, characteristic of the nodes. Although we were hearing the reproduction over the full speaker, the dominant sound was of the nodes themselves, because of the amount of gain in the recording.

5) A musical selection with voice, simply played over the full speaker. This was the "control" for #6.

6) The same music played while the driver was scanned. The driver used was a Polydax 6.5" midrange driver. It used a butyl laminated cone, a butyl rubber surround, and a decent magnet. It is generally considered to be a good driver.

The music included cymbals and voice, which Johnson considers most clearly show the problems with drivers. The character of the voice and the levels changed but we could still usually hear enough to recognize the sound although some portions were largely cone cry.

Listening to these types of signals is an exercise in ear training. By hearing the problems and distortions in the sound at high level, and unmasked by the total driver output, it should become easier to identify the same sonic "signatures" in normal playback.

In response to a question about how deep in the mix something can be before it is not audible, Johnson attempted to show such training by playing a recording with a "buried" sound effect. He played the complete mix twice, but played the effect alone in between the two, drawing our attention to how it was created, and to why it wasn't quite right.

7) Marching footsteps and vocals, guns firing. This was very complex and it was hard for me to tell where to focus my attention.

8) He directed us to listen to just the footsteps (isolated in this section). This is a sonic "object" that should be recognizable. In this tape, the "footsteps" were made by shaking a Scrabble box (i.e., not real footsteps) and the gun was a blast from a synthesizer.

9) #7 again. The sound effects were almost 50dB down but some people could still hear and identify them. Johnson commented that recognizing an additional signal 40-50dB down was like hearing a change of 1/20dB in frequency response. David Moran commented that this was not a fair test as musical material was not like noise and was not always at full level. Foster commented that this signal was not harmonically or time-related and therefore not well-masked.

David Moran asked if Johnson believed masking was amenable to will and Johnson replied that it was.

The conservative definition of masking is for everybody, under any conditions. People vary in their susceptibility to masking, from 10dB to 30-40dB depending on level and signal. A lot of this is training (as Johnson suggests).

Effects of filtering in digital systems

Johnson had a number of slides to show us, including SEM (scanning electron microscope) pictures of the hair cells in the cochlea. Included was a series of slides showing oscilloscope photos of frequency sweeps through a brickwall analog filter of the kind used in early digital systems. Because of the phase shift and group delay in such filters, as the sweep was done faster, the photos showed the effects of delaying some frequencies into later portions of the sweep. These were not frequency shifts, but time shifts showing up on the swept 'scope face. The amplitude envelope of the sweep also changed because of the addition of delayed components. This is a familiar phenomenon to anyone who has studied the tone-burst behavior of other stored-energy systems, e.g., loudspeakers.

These effects in the electronics are important when dealing with speakers because variations in HF waveshape can lead to TIM (transient intermodulation distortion). Even though the frequency response has not changed, the TIM components have changed drastically.

Al Foster commented that this looks interesting but that these signals do not resemble music, much of which has little energy above 4-8kHz. At 20kHz the energy level is usually very far down.

Johnson replied that these effects were acting at about 8kHz but appear in the sweeps to be part of the high frequencies. The usual mikes respond to about 16kHz but they are beginning to roll off by 20kHz. After all the ICs in the recording chain, the limits are probably about 15kHz unless a distortion producer like the Aphex is used. He said that this was not related to frequency response. These things cause cymbals to begin to sound hard and lose their characteristic metallic sound. The saxophone-reed sound is also affected. If everything is working right then you can hear the reed mechanisms.

The recording fed directly to speakers from the mikes is quite wonderful. But when fed through the recording chain, whether the analog or the digital, there is a loss in the transparency and effortlessness.

Foster was not convinced.

Johnson suggested that the ear can be trained by listening to these effects at half speed. After training, he claimed, they were apparent on $200 speakers or using cassette copies.

(The following demonstrations were from digital tapes made on a Technics portable DAT machine.)

To demonstrate the effects of digital processing, Johnson played an analog tape through a digital filter, and then listened to the resulting copy at 1/2 speed. This was with a brickwall low-pass filter like that used in the Sony Fl. For comparison, the original analog signal was also copied once without the filter. Female voice at half-speed sounds like singing (?) cows, and cymbals lose their metallic nature. Once you hear the effects, Johnson claims that they are audible in real time. The analog recordings are claimed to go out to 30kHz.

The brickwall filter he described as adding "hardness" and envelope distortion which is equivalent to TIM distortion because of the high Q filters. He claimed that we were hearing a lot of different things happening and that the digital sounds brighter because of TIM and other distortions.

There was clearly a difference in frequency response. He claimed that the analog recording would go out to nearly 30kHz while there was only a 1/4dB rise at the top for the analog version. David Moran pointed out that across the 1-8kHz band a 1/2dB rise would be clearly audible. Note that the change of tape speeds makes the 20kHz digital-filter cutoff occur at 10kHz, which should be audibly different from the 15kHz cutoff claimed for the analog recorder.

Johnson claims that the harshness and granularity cannot be removed by equalization.

Compared with the analog copy, the digital was different in hiss, solidity, and the "crispness" of the cymbals. [We should note that all this is dependent on the filter design and construction. He never said this filter was from any commercial product, just that the design was similar]

He said that there was a dullness and a brittleness to the sound after the filter. He felt that the TIM problems caused a crispness that was desired in studio recordings.

All of these examples were only band-limited. There was no sampling or conversion to digital. Johnson said that if he did complete the conversion to digital, there was only a slight added dulling, which he felt was related to the quantization floor. However, if the system is ultrasonically dithered then there is a little aliasing or folding down into the audio range. But he stressed that these are really nitpicking, very small effects.

His conclusion was that most of the digital effects he dislikes are related to the input anti-aliasing filters. But there are now quite a few filters available without TIM or time shift. For example, Apogee (on the West Coast) makes one. Some of the Japanese products are also good.

For example, the Technics DAT recorder uses a form of delta modulation, with a very fast flash converter and a ranging converter to keep the fast converter in range. This whole circuit is kept in a feedback loop so that it tracks the signal. Then the filtering can be done in the digital domain, which gets rid of the high-Q filter problem. He felt that about 3/4 of the problems he found with this machine were in the front end and IC chips that could not transfer the signal.

He had tried the dbx 700 (which also uses a form of delta modulation) but he felt that it suffered slew-rate limiting.

He felt that these were typical of industry-wide problems dealing with complex information instead of test signals. [Publisher's note: the dbx 700 was designed to do (and does) better on music than on test tones!—drm] In the case of the Technics DAT, there is an asymmetrical slew rate leading to offset or subharmonic components. That is, there is distortion generated by the component itself in the presence of complex signals.

Johnson demonstrated this with music using a tape of a pipe organ at St. Mary's cathedral (for the complex example) and a single female voice, which is easy to record digitally. Both were recorded on DAT. The bass was hard and solid but the female voice quite wide, while the corresponding analog master had a central image.

He then played a soft passage of the DAT tape of the organ and raised the gain to compensate. The sound tended to fall apart (became muddier and less pleasant to hear) when it became more complex. There was a lot of breakup and sonic confusion. The recording levels were kept well down and not close to the digital clipping level.

Moran pointed out that some DAT decks and CD players can have 1-5% distortion in the -30 to -60dB range.

In the *Reference Recordings* "Blazing Redheads" CD there is a flute that causes spurious noise on some CD players. The noise can be at different levels depending on the particular sample of the player. The noise is not related to the flute. In this recording all the main tracks bypassed the console except the flute (and some other instruments). Micha Shattner said that this was related to scrape-flutter IM that interacts with analog tape machines to drive them crazy. This makes some high-frequency instruments like the flute, the piccolo and the Irish tin whistle almost impossible to record.

The analog tape machine used by *Reference Recordings* has no guides in the tape path near the heads and only 1/4" of unsupported tape between the capstan, the heads and the first idler. This reduces scrape and modulation a lot but there is still some there. But Johnson said that digital recording was far worse.

In all of his recordings, the analog chain is just analog. *Reference Recordings* uses the analog chain to make "pure analog" LPs, and there is a parallel digital chain for the CD releases.

After the mikes (used for both recording systems), the digital system has a line-level passive mixer and an amplifier (made from discrete circuitry) that drives the digital filter. Thus there are a total of 3 amplifiers (counting the mike preamps) before the A/D converter. Finally, after the D/A converter there is an additional amp to drive the recording head. These are many fewer stages than for a typical recording.

"Blazing Redheads" marks the first time *Reference* used their own converter and filter. They originally started with the Sony Fl but were not happy with it. They then tried the Sony 701. They used the 701 as a starting point because all the electronics were on a flat card and so easily accessible. So far he has replaced A/D with an instrumentation unit, put in a fairly good S/H, and a discrete brickwall filter. This is the current release.

Johnson is not a minimal-miking purist in the high-end-audiophile sense. And his comments about digital recording left me with the feeling that, if the problems of sampling and conversion can be reduced, the convenience and potential quality of something like DAT would be very welcome in the recording studio.

Precise Acoustics loudspeakers

Johnson has designed the Precise series of speakers for good home systems which have better resolution than most studio systems. They are not targeted at the ultra-high-end audiophile. His designs for Precise have involved some dealer feedback in terms of what represents sellable sound, as well as his own experience in hearing what is on his master tapes.

The cone materials are from Sweden and Iceland. Cone materials, he feels, are a black art—when to harvest the trees and what sheep to shear. Basically, they are doing what everybody else does: looking at materials, modifying, and making small changes to come closer to their ideal sound.

The Precise speakers all have solid bass for their size. Although the speakers are vented they are not tuned according to the Thiele equations. He did this because he felt that two speakers in a room will interact and thus alter the alignment. [Editor's note: if the speakers are more than a half-wavelength apart, their outputs will no longer sum coherently, so this should not be a problem above the extreme bass.—mpfl

The crossovers are conjugate-damped variable-rate crossovers similar to elliptic filters. They are designed to nudge the drivers into a faster rolloff. They do not have a simple, constant attenuation slope (dB/octave); the response drops off at an increasing rate. The network matches the acoustic behavior for a minimum-phase transition.

The capacitors in the simplest speakers are high-voltage nonpolarized electrolytics, bypassed with mylar capacitors. As you go up the line the parts become better, going to polycarbonate caps in the most expensive units.

The Monitor 3 speakers that we listened to during the demonstrations cost $300/pair. The rest of the system was an Onkyo Integra A8190 amplifier, a Sony TC D5M cassette deck, and the Technics DAT portable.

Intended distribution of the Precise line is through independent audio dealers rather than major chains.

Johnson is quite intent on making a good small speaker, although I think he has focused largely on the driver design. He did not discuss the crossover to any great extent, so I am not sure how much of his efforts have gone into that, and he apparently did not spend much of his own time on the cabinet.

I thought that the speakers we heard sounded fairly good, although not wonderful. The room that we meet in is quite unlike most home listening rooms, with brick walls and fully carpeted floor, as well as being larger than most living rooms, so I would want to listen more before making a judgment. But I think that the speakers do deserve a listen.

— *Carl Deneke (Massachusetts)*

[Publisher's note: Despite DSMA and all that, the models 3 and 5 have been measured and reviewed in a recent *High Fidelity* and *CD Review* (formerly *Digital Audio)* respectively, the latter article by me. Both models were reported to sound only fair—"forward" is how Bob Long described the 3, "honky" is how I found the 5—and both measured with significant bumps and peaks, with each unsmooth tweeter run in at a strangely low level. Weird, given this meeting.—drm]

October 1988 BAS meeting

When we entered the meeting room, there were a TEAC 3440 4-track open reel tape recorder, a Crown D60 amp, a Harman-Kardon Citation 12 amp, a Lafayette LR-111 amp with built in SQ decoder(!), an Akai CS-F14 cassette deck, a Dynaco PAS-3 preamp, and a Phase Linear 1000 preamp (which has noise reduction) whose input IC had been rewired as a polarity switch—all sitting on tables at the front.

Among the more modern equipment was a dbx CX1 audio/video preamp and a dbx DB5 CD player.

The first comment anyone made was that the equipment made us think of a BAS meeting of 10-15 years ago. But then this equipment would have been much more up-to-date. Someone else commented that they felt the same way looking at the rest of us.

Open Forum

Poh Ser Hsu is still working on subwoofers. The latest is a 12-inch diameter cylindrical column that is 7 feet high and has two 12-inch woofers. The woofers are clamped together front to front and are mounted at the top of the enclosure. In this arrangement, they can generate 100 to 115dB at 16Hz. The tube is made of cardboard so the system is surprisingly light (about 40 pounds) and can be carried under one arm.

There was some debate about the usefulness of such low frequencies. One member commented that it would be "like living near the T" and another noted that in Poh Ser's last demonstration we could talk normally; the only way to tell the speaker was working was to feel the walls moving because people cannot hear frequencies as low as 16Hz. Of course, the absence of higher harmonics is a sign of a good speaker and the fact that these can go so low does not preclude their use in the vital 32 to 64Hz octave, where there is a great deal of both musical material and hall sound. Some organ freaks would of course like substantial output at even lower frequencies. Other members felt that bass was seductive and that exceptionally deep bass was the next frontier in hi-fi.

Mark Fishman commented that Analog Devices in Norwood has announced a laser-trimmed true 16-bit digital-to-analog converter (DAC) to sell for about $10 in quantities of 100+. This will be a major improvement on the individually trimmed units now used. He hoped the AD parts will not have the distortion in the -30 to -60dB range produced by many current DACs. These should appear in audio products in about a year.

In dbx news, Bob Adams has designed a true 18-bit analog-to-digital converter that is being sent out as samples to the rest of the audio world. Apparently it is working better than expected and is good to 20 bits. It has been completely converted to VLSI form and they are now working on a 18-20 bit decoding version. The best current competition is from NEC who have an integrated 16-bit converter.

John F. Allen announced that his HPS 4000 sound system will be installed in the Showscan movie projector house in France. Showscan uses 70mm film projected vertically at 60 frames/sec. In contrast, IMAX (which also uses 70mm film) projects at 15 frames/sec, or 24 frames/sec when the film is sideways. In the IMAX system there is some flicker or strobing which does not exist in the Showscan system. He described the film as having pictures that are "almost better than real life." There is lots of light and good color saturation. There are currently three feature films scheduled for release in Showscan, and there may be other releases as well.

In a direct A-B test with the standard sound system, Allen's HPS 4000 sound system gave a bigger and cleaner sound and the sound also had a different character. All six audio tracks on 70mm prints use Dolby SR noise reduction now but they ultimately will all be digital.

David Moran, who has been described as a writer not always, uh, sympathetic to high-end audio, recently was invited to the house of a fellow BAS member who had a new preamp from PS Audio. The preamp was supposed to be very musical, but it reminded Moran of listening to a broken Dynaco PAS-3. The unit was unbelievably noisy, and the noise and hum changed in level whenever the volume control was touched (independent of the volume control setting). The switches all made loud pops and there was severe crosstalk among tuner, CD, and phono inputs. He described it as "the audio equivalent of picture-in-picture TV; you could listen to two programs at the same time."

Moran said that he had not heard anything like it in 20 years. Nothing seemed grounded. The knobs had play on their shafts. "And this is a unit whose power-supply transformer is on a cord so it can be 10 feet away." (Perhaps there is a reason for that isolation.) The owner "wanted a second opinion as to whether anything was wrong." Micha Shattner commented that PS Audio had always assumed that the dealer was the last step in the assembly line. Discussions in underground magazines on lack of reliability may reflect poor workmanship; if this unit was typical of the whole high end, it is surprising (and depressing) that anyone would put up with those products. Micha commented that his 26-year-old Marantz 7 preamp now does have a little potentiometer noise in one channel. A small difference in assembly quality?

In a related discussion, one member commented that insurance companies might not cover damage from an electrical fire that is caused by equipment not Underwriters' Labs-approved.

Meeting Feature: John S. Allen

The speaker was BAS member John S. Allen, who is a student of 78-rpm playback and who also translated Jens Blauert's famous *Spatial Hearing* into English. John had gathered all the special (old) equipment at the meeting to discuss and demonstrate the creation of "accidental stereo" by the synchronization of two (or more) recordings of historical events. Many events (and some musical performances) were recorded with multiple microphones feeding separate recorders. If these can be kept in synch, the result is a stereo recording with the microphones in somewhat less-than-ideal locations.

In describing the equipment listed earlier, he said that the Dyna PAS-3 preamp did not make terrible noises although it did get microphonic sometimes, and he got it for $5 at a rummage sale. The turntable was a Bogen B61, which has variable speeds. He removed the spindle completely because he "has never found a centered record, except for a direct transcription." In order to synchronize two recordings, wow and flutter must be kept to a minimum or the hoped-for central image will swim back and forth.

The Phase Linear 1000 preamp was included for its noise-reduction functions and to act as a polarity switch, as its input IC had been rewired. A switchable-polarity buffer in the input is needed because Allen often works with two recordings of the same event which turn out to have been recorded in opposite polarities.

As an introduction, he played a cassette prepared by Brad Kay of a 1932 Duke Ellington session. The current interest in re-creation of stereo from matched pairs of original recordings began when Brad Kay and another jazz enthusiast were comparing recordings looking for different performances. They found two 33-1/3 records (this was the earlier coarse-groove format with 7.5 minutes per side, not the later microgroove LP) which appeared to be of the same performance but recorded with different aural perspective. After months of work they were able to reassemble the two records into the stereo tape we heard. (This case was relatively easy to synchronize because of the direct-to-disc recording: there were no intermediate generations to add wow and flutter, as compared with later recordings that were taped. Disc molding does not add wow and flutter like tape.)

This was clearly stereo. John Allen commented that having stereo instead of mono recordings "provides a wider window into some of the greatest American music ever written."

Brad Kay has found a number of other examples from that era, mainly classical, where a single event was recorded with two completely independent sets of microphones and recorders. For an individual, however, it is hard to "re-stereoize" music without access to the archives of the record companies. In contrast, documentaries are made of almost every major political speech, and many of these recordings have been issued commercially from multiple sources.

Allen became interested in this and described the experience of listening to a Boston University-University of Lowell hockey game in stereo with each school broadcasting a single channel. Because the announcers were on opposite sides of the arena, the players, fans, and puck sounds were in a sort of stereo. However, the announcers were in dual (unconnected) mono, and as one announcer became excited the other would become glum.

Much of the meeting was a demonstration of "stereoization" using President John F. Kennedy's inaugural address. In pictures of the event it is obvious that there were 4 mikes on the lectern. In contrast, pictures of Franklin Roosevelt's speeches show as many as 15 mikes, each presumably from a different radio station or network. In the case of the JFK speech, Allen did not have any original tapes, instead working from commercial documentary LPs, which are not all from the same source. In this case he worked mainly from the NBC feed and the U.S. Army Signal Corps recording.

Working from this type of material, sound quality is a problem. The best sound of the JFK speech was on a Camden recording that he found in the Lincoln, Massachusetts, public library and on a tape from the Kennedy library. [If any audiophiles taped this speech live (for example, off the live radio feed), in 1961, Allen would like to get a copy of the tape. A digital copy would be best, as it would avoid adding flutter. Please write to the *Speaker* with information.]

Allen played a portion of a previously made stereo tape of this speech from 1961, thus demonstrating the goal. This also showed that there was a 1/4-second gap in the NBC feed. He made this tape by copying the mono record to one track of a stereo tape deck, then playing that tape track into a single headphone channel and playing the other record into a second track on the tape recorder and the other headphone channel. He would then drag his finger against the turntable to center the image which would synchronize the two channels correctly. (Clearly this process works only on a deck that uses the same head for playback and recording. Otherwise there will be a delay equal to the time it takes for the tape to travel the distance between the two heads.)

In this demonstration, the rear speakers were fed the main signal delayed 33ms, but were otherwise unprocessed. The left rear channel received the 33ms delayed feed from the left front only, i.e., there was no crossfeed.

While the image was somewhat unstable, occasionally it would "lock in" and become quite solid and spacious.

Techniques

In order to create a stereo image, two channels have to be within 10 microseconds of each other to be in synch. The limits shrink to 1 microsecond if you plan to apply Dolby surround-sound processing (which uses L-R information in the rear speakers). In this case, if the two channels are a few microseconds apart, sibilance and a shimmering appear in the back channels. Micha Shattner commented that tape skew can cause this amount of shift. The Lexicon CP-1 can correct automatically for this effect, because it is designed to compensate for azimuth error in tape playback.

A 10-microsecond inter-channel delay (good enough for normal stereo) is only 72 degrees of phase shift at 20kHz and would cause only a 1.5dB drop at 15kHz after summing to mono. Ten microseconds is generally accepted as the inter-channel match necessary for the most trying conditions—such as hearing clicks over headphones. For most purposes, greater inter-channel differences are acceptable. The channel differences at the beginning or end of reels of tape in a typical cassette player can be in the 30- to 50-microsecond range. This would usually be heard only if the channels were combined into mono.

In general, a 50-microsecond match will give stereo as good as a cassette tape.

Brad Kay has gotten superior results using direct-to-disc sources (e.g., some 78s) and a millstone-like turntable. Wow and flutter cause problems because the 2 channels randomly change phase with respect to each other.

Eventually, Allen hopes these techniques will become routine with digital technology. This would involve digitizing the two channels, indicating some moments when the two channels were synchronized, and then a production computer would vary its sampling rate to make the two segments come together. This requires a constantly variable sample-rate changer. There could be either a manual mode where you pick the points and the equipment synchronizes the tape between the points or a variety of automatic modes.

Ultimately, he could imagine a purely automatic correlation, using a number of indicators. These could include 60Hz hum included in the original recording or horizontal TV synch leakage. If anyone is interested in working on either the hardware or the software, please contact the *Speaker:* we will pass your interest along to John.

Different versions of historical recordings can sound very different. For example, the Army Signal Corps tape (from the Kennedy Library) and the NBC radio feed on the Camden LP have the same "bump," which sounds like someone bumping into either the lectern or the microphone, but the equalizations of the tapes are very different. This would lead to a "treble on the left, bass on the right" type of stereo if not corrected. In this case, it was necessary to sacrifice part of the frequency range, because the full frequency range was not present in both recordings. On the final copy he used the Phase Linear noise reduction.

The NBC recording on Camden is from a 33-1/3 transcription disc. Most of the crackle is in this recording and not in the Army tape. There is 1 /4 second missing, probably while somebody was changing the discs. Then there is a splice every 2 seconds for 30 seconds. This probably was to correct for a gouged transcription.

Allen's technique is to try to maintain a centered image with headphones by using the pitch control of the tape recorder while taping the second recording on an unused channel. To make this easier he has moved the speed trimpot out of the enclosure and added a mechanically geared wheel. This gives him fine control over the tape speed.

It is very hard to monitor and correct anything. There is a very small allowable time window, and the method does not always work on unstable material. It may become easier to do this work with more recent material because most major speeches are now recorded in color on video tape and there is an inherent quartz time reference to maintain synch. However, using old analog tape masters is very hard but important, and Allen obviously feels that it ought to be done.

Allen also feels that the recreation of major speeches in stereo is very important (and he spends a great deal of time and effort doing it) because "it allows a unique window into an earlier era." He feels that it is possible to "come face-to-face with Calvin Coolidge, Huey Long, and FDR."

Recording speeches became common with the wide availability of radio in the late '20s and early '30s. Electronic recording equipment became widely available in 1925-'26. Essentially all major public speeches were recorded in the 1930s and some in the 1920s. Allen estimates that there is probably an 80%-90% chance that any major public speech today will be recorded in at least two separate versions.

Bell Labs conducted a series of experiments in the early 1930s in which people preferred stereo telephones to wide-range playback. Using telephones, people preferred a 5kHz lowpass filter to an 8kHz flat frequency response. However, on the low end there was subjective improvement as the bass cutoff was lowered to 30Hz. High frequencies bring along a variety of other problems including phase dispersion and, crucially, an increase in the noise floor.

Ira Leonard commented that stereo telephone, i.e., talking to one person over two lines, seems more real and you can hear more detail. Ira commented that more people now are being fitted with 2 hearing aids because it is more helpful, giving more auditory cues.

In preparing to give us a demonstration, Allen mentioned that he had done all his work before at half speed. This demo was done with the recordings played at full speed. He used his finger on the turntable as a coarse adjustment and another finger on the tape deck pinch roller for a fine adjustment. (This is the "two digital converter" method.)

Basically, he cues on the groove echo of the previous groove to tell when to start. He copies the first recording on channel 2 of a 4-track tape recorder. Then he tries to synchronize the turntable to the tape and records the second recording on another channel, moving to different channels as needed.

The external, geared speed control gives a 700:1 advantage over the tape deck's original speed control. Tonight was the first time he had used this control, which consisted of a 6" wheel mechanically tied to a ten-turn pot. Fifteen turns of the wheel change the pitch of the recorder by one percent. This gives 0.02% resolution for a 3-degree movement of the wheel; the flutter of the tape is greater than that. He used both the wheel and finger-dragging for the demonstrations.

When asked how he found the beginning, he replied that at first he just did it over and over until he got it right. Now however, he copies each track to a new tape and splices a timed leader on in front. He can then synchronize with the leader tape.

Ralph Dopmeyer (Titanic Records) asked how much music could be reconstructed into stereo. Allen replied that probably very little was available compared with documentary. There was a balance between the expense of redoing a take (in case of equipment failure) and the expense of an extra set of equipment, so probably only 10 or 20% of the largest works were multiply recorded. With small groups multiple recording was less likely because it was easier to ask the musicians to play again.

The Toscanini/NBC Symphony recordings are one major exception. Apparently different transcriptions were used for North and South American delayed broadcasts and each had a separate mike for the orchestra. These had either English- or Spanish-speaking announcers. This happened for all the Studio 8H concerts from the late 1930s to the late 1940s. This is an amazing treasure and should be put into stereo.

One channel is recorded in its entirety on track 1. He then puts the rest on tracks 2, 3, and 4. He starts with track 2 and tries to hold that track in synchrony as long as possible. Then he will go to track 3 and so forth. Finally, when the two channels match (i.e., track 1 and segments of 2, 3, and 4), he will draw up a cue sheet and make the final stereo copy by switching among the individual track outputs. This way he does not need to do any tape splicing.

Allen synchronizes by listening for the center image. This is easy with speech because there is only one major sound source, but music would be harder. When asked if it would be easier to synch with a difference (L-R) image, he replied that this did not work because in general the mikes were not matched and their frequency responses were not the same.

Using a L–R null would also make the system level-sensitive. Using headphones decreases the importance of level differences and increases his sensitivity to arrival times. He corrects level differences in the final pass.

It may be necessary to do some preliminary equalization just to get started in finding a center image. For example, the NBC recording of the JFK address had a 4kHz peak that had to be knocked down before he started work.

When asked if any of the record companies were looking through their vaults, he replied that RCA had looked but had no commercial interest in stereo releases of the studio 8H recordings. Most of the re-stereoizations are being done by amateurs at this time.

Some of the recordings used:

1) *John Fitzgerald Kennedy Memorial Album,* Premier #2099. This used the Army Signal Corps feed. This record is from the National Archives.

2) *U. S. Presidents series: Actual speeches of Franklin D. Roosevelt and John F. Kennedy,* Somerset P16100. This came from the NBC radio feed and perhaps was the television feed as well.

3) Diplomat 10000-A. This was probably a bootleg of the television feed and has video buzz.

4) *John F. Kennedy: A self portrait. The gallant warrior of the thousand days.* Caedmon TC2021. This uses the NBC feed and was taken from transcriptions.

5) Tape from the Kennedy library. This uses the same Signal Corps source as number 1 but has better audio quality.

6) Cassette from the JFK library, originally from the National Archives.

Allen also has a better copy of the NBC source on order from the National Archives and a CBS feed on order from the Library of Congress.

— *Carl Deneke (Massachusetts)*