THE BAS SPEAKER

THE PUBLICATION OF THE BOSTON AUDIO SOCIETY

This is the fourth, and final, issue of volume 25.

Please pass along the membership form, on the inside back cover, to interested friends!

ABOUT THE BOSTON AUDIO SOCIETY

PO 211 Boston, Mass. 02126 USA
(603) 899-5121; fax (603) 899-6415
Email: dbsys@attglobal.net
Web site: www.bostonaudiosociety.org/

The Boston Audio Society (BAS) is an independent member-supported organization promoting the highest quality of music reproduction in the home and high standards in recording and transmission.

More than a local society, the BAS speaks to the worldwide community of audio enthusiasts. Founded in 1972 and now in its 32nd year, the BAS meets monthly to hear and discuss developments in audio. Guest speakers over the decades have included prominent engineers, designers, researchers, editors and reviewers, musicians and critics, and broadcasters and recording producers. On occasion we hold joint meetings with the Boston chapters of the AES (Audio Engineering Society), SMPTE (the Society of Motion Picture and Television Engineers), and the ASA (Acoustical Society of America). Our non-commercial newsletter, the *BAS Speaker* (*BASS*), includes comprehensive and lively coverage of these meetings, as well as reviews, news columns, features, letters and other articles on a variety of audio and such related topics as home theater and video.

Membership ranges from the novice enthusiast to the technically sophisticated. Consumers and producers of audio equipment are both represented. Members include freelance journalists, reviewers, and editors at the major audio magazines, as well as design engineers, consultants, and researchers who influence product development and thereby the course of the industry. Some members work for manufacturers (as technician, engineer, or marketing manager), others for dealers. All are devotees — audiophiles in the best sense of the term — and tend to be technically and technologically aware, informed about the marketplace, and keenly interested in scientific approaches to audio.

For these reasons, the BAS and the *Speaker* are a vital forum. As someone involved in audio, you likely will find the group an interesting, helpful resource. Our meetings and newsletter may help shape the future of consumer and pro audio even while clarifying its past. If you are a manufacturer, for example, you can use the BAS to keep up with trends and developments, or to learn informed reactions to products and events. At the least, we attempt to be a clearinghouse for ideas, helping various parts of the industry keep in touch with one another.

To join, or to obtain more information, please use the form in the back.

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Editor: David J. Weinberg
Assistant editors: David R. Moran, Al Southwick

Contents

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From the editor

Beware and be wary: Consumer electronics equipment isn't simple anymore. Much of it is far from plug-and-play (in spite of the FCC's recent approval of a cable-HDTV p&p 'standard'). Multichannel receivers and DVD players require setting up, with some of the menus containing terms alien to the average user. As an example, although it is critical for a decent surround system to have its levels calibrated at least with respect to each other, consumer research reveals that fewer than 1% of home theater users have ever used the internal noise source and controls to make the adjustment.

I've run into problems understanding how to set up my own system, and I don't get enough help from manufacturers.

We all know that advertisers' and manufacturers' claims don't tell us all about any product. They generally don't tell us enough to enable even cursory informed comparison.

The *BASS* and a few other magazines try to provide you the data and information you need to begin making intelligent decisions and get something close to the optimum from your equipment. Good luck!

From the president

Give us your ideas, please: J.P. Leger has agreed to be program coordinator. If you have suggestions for new meeting topics (anything to do with audio and music is fair game), please contact him at jpleger@genome.wi.mit.edu; we are always looking for interesting presenters.

To the editor

ADC Distortion?

from John S. Allen (Massachusetts)

About Carl Deneke's digital analysis in v25n3: If we assume, as the article does, a theoretically perfect sample-and-hold amplifier which does not introduce intermodulation between the input signal and the sampling pulse train, there is no "error." All of the frequencies which result in the amplitude variation are ultrasonic aliases. Furthermore, these aliases are deeply suppressed by the playback lowpass filter. (An Excel spreadsheet in which I demonstrated how the pulsating, sampled signal relates to the steady audio signal is available on the BAS website, linked from the page http://www.bostonaudiosociety.org/articles.htm.)

It is indeed possible, even probable, that a sample-and-hold amplifier, an analog device, will generate some intermodulation between the audio signal and the sampling pulse train. However, modern professional-grade ADCs do not use a sample-and-hold amplifier at the final, 44.1kHz CD sampling rate. Neither do most modern DACs, professional or consumer-grade. Rather, they operate at much higher frequencies. Oversampling greatly reduces the amplitude of distortion components resulting from intermodulation between the audio signal and the sampling frequency, and the likelihood that they will fall within the audible range. Digital sample-rate conversion, made necessary by the oversampling, does not generate distortion components unless the math is faulty.

Unfiltered ultrasonic components remaining after D/A conversion might lead to audible distortion due to intermodulation in the analog playback chain, but the distortion would consist of new, lower-frequency components, not a pulsation of high-frequency components. And in any case, in an oversampling system, no pulsation at the low rates described in the *Speaker* article exists even as a first-order effect.

David Griesinger's PowerPoint presentation about the audibilty of ultrasonic components in any case is at http://world.std.com/~griesngr/intermod.ppt .

from E. Brad Meyer (Massachusetts)

About Deneke's analysis: I think the diagram purporting to show the erroneous points on the high-frequency sinewave is incorrect. Data points that are in reality spread out over many cycles, because the samples at that frequency occur barely more than twice per complete cycle, are collapsed onto each other. This gives a false picture of what happens and probably describes as errors things that really are not.

Also, if the writers claim that distortion products are generated, why don't they measure them at the analog output of the device, which is the only place that counts?

More on CD Levels

from E. Brad Meyer

I read with interest Mark Fishman's letter comments in v25n2 about the levels on the original and the reissue of Michael Jackson's *Bad* CD. He is at pains to point out that the signal never reaches digital full scale and so is not technically clipped. Sure it is. The producer compressed the living hell out of it and let it clip, but with the digital processor's maximum set to 98.8% (a whopping 0.1dB below full scale) so the pressing plant wouldn't reject it. The sonic result is the same. [Fishman and Meyer are not at odds; what Fishman wrote was: "…despite the flattened tops of waveforms in the newer mastering, any clipping isn't coming from running out of bits on the CD. …The zoomed fragments of each file also show that, indeed, peaks have been severely limited in the remastering. … Limiting was introduced deliberately at some earlier stage in mastering" — DJW.]

Recently I completed an editing and mastering project for a long-time folk client. His previous releases had been pretty much within the genre — vocals, acoustic guitar or piano, electric bass and harmonica, and maybe an occasional electric guitar part. Over the past two years he has begun to use multitracking to assemble his own mixes, which have more of a pop sound. For the most recent project, some of which sounded like Stevie Wonder, he decided that the new CD had to sound as loud as anything else on the market. If he could remove his CD from the player in my studio and pop in another one that sounded louder at the same volume control setting, then I needed to do more. All at once I was competing in a contest that has been going on around me for many years without affecting me, and I wasn't ready to play.

I protested that too much compression made the music less vital. I pointed out that every listener sets the volume control for every CD, and if his was 3dB softer they'd turn it up. I reminded him that radio stations have processors that give everything the same squashed sound. None of it helped. If a DJ was doing a mix and didn't have time to adjust the level, his CD mustn't suck the energy from the dance floor by being even a bit softer than anything else.

The whole mastering process took so long that I wound up working for an hourly rate of one-and-one-half pittances. And still it wasn't loud enough. He took my final product to someone else who had a multiband compressor of a kind I was too dimwitted to rent, and in one pass the other guy made it sound 2dB louder and a bit more squashed, and got equal credit in the booklet.

The principal lesson from this is, of course, that it pays to be attentive, figure out what the client wants and give it to him. But this was an avenue I never thought I'd be dragged down kicking and screaming. In the last few months almost every CD I buy makes me turn my preamp volume down, sometimes from 10:00 (the common setting four years ago) to 9:00 or even 8:30. The first notch above zero on the control is now too loud to keep the music in the background. Everything is much brighter than in the good old days, way back in '99. Worst of all, people tell me when I talk about this that I'm beginning to sound like Gabby Hayes or Walter Brennan.

The Musician's Perspective

from Carlos E. Bauza (Puerto Rico)

BASS v25n2 was excellent throughout, and the feature interview of professional trombonist Paul Feinberg was special for me.

The musician's perspective is too seldom heard in audio circles. One could regard musicians as a given resource, which of course is a grave injustice. There is a universe of human toiling, elation, sweating, and relating with life for the music performances we coolly play through our systems whenever the whim strikes us, and there is electricity to juice it up.

Recording engineers are privy to some of this, as are producers and some others. But the typical audiophile is worlds apart from the human drama behind recordings.

When Feinberg told of the difficulties he had in certain moments of his professional life, I could nod with understanding.

As an amateur singer all my life, I have participated in most manifestations of singing, including solo, trio, quartet, quintet, choral, opera (comprimario or leads), and choral with orchestra. I know what Feinberg is talking about.

Besides all the clowning around we did as choral members in opera productions, and watching the stage directors despair over this, there was a lot of human drama going on.

Imagine a young person being struck one day with the image of his vocation — to be a conductor, perhaps. It takes huge resources and time to complete the preparation. Then come the first steps in the profession. And maybe some clowns are manifesting themselves behind the curtain, and the costumes did not arrive on time from New York's Stivanello, or the lead singer is indisposed, with no replacement under contract. More distressing still, orchestras are closing operations — even previously powerful ones.

I remember a concert production (neither scenery nor costumes) of Wagner's *Rienzi* in which I was a member of the chorus. We were challenged by this horribly difficult choral score. For the first time ever, our chorus failed to sing some parts correctly in performance. The choral director was livid! I have not returned to that chorus. The price was too high for an amateur to sacrifice substantial parts of regular life in order to receive a director's heat.

We had a visiting German director who knew *Rienzi* very well. The Puerto Rico Symphony Orchestra and our chorus were learning it from scratch, without the benefit of a local German tradition to build on, with the added aggravation of a very disorganized choral score that omitted the soloists' parts.

During one rehearsal, one of the horn players (that notoriously powerful, recalcitrant instrument) did not jot down a correction dictated the day before by the conductor. The director kindly notified the player of his mistake, and the row that ensued caused the director to resign on the spot, while the orchestra remained adamant about the director's affront to one of their members — two days prior to the performance.

The Casals Festival was almost canceled, but the director was decent enough to accede to the panicked requests of the festival's staff. He was a perfect gentleman, rising above the stupidity of the orchestra. During a coffee break I strolled passed the concertmaster and congratulated him for reaching a solution. Bad mistake: the concertmaster was still adamant about the director's affront. It seemed to me that he, as 'leader' of the orchestra, was afraid to lose status with his cronies.

Two years later I happened to be in the same place as one of the orchestra's officers. I asked him about the incident, and he confirmed that the horn player had not jotted down the director's instructions and the librarian had to write it down for the horn player (as in "You're the help, boy; get to it"!).

We have an expression: "May I see you among musicians" ("Entre músicos te vea"). It is a curse more powerful than Monterone's damnation of Rigoletto.

It is easy to extrapolate from all this. Recordings have human drama in their creation, sometimes happy, sometimes not. Bear this in mind while coolly setting the stylus in the lead groove, and may I see you among audiophiles.

Open Forum

by David Hadaway (NH)

Noise and Moving-Magnet Cartridges

in the October *Electronics World* tells of a circuit technique that results in better phono preamp noise performance than has been achieved before. With moving-magnet cartridges, the limitation at high frequencies is noise generated by the 47-kohm terminating resistor. This standard input resistance cannot be changed. However, the author replaces it with a 1-megohm resistor and actively drives the end that would normally be grounded, in order to simulate an input resistance of 47 kohm. The inductance of the cartridge shunts the noise of the 1-meg resistor so it doesn't appear. An improvement of about 3dB is achieved. The article goes into great technical detail on the noise sources and their minimization and is highly recommended to phono preamp mavens.

Beyond Mahler

As a followup to last month's "ASLSP" item, astronomers say they have heard the sound of a black hole singing, according to the 16 Sept *New York Times*. What it is singing, and perhaps has been singing for more than two billion years, is B-flat — 57 octaves below middle C [not too far from dc, in other words — DRM]. It appears as pressure waves through a hot thin gas that fills the Perseus cluster of galaxies 250 million light-years distant. The waves are 30,000 light-years across and have a period of 10 million years. "It's the longest-lasting symphony we know of," said Bruce Margon, an astronomer at the Space Telescope Science Institute.

Raisins and the Law of Unintended Consequences

In "News of the Sections" in the July/August *AES Journal*, there is an unusually long writeup (2½ pages) of a Pacific Northwest Section meeting, reminiscent of the detail and color of the better *BASS* writeups. This one is about Greg Mackie and his audio companies (Tapco, Audio Control, Mackie Designs). One illustration: they liked the feel of Alps pots but couldn't afford the minimum order, so they used cheap CTS pots. To get the proper viscous feel they tried STP, silicone heatsink grease, Crisco, and even raisin juice on the shaft. Finally someone came up with a compound that worked, although using it meant that they had to disassemble and reassemble each pot. Sometimes the compound got on the carbon element, causing clicks. Eventually they hit on the idea of laying a bead of the compound around the shaft-bushing junction and heating the pot in an oven, where capillary action did the rest. Years later they discovered that the goo glued the polystyrene knobs to the shaft, making service a challenge.

Spice Play

A July *Electronics World* circuit note, "A Linear-Voltage Amplifier," should be of interest to those who eschew negative feedback. The author used a Spice simulation, not measurements, to analyze a grounded emitter bipolar-transistor amplifier that has inherently high distortion due to the nonlinearity of the base-emitter diode. From making the collector load a string of diodes (instead of a resistor), the distortion is canceled. He shows that for a 10mV input the distortion is reduced from 1% to 0.0056%. However, the signal level is low, and the gain is equal only to the number of diodes, so it has limited application. Perhaps as a low-noise head amp for MC cartridges.

Out of Synch

Audio advances in TV transmission are surveyed over the last 20 years in the 17 September *TV Technology*. One area that has gotten worse is audio-video synchronization, which "has become a gigantic mess." Every level of video processing introduces delay, so the sound comes out early (which is unnatural) unless it is delayed too. Tektronix has developed a system called the AVDC100 that allows lip-synch correction via a video watermark. (I was watching *This Old House* on PBS Boston and sledgehammer blows were sounding 0.3 seconds before the visual strike. Steve Owades suggested also that this can result from the user's system connections: if he is routing the sound directly to the hifi and the image is being line-doubled in the television (which causes delay), then the sound can arrive early. (Owades would like to hear from anyone with experiences and ideas in this area; contact him through the BAS.)

"Loud, Not Fast, Wins These Races"

The 5 September *NYTimes* led an article on a dB Drag Racing event in Ohio with that headline: The loudest sound wins but no one can hear it. The event has corporate sponsors and championship circuits, top competitors put as much as half a million dollars into their cars, and it can be found from Finland to Bangladesh. This season the average spl measurement was 142.1dB, which is well above (nearly four times as loud as) the 125dBspl emitted by a commercial jet taking off. The international record is 171.5dBspl. No one stays inside. A Plymouth Voyager being "raced" (seats ripped out to make room for 32 batteries and 16 amplifiers totaling 44,000 watts) was reinforced to keep the sound from leaking out: in place of windows it had inch-thick sheets of Plexiglas (soon to upgrade to 3") and the door panels had been replaced by plywood covered with duct tape (some contestants fill their doors with concrete). The contestants and crew sat on the roof and windshield to keep sound from escaping. There are three classes of competition: everyday street cars; super street, in which everything has been removed and replaced with equipment; and extreme class for the real pros. See www.termpro.com.

Like Lapels

Bob Katz (NYC) wrote: DSD-Wide = PCM Narrow is how [I] (www.digido.com) characterized this 8 August 2003 message from Vicki Melchior on www.ilovehdtv.com: Yes, DSD-wide is 8-bit PCM at 2.8224 MHz. If you check Peter Eastty's paper from the Amsterdam May 2001 AES (preprint 5377), which is called "DSD-Wide, A Practical Implementation for Professional Audio," you'll see the steps he recommends for DSD. Generally it involves converting the single bit stream to 8-bit PCM at $64f_s$, doing all of the processing (EQ, dynamics etc) and data transfers with 8-bit or wider PCM (at $64f_s$), then returning to a DSD bitstream at the output.

Build Your Own HD HT PC

Stephan Chan (Maryland) recommends, with the growth of PCs as a hub for home theaters, writer Loyd Case's work at www.extremetech.com/article2/0,3973,1265796,00.asp), which walks through the entire do-it-yourself project.

A Most Noteworthy Museum

David Temple (Massachusetts) visited the Frederick Historic Piano Collection in the former Stevens Library Building in Ashburnham, Mass., 55 miles northwest of Boston. Their "expectations were grandly exceeded!" Patricia Frederick, half of the husband-and-wife team who own and built the collection of Erards, Pleyels, Boesendorfers, Broadwood, etc., played each of the instruments and on occasion sang. She is very knowledgeable and enthusiastic about pianos, and a better singer than pianist. She demonstrated the advantages of playing Debussy on an 1893 Erard instead of a modern Steinway, and she sang a Schubert song accompanying herself on a Graf from 1828, a piano with which Schubert was familiar. "We dropped in and stayed at the piano museum for only an hour — if you call ahead, you can arrange for a complete tour, which would probably be two well-spent hours." [As it was the site of a recent BAS meeting, watch for the upcoming summary — eds.]

Commentary and News

by David Weinberg (Maryland)

Home Entertainment 2004

The show (http://homeentertainment-expo.com/index.shtml; sponsored by *Stereophile* and others) will be held at the New York Hilton, 20-23 May 2004.

MultiMedia Manufacturer

This new business-to-business periodical is from Ed Dell (Audio Amateur; 888.924.9465; www.audioXpress.com) and is intended to advise consumer electronic equipment (CEE) companies of the problems and solutions related to contracting with overseas manufacturers. It is subtitled *Manager's Guide to AV Design & Development*, and is (from the press release) "targeted exclusively for management-level personnel who have direct responsibility for all aspects of the design and manufacture of audio and audiovisual hardware." Features will include profiles and interviews of CE equipment company decisionmakers; plant visits and evaluations; surveys of new chips; surveys of specialized-service providers (such as metalwork and finishing, product design, circuit boards, and OEMs); plus QC standards, issues and procedures. Although intended to be published monthly, to get started the first couple issues will be doubles — January/February and March/April.

A Physicist Meets the Twilight Zone

This is David Griesinger's subtitle to his PowerPoint presentation "Perception of Mid-Frequency and High-Frequency Intermodulation Distortion in Loudspeakers, and its Relationship to High-Definition Audio" (http://world.std.com/~griesngr/intermod.ppt). He referenced Karou and Shogo's "Detection of Threshold for Tones Above 22kHz" (preprint 5401, 110th AES convention, Amsterdam May 2001), which showed incontestably the inaudibility of odd harmonics (which we are most sensitive to) above 20kHz as long as there are no distortion products generated below 20kHz. Griesinger then recounted his difficulties trying to find PC hardware that enabled him to make clean 24-bit/96ksps recordings, particularly choral performances. One conclusion from his experiments was that amplifiers can cause audible subharmonics of ultrasonic distortion products, but speakers can't. He tested a number of ultra-wideband professional music samples and recordings and found none that had music signal above 23kHz (and such ultrasonic harmonics as there were were 40dB or more below the fundamentals), and the only signal above that was noise from the noise shaping used in SACD processing. From additional testing, and an analysis of our auditory system, Griesinger concluded that "nonlinear distortion in human hearing appears to account for the audible distortion in full chorus." His ultimate conclusion was that "no evidence was uncovered in this study that would invalidate rapid, blind, A/B tests as the gold standard for audio research."

LCD vs Plasma Displays

Kirt Yanke (18 August 2003 *Sound & Communications*) described the contrasting technologies and the relative performance characteristics of these two popular pixelated displays. Yanke points out the different effective brightness level and effective contrast ratio in various ambient-light conditions (plasma is better only in a truly dark room), color palette and uniformity, and operating cost.

Surround Professional *(August 2003)*

- ***Pump Down the Volume*** is Kevin O'Connell's editorial on the excessive loudness of recent movie soundtracks. He quoted a reviewer who wrote that "seeing one particular film was 'an ear-shattering and mind-numbing experience.'" O'Connell described the filmmakers' drive that iteratively raises levels — from gunshots to explosions and also to music and dialog — until the whole movie is far too loud. "You see, for whatever reason, many filmmakers believe that they need to rip people's heads off to make an impression. They pump up the volume of their movies as if they're competing in the sonic Olympics, where they highest volume level wins." He has talked with theater owners, one of whom said he turns down the level until the complaints stop. O'Connell suggests that a possible solution is for features to adopt the loudness standards set for trailers, since he and many other sound mixers have "tried every trick in the book [to keep the levels down] and it hasn't helped."
- ***The Big Squeeze*** is Frank Wells's description of Meridian Lossless Packing (MLP; www.meridian.co.uk), which is used to enable DVD-Audio discs to hold a commercially acceptable length of music and features at the higher word length and bit rate the format supports. Wells explains that MLP is lossless, meaning that bits in equals bits out, and reported on tests showing that while MLP performed as advertised, certain dithering and reverb algorithms in specific hardware had caused some audible anomalies, which have since been corrected.
- ***Setting Up Multichannel Systems.*** Tom Holman once again explains how to calibrate individual playback system channel levels, including the subwoofer. He advocates bandlimited pink noise (500-2000Hz) to prevent room and speaker anomalies from effecting errant adjustments.

The Double Advent System

Harry Pearson's voice from the past (spring 1973 *Absolute Sound*, v1n1) celebrates what the title refers to, in a reprint as a 30th-anniversary feature in the August/September *TAS*.

American Cinematographer *(Sept 2003)*

- ***Pros and Cons of HD Dailies.*** Debra Kaufman (*The Post Process*) discusses the use of high-definition video instead of 35mm film for dailies during filming.
- ***Golden Years*** is how David W. Samuelson describes the evolution of widescreen film technology.

AES Journal *(September 2003)*

- ***Effects of Down-Mix Algorithms on Quality of Surround Sound.*** Søren Bech, Slawomir K. Zielinski, and Francis Rumsey evaluated the basic audio quality of eight algorithms designed to mix 5.1-channel sound down to fewer channels. "The results obtained prove the importance of the center channel, especially in the context of audiovisual presentations."
- **Measuring Envelopment.** Gilbert A. Soulodre, Michel C. Lavoie, and Scott G. Norcross tried to take *Objective Measures of Listener Envelopment in Multichannel Surround Systems*. They came up with a frequency-dependent means to quantify this effect, which "was shown to outperform other objective measures significantly."

ARRI News *(Issue 09/2003)*

This is a PR publication from Arnold & Richter Cine Technik (ARRI; www.arri.com), a worldwide company headquartered in Munich, Germany. ARRI is respected for its motion-picture cameras, lenses and lighting systems. They are expanding into digital cameras, recording systems and projectors designed for comfortable use by those familiar with film equipment.

- ***ARRI in the Digital Age of Film*** is an introduction by ARRI managing director Franz Kraus. He outlines some of the intense comparison-testing ARRI has conducted of CCD and film-camera technology, including Sony's 24p digital camera vs Kodak film. They have joined with Lockheed-Martin in the Blue Herring project, further comparing film and digital equipment, using an experimental camera with three Lockheed-Martin 12-megapixel 60x40mm CCD sensors (which theoretically provide the resolution of 35mm film). Kraus concludes that "if you aim for the big screen, highest possible image quality and the flexibility that is required in features [movies] and commercials, film is and will stay for many years to come the medium of choice." He recognizes the advantages of combining location/studio filming with digital video post production, and ARRI has systems for each function — "with the new generation of telecines and scanners you can exploit this latitude and … show details in dark and highlight areas you never could before and still cannot capture with digital acquisition."

- ***No Film But Still an ARRI*** is Michael Koppetz's writeup about their D-20 "digital film-style camera for TV applications." The camera is designed to function in every way possible just like a film camera, so directors of photography can make adjustments the same way they do for film cameras. The D-20 prototype "is based on a single, specially designed 6-megapixel CMOS sensor that features an image area comparable with that of a 35mm full-aperture negative." Because of oversampling at image capture, the 1920x1080 HD-format output is of higher quality than if a sensor of that pixel count had been used. The D-20 is being developed with support from "broadcasters BBC and France 2, the technology company Snell&Wilcox as well as researchers from INESC in Porto and the University of Padova." Another part of the project is to 'develop innovative methods of capturing and handling digital image data and to adapt these solutions to professional working practice." This camera can generate data at almost a billion pixels per second, based on the 6-megapixel sensor and 150fps (used for very slow motion); at two bytes per pixel, this converts to about 1.8GB per second. "The sheer volume of the captured image data remains a problem."
- Reimar Lenz, who is responsible for the specification and conceptual design of the CMOS-sensor used in the camera, notes that aside from the number of pixels, the readout noise, dynamic range, homogeneity and color fidelity have the greatest imfluence on image quality. He explains that readout noise determines how much detail can be perceived in the dark image areas, and that homogeneity problems — caused by offset errors in dark areas and gain differences in bright areas — can easily be corrected with signal processing.

Self-Destructing DVDs Bomb with Consumers

This report in the September 2003 *CE Pro* (www.ce-pro.com) on the FlexPlay DVD technology that causes the discs to be unreadable 48 hours after the package is opened. "CBS marketWatch reports the study found that 76% of consumers who participated in the survey indicated they weren't interested in renting a self-destructible DVD." Disney's Buena Vista Home Entertainment was rumored to have planned their use.

Reality TV

New ultra-HD technology has been demonstrated by NHK researchers (reported by Andrew Lee, 26 September 2003 *Engineer*; url posted on www.ilovehdtv.com) at the broadcast technology conference IBC in Amsterdam. UHDV has an image resolution 16 times that of HDTV, with about 4000 horizontal scan lines per frame. "The camera was built by aligning four 2.5" charge-coupled-device (CCD) image-capture panels. The projector system uses four liquid-crystal-on-silicon panels, two of which process green light while the other two each handle red and blue. …NHK engineers were originally able to make only 34 seconds' worth of recording. They have now built a disc recorder system made up of 16 HDTV recorder units with a capacity of about 3.5 terabytes, allowing them to shoot 18 minutes of UHDV footage."

Dolby 5.1-Channel Music Production Guidelines (www.dolby.com/tech/Multichannel_Music_Mixing.pdf) has been reported in the September 2003 *Pro Sound News* to be "the first such document to clearly present a technical blueprint for creating music in 5.1 channels." The suggested speaker layout is based on ITU-R BS.775-1, which shows the speakers at specific angles with their faces tangent to a circle around a single primary listening position. This is certainly different from the typical surround system layout, and implies a surround philosophy different from that used in generating movie soundtracks, the former being localization of instruments in the surround channels for a single listener, the latter being more general surround effects signals for a large audience.

Perfect Vision *(September/October 2003)*

- ***Home Theater — Commodities?*** Editor-in-Chief Robert Hartley sees the "Costco-ization" (as senior video editor Gary Merson put it) as an indicator of a problem: "The knowledge gap that develops when new and complex technologies are sold by folks who don't understand what they're selling to folks who don't know what they're buying. … The average consumer, already flummoxed by the jargon and complexity of HDTV interfaces, may just decide to hang on to his trustworthy analog TV rather than deal with [the many interface and encryption questions that face him]."
- ***LCD Widescreen HD Monitors — Consumers May Be Getting Less Than They Paid for*** is Gary Merson's warning as he points out that some LCD panels have an aspect ratio of about 1.67 (including a Sony and a Sharp), instead of the ATSC specification's 1.78 (met by all of the Panasonic units he measured).
- ***Who Will Retail Push HD Cable?*** is Merson's introduction to his report that Comcast has admitted to paying Best Buy for digital cable activations.
- ***When "High" Means "Standard"*** heads Merson's article informing readers that Starz is introducing a 480i widescreen channel called "Starz Hi Rez" but which is standard-definition video in widescreen format with a Dolby Digital soundtrack (which might offer 5.1 channels).
- **Fox Goes 720p.** Merson announced that Fox will begin transmitting about half of its prime-time programming in 720p60 high-definition format beginning with the 2004-2005 season.
- ***High-Definition Movies from Your PC*** are available using Windows Media 9. Patrick J. Megenity reports that WM9 is capable of playing back 1920x1080p30 high-definition video. I have seen a presentation at CEDIA in which Joe Kane used Windows Media 9 to present high-definition images in exhibition of a high-definition front projector, with exceptional results. Megenity explains how to use WM9 for high-def, and notes that it "could be a breakthrough for independent filmmakers shooting in HD to distribute their movies." Microsoft's web site offers HD clips for download.

TecHome Builder *(Sept/Oct 2003)*

- **Music Top Choice?** A Parks Associates study revealed that among media applications, US heads of households prefer listening to music (59%) over using the PC (53%), with watching tv third (51%), followed by watching movies (38%), viewing home photos (31%) and vewing home movies (16%).
- **Really High-Tech.** Although this is hardly audio or home theater, it belongs in the "technology has really gone too far" department: "With Toto's Neorest toilet you may never want to leave the bathroom [now there's an article lead — DRM]. Besides offering a heated seat, the unit will wash you, massage you with a warm oscillating spray, and dry you when it's done. It will automatically flush itself (varying the amount of water to suit the job at hand), then deodorize the air. The seat opens and closes by itself. The bowl is coated with Toto's SanaGloss glazing, which seals the porcelain with an ionized barrier that prevents particles from adhering to it. Price: $4500 to $5200" (www.totousa.com). [It better have a speakers option if we like to listen to music so much — DRM.]

Voice Coil *(November 2003)*

- **The *BAS Test CD*** is succinctly described, including track and acquisition information.
- ***Mastering Audio: The Art and the Science*** (BKB88; $40; available from Old Colony Sound Lab, Peterborough, NH; 888.924.9465; custserv@audioXpress.com) by Bob Katz was reviewed by David Moulton. The five-part book is a guide both practical and theoretical to mastering the art of mastering recordings for commercial release. Moulton's review not only thoroughly describes the book and its benefits but points to its few failings. Katz's style could be more fluid and organized, Moulton finds, and he takes firm exception to Katz's unfounded subjectivism, but otherwise he lauds the book and message.

Nonlinear Distortion and Perceived Sound Quality

Chin-Tuan Tan, Brian C. J. Moore and Nick Zacharov (November 2003 *AES Journal*: "The Effect of Nonlinear Distortion on the Perceived Quality of Music and Speech Signals") examined "the effect of various types of nonlinear distortion on the perceived quality of speech and music signals. … The subjective ratings were compared to physical measures of distortion based on multi-tone test signals. A distortion measure, DS, derived from the output spectrum of each nonlinear system in response to a 10-component multi-tone signal gave high … correlations with the subjective ratings. … It was concluded that an objective measure of nonlinear distortion based on the use of a multi-tone test signal can predict the perceptual effects of nonlinear distortion reasonably well."

Really Big Shows

Michael Riggs's straightforward advice (December 2003 *PC World*) is aimed at those looking for large-screen HDTVs. Riggs explains the SDTV and HDTV formats, describes the advantages and disadvantages of plasma, LCD and DLP technologies at different as well as overlapping price ranges, and reminds the reader to think about the digital video connectors needed and, for front projectors, to look into fan noise. He does not mention the need for proper set adjustment by a professional in the home.

Widescreen Review *(November 2003)*

- ***Digital Video Essentials.*** Joe Kane provides *Test Pattern Descriptions* in part two of the series on his new test DVD that also comes in 720p60 and 1080i30 high-definition D-VHS D-Theater tape formats. Kane continues explaining why DTV performance isn't up to snuff: "Trying to describe how to properly set brightness and contrast in a world of digital circuits improperly driving the display becomes a daunting task. There are so many places where things do go wrong ahead of the display that pinning down the cause of a display error requires test equipment to look at every stage of processing. I've had to develop new test signals to help point out where things are going wrong and I'll have to put up with these errors until someone [who controls DTV circuit design] understands that even in digital circuits they should be designing for the real world. … Many DVD players, D-Theater vcrs, computers and some HDTV set-top boxes partially destroy the signal quality before it ever gets to the video output by not including dynamic range below black and/or above white. Many video processors won't accommodate a 100% chroma dynamic range, so you'll see colors going into hard stops [flat, pasty areas — DJW] as well as black and white." Kane's new *Color Bars with Reference Grey* test pattern is a phenomenal advance in test pattern design, since, with no test equipment other than the red, green and blue filters, even the consumer can readily see if the color decoder (whether it is in the display or in a separate video processor) is working properly or emphasizing one or more colors (which most are, for marketing reasons). Kane also explains quite clearly why the contrast ratios advertised are meaningless and, in my opinion, fraudulent.
- ***FCC Adopts Plug&Play Compatibility Standards*** is reported by Paul Sweeting, who clearly and succinctly explains what the agreement covers and the politics behind it. The agreement doesn't solve all the problems, but helps with some of them. It is nice to learn that the consumer wasn't totally sold out, yet — the broadcast flag, wanted so strongly by Hollywood, will be considered separately.

Futurist *(November/December 2003)*

- ***Better Music Through Science*** is an uncredited report from the Imperial College of London press office (www.ic.ac.uk) that "Neurofeedback techniques designed to help improve memory may also help musicians improve their performances." A study coauthored by Tobias Egner at London's Royal College of Music indicated that using neurofeedback "improved [the test subjects'] scores on a standard evaluation in comparison with students receiving other forms of performance-enhancing training and those who received no additional (non-musical) training."
- ***Online Music: The Sound of Success*** is Eric Garland's take that "the online music industry is turning the traditional music industry on its head" by making "music into nearly a pure service industry," which it used to be before recording turned it into a hard good (records, tapes, CDs) industry. "These new business models will allow customers to buy the experience of music wherever they are, bypassing expensive middlemen, supporting creative artists, and likely increasing profitability both for the musician and their record labels." He sees the current hard-

product distribution part of the business as the moneymakers, with the artists typically getting little for their efforts. He also sees that same group as those most likely to lose in the new e-music distribution scenario: “Musicians want to be paid for creativity and showmanship, and record companies want to be paid for making stars. Information technology does not threaten the need for either of these skills in the coming decades.”

Widescreen Review *(December 2003)*

Calibration Equipment Shootout. This is an interview/discussion among Jim Burns (contributing editor), Gary Reber (publisher/editor-in-chief), Perry Sun (managing editor), Mark Hunter (Milori), Cliff Plavin (Progressive Labs), and Jeff Murray (Sencore). Burns directed a comparison of three color analyzers (Milori’s ColorFacts CF-6000, $2500; Progressive Labs’ CA-1SE, $1600; and Sencore’s CP5000, $5000) against two Photo Research PR650 units (~$15,000), which are “considered to be the reference spectroradiometer in the industry.” The analyzers measured seven displays that included LCD, DLP, CRT and D-ILA front and rear projector and direct view technology. “The projectors were either brand-new out of the box and uncalibrated, or they were slightly off calibration. … We didn’t want everything to be exactly perfect, … so we could get a good feel for how these would work in the field.” Burns noted that “the accuracy of your display’s image depends on many parameters, [including] the color of white [which must be correct or] none of your display’s other colors will be correct.” There were inconsistent mismatches in the readings of the two reference PR-650s, attributed in part to different angles at which they were aimed at the display. He also pointed out that the color measured is affected by ambient light reflecting off the display. Plavin suggested that for non-CRT displays, calibration should await about 100 hours use so the lamp can stabilize (others agreed). Hunter noted that all the devices have some problems providing accurate readings at low display light output (around 20 IRE), which he said “was actually closer to about 2% of the light output due to the gamma characteristics of the display.” Plavin also recommends turning down the contrast right out of the box to prevent burn-in, which can occur in plasma displays within relatively few hours.

Digital Video Essentials DVD. Greg Rogers walks us through the contents of this “Next Generation Home Theater Calibration” DVD.

Digital Video Essentials: Test Pattern Descriptions. In this third part of the serie, Joe Kane continues his description of the test patterns and provides background for their creation.

Apologia Anamorphic is Jim Taylor’s “formal written defense” (the definition of *apologia*) of using the term ‘anamorphic’ to describe squeezed images on DVDs. He explains that the word predates even film, so “there is nothing special that restricts its use to the world of camera lenses and projector lenses.”

Tech Insider is Perry Sun’s “news and views in technologies for audio and video.” Sony has chosen to include a hyphen in SA-CD, apparently to emphasize its relationship to the CD. Sun also reports on the rise of Microsoft’s Windows Media 9 in the world of high-resolution audio.

One Installer’s Opinion includes Terry Paulin’s realization that it is “impossible to overstate the current proliferation of misinformation. … I attended a product line show last week and heard a factory representative commit four technical errors in the same sentence while he espoused the merits of his new line of DLP rear projectors. I am sometimes embarrassed for our industry. Advice — read vociferously and question it all.”

The Fear of Piracy. Paul Sweeting has concluded that “as the battle over how to respond to the digital revolution has grown more pitched, it’s drawn unwelcome but overdue attention to whose interests the major industry lobbying organizations really represent, and to the nature of the relationship between the artists and the companies they work for. … In the music industry, … the labels’ increasingly desperate efforts to stem the loss of revenue from declining CD sales and inadvertently focused attention on how little revenue most artists ever see from the sales of their recordings, further driving a wedge between the two. In the long run, that wedge could do more to undermine the labels’ control of the business than their losses from piracy.”

Reprinted, with permission, from the 19 May 2003 Stereophile; *http://www.stereophile.com/news/11649/index.html. Some of the graphs referred to in the text have been omitted in this reprint for reasons of legibility; please go to the website to see them in color and with increased clarity — eds.]*

Dark Side of the Disc

by John Atkinson

The June [2003] issue of *Stereophile* … spills some ink on the 30th-anniversary reissue of Pink Floyd's *Dark Side of the Moon* as a two-layer Super Audio CD (Capitol CDP 582136 2). Jon Iverson nominated the disc as June's "Recording of the Month," while I mentioned it in my "As We See It" column. This "fully loaded" SACD includes both multichannel and two-channel mixes encoded with the DSD system on a high-rez SACD layer and a two-channel "Red Book" transfer (16-bit word length, 44.1kHz sampling) on its CD layer.

Jon wrote at length about the audible differences between the SACD-layer mixes and the CD-layer transfer, while I wrote about the measured differences. As the single-page format of the paper magazine's "As We See It" and "Recording of the Month" columns doesn't allow space for illustrations, this web article should be read in conjunction with both. (Jon also wrote about the Crest-pressed SACD of *DSotM* developing radial cracks at its center in an online article at http://www.stereophile.com/news/11635/.)

Both two-channel mixes of *DSotM* were claimed by EMI to have been transferred straight from the original analog master tape. It came as a surprise to both Jon and me, therefore, to hear relatively large differences between the SACD and CD versions, not the least of which was that the latter was louder. [As their analyses presently show, and the authors are too polite to say so, this EMI statement appears to be a bald lie — DRM.]

In addition, since I recorded an album at Abbey Road Studio at the same time that the Floyd were there making *DSotM*, I always thought the album did an excellent job of preserving the characteristic sound of the studio with which I had become so familiar. Yet when I first listened to the CD layer of the [SACD] reissue, it didn't sound like Abbey Road at all. The sonic subtleties that identify the recording venue and its unique reverb chamber had been eliminated or smoothed over. They *were* there on the SACD [DSD layer], so some investigation was called for.

I used the analytical capabilities of Syntrillium's Cool Edit Pro digital audio workstation PC program to look at the differences. I first ripped the CD-layer version of "Money" using Exact Audio Copy (www.exactaudiocopy.de), a freeware program that is the best of all the PC-based programs I have tried. SACDs can't be read by computers, so to get a version of the DSD data that could be read by Cool Edit Pro, I digitized the analog outputs of a Musical Fidelity Tri-Vista SACD player (www.stereophile.com/showarchives.cgi?838) with a Metric Halo (www.mhlabs.com) Mobile IO A/D converter connected to my Apple Titanium PowerBook with a FireWire link. To make sure I captured all the audio content of the DSD-encoded layer, I ran the Mobile IO at 96kHz sampling and 24-bit bit depth, and I used Bias (www.bias-inc.com) Peak 3.2 to create an uncompressed two-channel AIFF file that I ported to my PC for analysis.

To avoid inadvertently clipping the music in the transfer, I used the Mobile IO's excellent meters to keep the peak level below -3dBfs, then normalized the file to 0dBfs using Cool Edit Pro.

The table at right [1] shows the statistical data generated by Cool Edit Pro for the two PCM files. Both peak at 0dBfs, though the right channel is loudest for the CD data, the left for the data derived from the DSD layer. There are no clipped samples in the DSD data, but a whopping 362 in the CD data. There is also some dc offset apparent in the latter. More important, while the peaks of each file are the same, the average rms power of the CD data (-15.29dB average of both channels) is significantly higher than that of the SACD layer (-17.7dB). No wonder the CD layer sounded louder. This suggests compression or peak limiting was used in the mastering to reduce the song's dynamic range compared with the SACD [DSD mix].

Figure 1 (not shown; see website) shows an FFT-derived spectral analysis of the SACD data, looking at the entire file. A similar spectral analysis for the CD data was more or less identical below 20kHz (vertical white line), which surprised me. Above 20kHz, the musical content on the SACD falls off rapidly, so that above 30kHz or so, all that remains is random noise, presumably from the original analog recorder used to master the recording. (Although SACD has a nominal signal passband to 100kHz, this graph shows that it was perfectly appropriate for me to digitize the DSD-encoded "Money" at 96kHz.) Above 30kHz in this graph, you can see the rising noise floor typical of DSD data. This results from the aggressive noise shaping used to get sufficient dynamic range in the audio band from a 1-bit quantizer.

Figure 2 (not shown; see website) shows the entire "Money" file (SACD version) with time along the x-axis (horizontal), frequency along the y-axis (vertical), and amplitude encoded as color. (White is highest in level, then yellow, red, through purple, to black meaning no content.) The ultrasonic DSD noise can be seen as the pinky/purple hash running along the top of the two graphs. (The left channel is on the top, right on the bottom.) The horizontal line at 22kHz indicates the theoretical limit of the CD medium; all that lies above it as far as music is concerned are thin vertical red lines representing Nick Mason's cymbals.

Figure 3, next page, is a more conventional way of looking at the same file data, whereby the green trace is a representation of the analog waveform. The dynamics of the song can be clearly seen. The low-level jingling cash intro swells a little when the familiar riff joins in, and is followed by two verses and the saxophone solo, all of which peak at or below -3dBfs. The first electric guitar solo peaks a little higher, at -1dBfs, and is followed by a dramatic drop in level as the guitar, bass, and electric piano vamp over sparse drum figures. The second guitar solo raises the occasional peak to 0dBfs, with then a drop in intensity for the third and final verse and the fade-out.

Figure 4, also next page, shows the waveform display for the CD transfer of "Money." It is *very* different. The first two verses and the sax solo have had their overall levels raised to be almost as high as the following guitar solo. The solo, however, has the squared-off shape that results when the music has been run through a peak limiter. This is a device that literally chops off the transient peaks, allowing the average level to be higher, hence louder. The vamp in the bridge is now significantly higher and the second guitar solo has the same squared-off shape as the first, with severely clipped transients. The right channel in the second solo bangs up hard against 0dBfs. This is where the 362 clipped samples reported by Cool Edit Pro come from.

Pink Floyd: "Money" Statistics, calculated with Cool Edit Pro

CD Layer (Sample Rate = 44.1kHz, Bit Depth = 16)

	Left	Right
Min Sample Value:	-31380	-32768
Max Sample Value:	31692	32767
Peak Amplitude:	-.29 dB	0 dB
Possibly Clipped:	0	362
DC Offset:	-.002	-.002
Minimum RMS Power:	-64.33 dB	-65.85 dB
Maximum RMS Power:	-5.74 dB	-6.4 dB
Average RMS Power:	-15.37 dB	-15.19 dB
Total RMS Power:	-14.28 dB	-14.2 dB
Actual Bit Depth:	16 Bits	16 Bits

Using RMS Window of 50 ms

SACD Layer (Sample Rate = 96kHz, Bit Depth = 24)

	Left	Right
Min Sample Value:	-32768.97	-31090.8
Max Sample Value:	32247.5	31246.24
Peak Amplitude:	0 dB	-.41 dB
Possibly Clipped:	0	0
DC Offset:	0	0
Minimum RMS Power:	-64.77 dB	-65.85 dB
Maximum RMS Power:	-7.96 dB	-8.8 dB
Average RMS Power:	-17.7 dB	-17.6 dB
Total RMS Power:	-16.61 dB	-16.62 dB
Actual Bit Depth:	24 Bits	24 Bits

Using RMS Window of 50 ms

Table 1

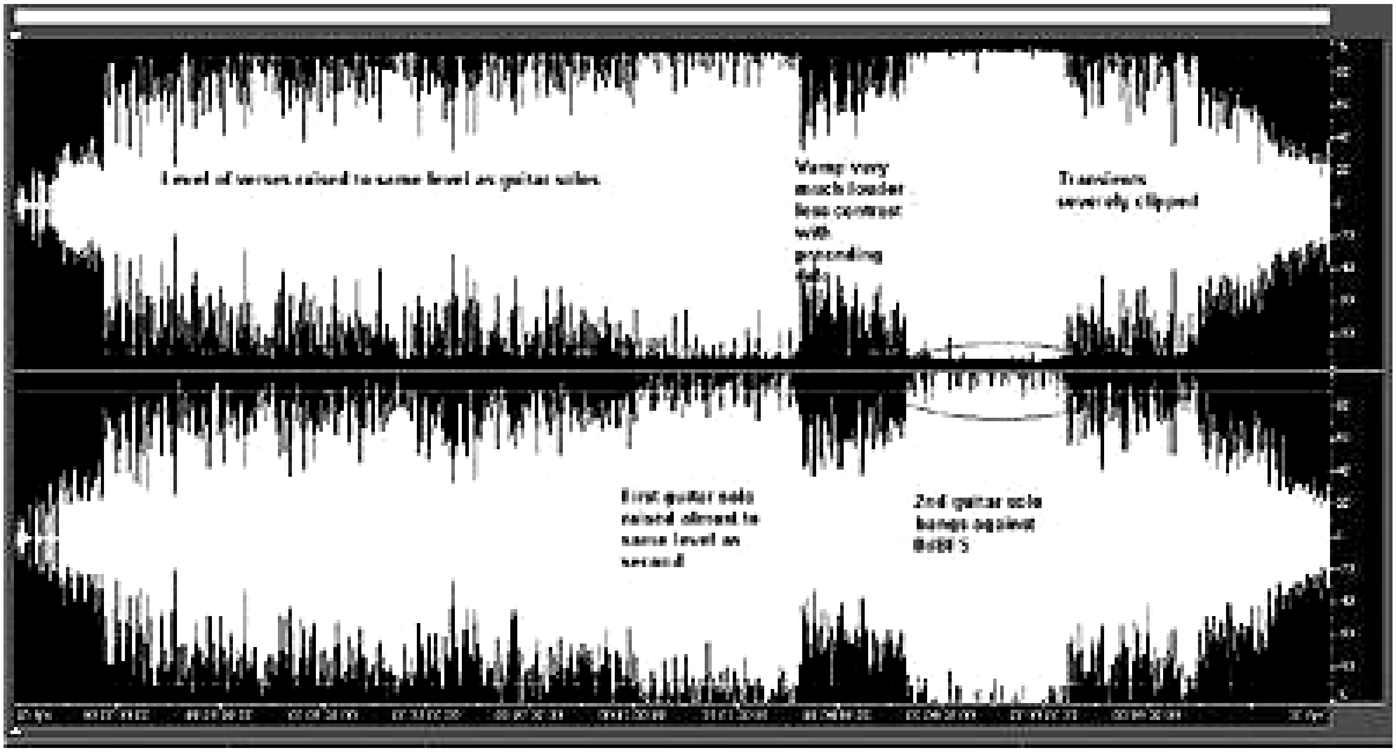


Figure 3: "Money," SACD layer, waveform view of entire track (20% full-scale/div vertical scale)

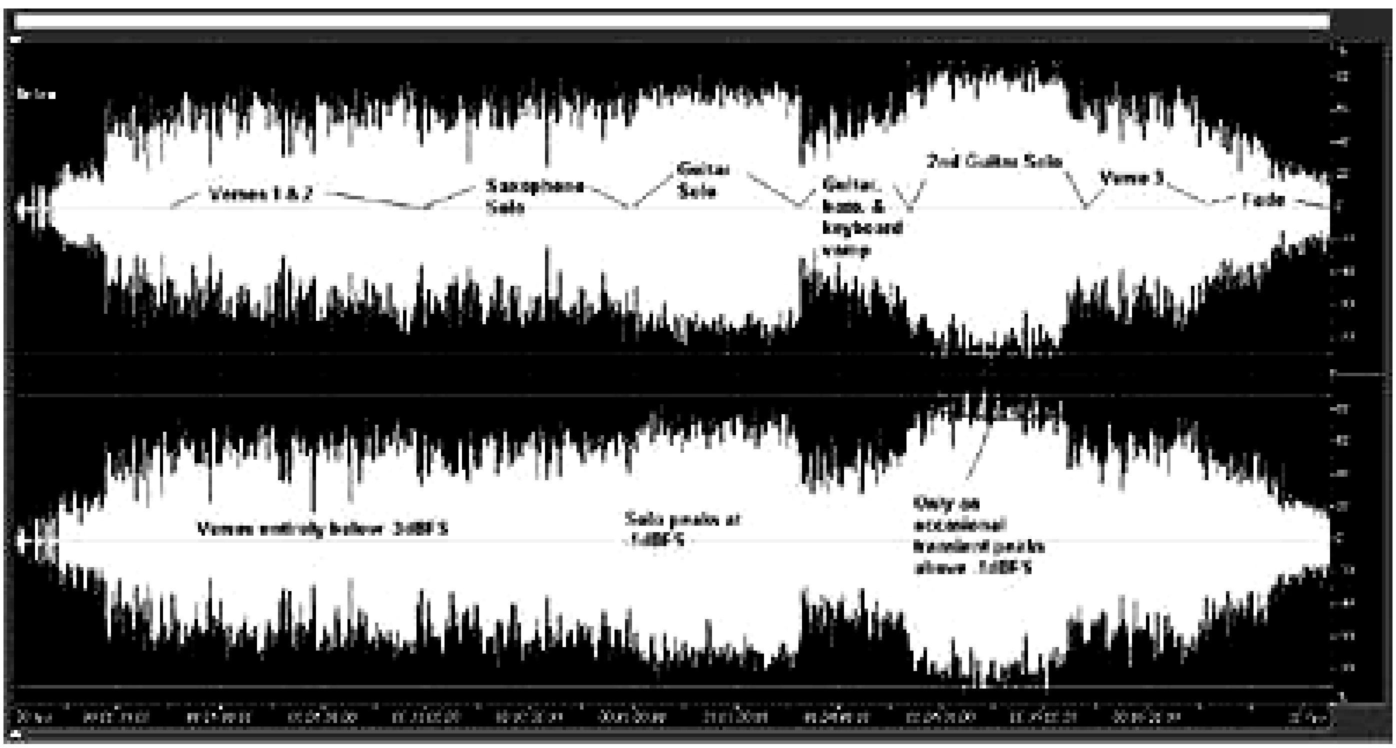


Figure 4: "Money," CD layer, waveform view of entire track (20% full-scale/div vertical scale)

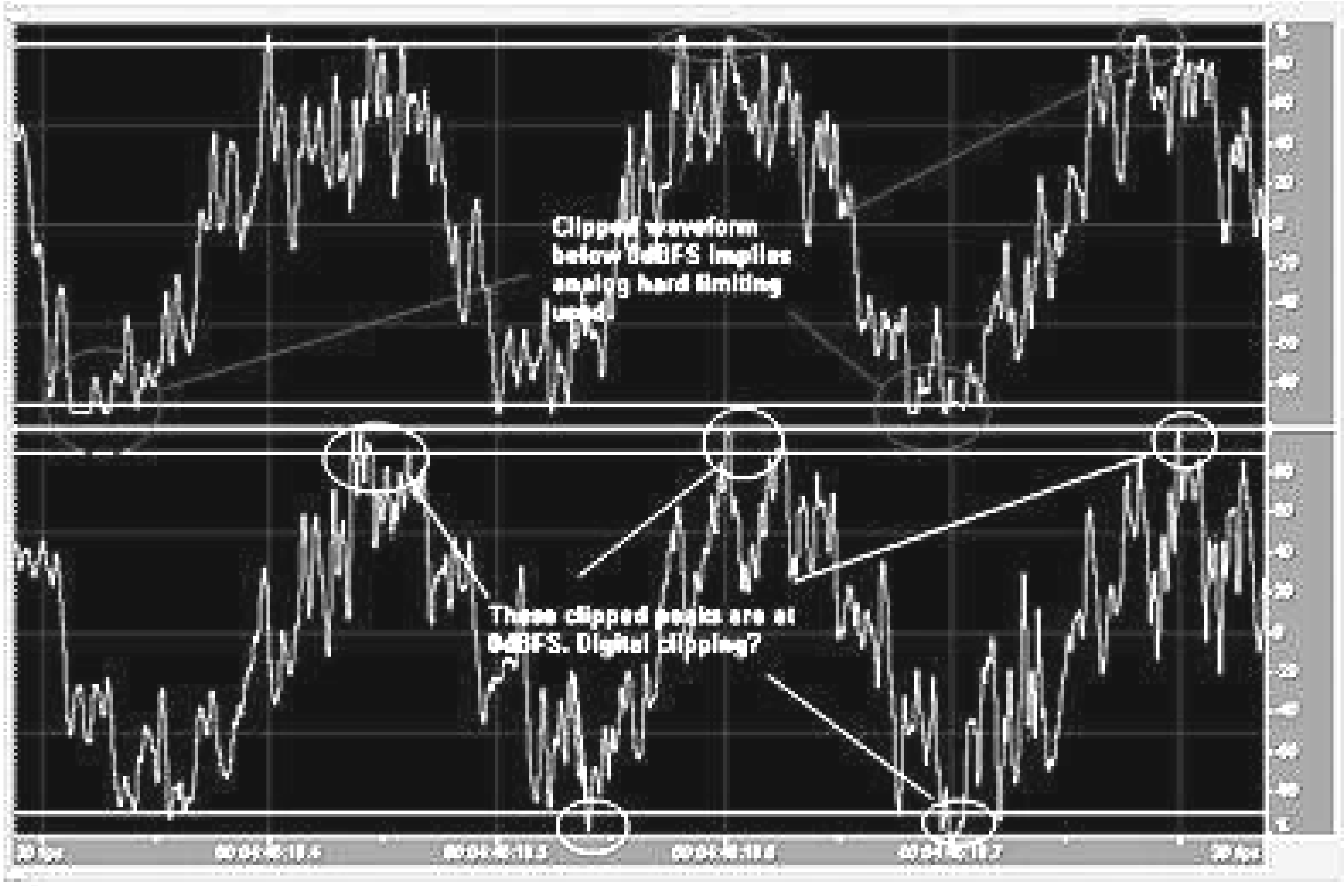


Figure 5. "Money," CD layer, waveform during second guitar solo at 4:46

Figure 5, above, shows a very short example from this solo. Yes, the transients in the right channel are flat-topped, suggesting digital clipping. But so are the transients in the left channel, and as this occurs about 0.5dB below 0dBfs, it can't be clipping in the digital domain. It looks as if the mastering engineer who did the transfer for the CD layer ran the analog tape machine into an aggressive peak limiter. He didn't do so for the SACD layer, however, as there is no evidence of flat-topping of the waveform in the SACD version of "Money."

The ear is relatively forgiving of this type of clipping when it happens on an occasional basis. It is over almost as fast as it happens. But when it occurs over and over again, as it does here, the result is fatiguing. It is, after all, distortion.

Why would someone do this? The cynic would suggest that it was to make the SACD transfer more transparent-sounding than the CD transfer, with less grain and greater dynamic contrasts apparent. Maybe. However, you then have to deal with the fact that to untrained ears, "louder" is always "better," and the CD layer does indeed sound louder.

I suspect, therefore, that the work was done on the CD layer so that it didn't sound too different from the dynamically crippled norm that non-audiophiles have come to expect as "CD sound." Certainly if you do a Google search for comments on this SACD, many people have written how much they like the CD layer, describing it as having more "impact," more "punch."

But the difference negates any comparisons between the two media, at least using this recording.

Jon Iverson Adds Some Comments

I was puzzled when reviewing the *DSotM* disc: the CD layer sounded more aggressive than the hybrid's SACD tracks. Not having access to the test equipment John Atkinson (JA) has on hand, I chalked the differences up to varying characteristics of the two analog-to-digital converters (one PCM-based, the other DSD) used for each layer and the more laid-back qualities of SACD sound. In the end, I found myself more closely comparing the SACD layer to the excellent new vinyl release that was mastered at AcousTech and pressed with metal parts by RTI.

Now, with the evidence from JA's graphs, I wonder what the point of limiting the CD's dynamics on a special release like this could possibly be. I've speculated that EMI may have wanted to give the CD layer more "punch" since it is likely the one to be played on the radio. Or perhaps, as JA notes, EMI and Sony have conspired to place DSD in a more audiophile light with this manipulation — which is troubling when you start to ponder which other hybrids might have been altered in this manner.

But, like JA, I'll guess the answer is actually more of a mundane "business-as-usual" attitude at the CD mastering house. The paranoid audiophile in me suspects that the major labels now make it standard practice to push the audio level on all of their rock CDs to give them a more in-your-face sound. The evidence I've read in *Mix* magazine and that JA and others have gathered would support this contention. I can almost see the young EMI exec jabbing his finger at the mastering engineer and shouting, "The audiophiles have got their prissy SACD layer, now make the other one ROCK!"

So, when it came to the new *DSotM* rerelease, they simply applied the standard substandard treatment.

Lexicon's Subwoofer Logic

by David J. Weinberg

[This was sent to Lexicon in the hope it might prove of value to them — DJW.]

With all my Lexicons (CP-1, DC-1, DC-2, MC-12) over the years, I have preferred a 7.1-channel configuration, non-THX. With my speakers I would prefer crossovers set at:

Left and Right Front: 40Hz highpass (HP);

Center: 80Hz highpass (the Lexicon feeds signal below this to both Left and Right Front outputs);

Left and Right Side: – 40Hz highpass;

Left and Right Rear: 60Hz highpass;

with the single subwoofer set for a 120Hz lowpass (LP), so it handles the whole LFE signal *plus* all the seven channels' audio *below* the above-listed highpass-crossover frequencies.

Of course this assumes that the processors feed the remainder (the signals below the seven channels' HP crossover settings) to the subwoofer, regardless of the subwoofer's LP crossover setting, as long as the frequencies are below that setting.

My assumption was that in no case would both a main channel speaker and the subwoofer radiate the same signal due to bandwidth overlap.

From studying my MC-12 manual (which I found inadequate in this area), running some tests on the MC-12, and communicating with Henry Hecking of Lexicon support, who has been diligent in providing assistance, I have come to learn that the MC-12 doesn't work this way.

Since I now have test equipment that enabled me to test the MC-12's subwoofer logic, I found what I consider a problem with how the HP/LP filter system operates. The compromise solution in the following applies to the MC-12 only, and assumes that I have set the main channel crossovers as listed above.

If I have only one subwoofer, it clearly must handle the leftover LP signals from the seven main channels (below the lowest HP crossover setting among the main channels), *plus* the LFE signal.

However, as the MC-12 currently works, I find that if I feed that subwoofer from the Subwoofer 1 Output (with the MC-12 set for a single subwoofer and no LFE speaker connected) and set the subwoofer LP to lower than 120Hz, the upper portion of the dedicated 120Hz-bandwidth LFE signal, which is capable of being up to 10dB louder than the main channels, gets fed to the Left and Right Front speakers, which might not cleanly handle the extra level at those frequencies.

If I set the subwoofer crossover to 120Hz so as not to risk overloading the main channels with this LFE signal, then the difference between the main channel crossovers and the 120Hz LFE crossover setting is now radiated by too many speakers: the respective main channel(s) and the subwoofer, resulting in double output (in this case between 40/60Hz and 120Hz), which results in boosting that overlapped range above the correct level.

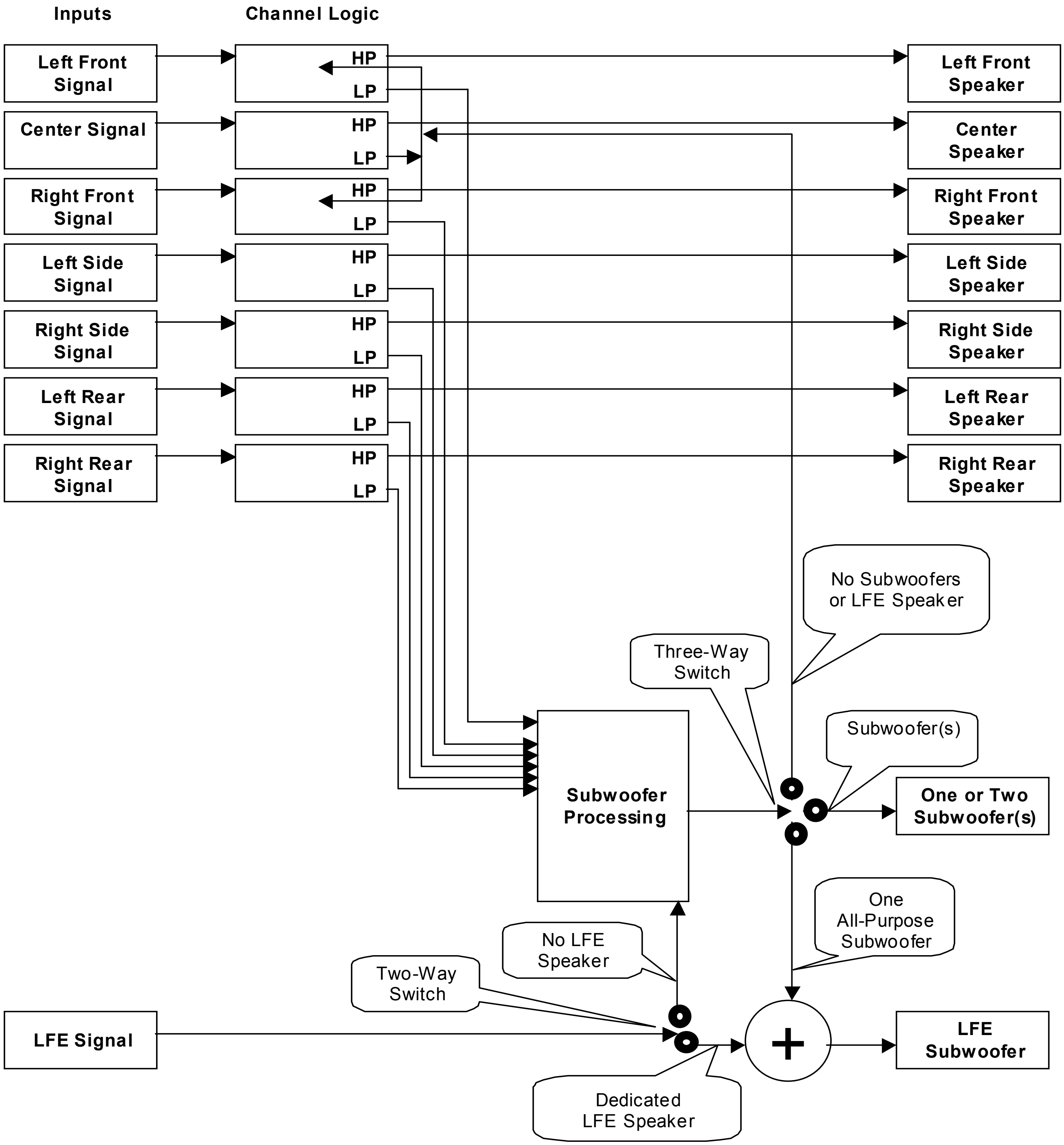
The only compromise solution I could find is to set the Left and Right Front to full bandwidth, so all the LP signal from the other channels is fed to the Left and Right Front (this also assumes that the Left and Right Rear bass frequencies are fed to the Left and Right Front, not to the side channels), and to feed the subwoofer from the LFE output, telling the MC-12 that I have no subwoofer, but only an LFE speaker. This stresses the main channels more than I want to.

It seems to me that the prodigious computational power and digital logic already in the Lexicon MC-12 would support the LF logic shown in the following diagram, which would allow any of three possibilities — one or two subwoofers, and even an additional dedicated LFE subwoofer — without overlapping output and without sending part of the LFE signal to the main speakers. The channel logic in this diagram is assumed to be such that once a channel's HP crossover frequency is set, that channel's LP is automatically set to match. I have also assumed that in the "No Subwoofer" configuration, the Left and Right Front speakers are set to full bandwidth (automatically causing zero output from those channels' LP filters), and the "No Subwoofer" feed from the subwoofer channel back to the Left and Right Front is added into the signal before being fed to the respective main speakers.

Note that all four subwoofer possibilities (from NONE through three) are addressed by this logic. The diagram shows simplified logic around having one or two subwoofers, but the concept is shown.

Only if there is no subwoofer would the main channels have to handle any of the LFE, and, with more convoluted logic, perhaps it could be fed to all seven main channels in parallel, allowing each one to handle as much of the signal as possible, thus sharing the load (based on their HP settings).

I believe I haven't missed any configuration idiosyncrasies, but would be interested to hear if I have.


Inputs
Channel Logic
Left Front Signal
Center Signal
Right Front Signal
Left Side Signal
Right Side Signal
Left Rear Signal
Right Rear Signal
HP
LP
Left Front Speaker
Center Speaker
Right Front Speaker
Left Side Speaker
Right Side Speaker
Left Rear Speaker
Right Rear Speaker
No Subwoofers or LFE Speaker
Three-Way Switch
Subwoofer(s)
Subwoofer Processing
One or Two Subwoofer(s)
One All-Purpose Subwoofer
No LFE Speaker
Two-Way Switch
LFE Signal
+
LFE Subwoofer
Dedicated LFE Speaker

CEDIA Expo 2003

by David J. Weinberg

The Custom Electronic Design and Installation Association (CEDIA) exposition has grown explosively over the last few years and taken many of the home theater manufacturers' new product exhibits away from the International Consumer Electronics Show (CES). From what I saw at CES last January in Las Vegas and at CEDIA Expo this recent September in Indianapolis, CEDIA has become *the* show for home theater, home automation and home systems integration. It will be held in Indianapolis for the next two years before migrating to Denver (more room).

CEDIA Expo 2003 claimed 22,000 attendees and offered them around 60 product-specific training sessions plus about 120 different courses (some more than once,to accommodate the interest; total student count was 16,010, although many individuals took more than one course). 458 took the courses designed to prepare one for the custom installer/designer certification exams (422 passed out of the 476 who took the three different exams given at this expo). Although www.cedia.org lists member companies and their certified employees, CEDIA has not yet set up procedures to assist certified individuals find employment in the field.

For almost all of the courses, the handouts could be downloaded even before the expo (under password control for each course purchased). I audited several:

Home Networking … The New Frontier. Gordon van Zuiden gave an overview of home networking and how integrated the home is rapidly becoming, with everything from PCs and home theaters to appliances and security being tied together, plus the ability to use the Internet to monitor and control it all remotely (secure password protocols are essential!). He also addressed the installer's perspective: how to get into and manage this facet of the business.

Room Acoustics: The Room & Loudspeaker System was a treat, since Infinity's Floyd Toole was the speaker. From the course outline: "The room is the final component in any audio system, and it, combined with the arrangement of listeners and loudspeakers within it, are major determinants of the quality of the listening experience." Toole covered all the bases, including his analytical and experiential conclusion that a subwoofer in each of the four corners, or at the center of each of the four walls, yields the best results, with the admonition that equalization should be used only as a last resort. He also suggested that less-solid walls is not a bad feature, since it allows bass attenuation (through absorption and transmission — lossiness) that more closely matches the mid- and high-frequency attenuation of most rooms, resulting in more uniform reverberation time as a function of frequency. Toole's research-based philosophy holds not only that the best-sounding speakers have a smooth and flat room response but that their dispersion with frequency is uniform. [By now Dr. Toole is entrenched as *the* loudspeaker-science guru, and one hopefully trusts that his thinking remains informed by a vital empirical spirit. Papers of his currently on the Infinity website and adaptations of same in a recent *Audio Critic* which discuss rooms and speakers, however, contain what seem to me to be some confusion and surprising, unsubstantiated assertions and "sweeping assumptions." He apparently conflaties the two different phenomena of near-corner boundary-augmentation ripple (the Allison effect) and room resonances; he alleges the audible inadequacy of third-octave equalization and hence need for finer, sixth-octave resolution (and this was for bass! One might believe it superior for, say, headphone auditioning, but for in-room playback for normally head-moving listeners?); and he takes as a given the supposedly hierarchical strengths of 1-, 2-, and 3-boundary reflections (that is, axial, tangential, oblique, the latter two of which Toole wrongly claims can be more or less ignored) — DRM.]

Room Acoustics: Isolation & Noise Control was Steve Haas's (principal, SH! Acoustics) contribution to our education, as he detailed "the methods for controlling noise and vibration from HVAC, plumbing, electrical, lighting and audio-visual systems."

Display Device Calibration with Imaging Science Foundation's (ISF) Joel Silver was entertaining and informative, although less technical than a Joe Kane presentation. There was enough substance to the session that it was a good review for someone like me who received his ISF certification from Kane himself.

High-Performance Home Theater Calibrations is a pet concept of Anthony Grimani's (formerly of LucasFilm THX), who covered both the video and the audio portions of the system. Showing his movie theater background, Grimani said that "Home theater is a group experience — not for one person," and that the goal is: "Every Seat Is a Good Seat." Grimani agreed with Toole about using four subwoofers to optimize bass distribution in the room, and that less-solid walls is not a bad feature, adding that it also dampens room resonances, smoothing out low-end frequency response throughout, for a larger listening area. He noted that some surround processors' internal test signals will generate an output about 4dB different from some test DVDs owing to a different setting of the Dolby Digital DialNorm parameter (27 instead of 31; www.dolby.com has explanations of DialNorm, although the descriptions could be clearer).

Understanding, Finding & Eliminating Ground Loops was Bill Whitlock's advanced level course on the subject. As the president of Jensen Transformers, Whitlock is knowledgeable, and effectively explained this most obdurate problem with 168 informative slides, including many well-designed graphics. He explained why some mythical solutions don't work and provided a clear, straightforward technique for analyzing various ground loop situations, including step-by-step troubleshooting using a clever and easily built test connector. He recommended Belden 8241F as a great audio cable because of its exceptionally low-resistance shield, which keeps ground loop currents to a minimum, and even explained why 85% braid shield coverage, combined with a foil shield, is fine for A/V use.

The 16 other course presentation materials I reviewed all provided information of interest that would help custom installers deliver more correctly designed and setup systems to their clients.

The Best High-Definition Image I Have Seen: Samsung contracted with Joe Kane to guide the design of their SP-H700A DLP projector. Kane gave the demonstrations, using Windows Media 9 high-definition source material. In all the years I have been following HDTV, this projector's image was the best home theater HD image I have seen. If I didn't own a projector, this Samsung would be the one I would buy.

Recently, every image I have seen from a pixelated projector (DLP or LCD) has been on a screen with a gain of no higher than 1.0, and most were on a screen with a gain of 0.8-0.9 (lossy, throwing away light to make black levels look better, since pixelated projectors have not been able to generate good black levels); this notwithstanding, the black levels still don't look right. Yet even though Kane used his Stewart StudioTek 130, which has a gain of 1.3, the black levels here were outstanding, with exceptional detail in dark images.

This projector includes features and design characteristics not found in other units, including digital video processing from below digital video black to above digital video white (amazingly enough, this does result in eliminating the squashed, flat character of the image near these two limits), very quiet operation, and the new JKP calibration standard built in, making setup much easier for the ISF-certified technician. To top all this off, the unit I saw didn't even include TI's latest HD2+ DMD, which will be used in production units. Kane has shown this unit to TI's people, who were impressed. At a $10,000 srp, it is the unit of choice, and to me beats anything else that I have seen up to at least three times that price.

HT Calibration Tools

Ovation Software (www.ovationsw.com; 800.572.3917) is about to release *AVIA Professional* ($400), a substantial expansion of the current *AvIa* DVD test suite. *AVIA Pro* will offer more than 1000 video and audio test signals, with over 3000 variations, many aimed at widescreen displays of various technologies. Additional components include *CEA Standard Method of Measurement for DVD Players*, SMPTE monitoring technology (to alert users to equipment errors in the creation of motion imagery), and RPG's *Room Optimizer* and *Room Sizer* (www.rpginc.com; software for the home theater and audiophile room designer that uses a least-squares-error method to optimize perceived frequency response).

GoldLine (www.gold-line.com; 203.938.2588) distributes the $150 *5.1 Audio Toolkit* DVD (www.audiotoolkit.com), created by Anthony Grimani, who worked for Dolby and THX before starting his own consulting firm. While the included tutorial is also on www.audiotoolkit.com, the test signals are only available on the disc.

Home Theater System: Boston Acoustics (www.bostonacoustics.com; 978.538.5000) announced and demonstrated the $4000 Avidea 770, a home-theater system complete except for display. The Avidea control center (the size of a small receiver), is ruled from an LCD remote, and includes a DVD/CD player and an AM/FM tuner; it even has a zone 2 output for another room. The powered subwoofer houses the other customized amplifiers, which feed the six satellite speakers. It sounded BA-good, and provided more clean sound than one would expect from speakers that small.

HT Speakers: Allison Acoustics was present with some new redesigns (in elegant cabinets) of the classic Allison speakers, including a center channel that is a variation on the old Allison Four.

Shhh!: Quiet Solutions exhibited soundproofing products, which include construction materials such as sheetrock and plywood substitutes, adhesives, and sound-deadening coatings. www.quietsolution.com offers download of five informative booklets plus independent-lab tests of their products.

For those who have music performed live in their homes, "*Concertino* is a unique acoustic system [evidently "electronic architecture" signal processing developed by Steve Haas (SH! Acoustics; Stratford, CT; www.shacoustics.com; 866.277.9700)] that enhances the sound of live performances in residential spaces. …*Concertino* is barely visible and can be installed in any room." The system gives the room the acoustics of venues like "Carnegie Hall, the Royal Albert Hall, or Birdland." This system has been reviewed by Brent Butterworth in the September 2003 *Home Entertainment & Design*, and by Jeremy J. Glowacki in the May 2003 *Residential Systems*. To me this concept appears similar to Lexicon's Acoustic Reinforcement and Enhancement System (LARES) but for smaller rooms.

Flat Wire: DeCorp Americas (www.decorp.com) offers flat wire up to 14 gauge at no more than 10 mils thick for electrical and speaker connections, plus a product they claim can carry CATV signals. The obvious benefit is that the wire can be run under carpet and adhered to walls and ceilings with little work needed to make it practically invisible.

Problem, Solution: As always at CES, I felt quite well cared for by CEDIA staff. The press room offered all the resources and support anyone could hope for, and then some. However, there was one weakness: the unavailability of small powered carts (for use or rental) to assist those in need to get around the large exhibit halls and session areas. In my attempt to help a friend who has trouble walking, I tracked down the source of the 30+ small carts that were already on site for setup and breakdown, and negotiated the ability to rent one at reasonable cost directly from the local distributor for the three days my friend would need it. I even obtained tentative approval from a member of CEDIA management. However, when it got to the last step, someone higher denied permission. After I got home, I sent email to CEDIA Executive Director Billilynne Keller, pointing out that such arrangements have been available at CES and at AES conventions and providing a source for small carts. Subsequently CEDIA Director of Trade Shows Patty Voss informed me that at next year's Expo, carts will be available at reasonable cost.

November 2002 Meeting

Headphone Test Clinic

by Alvin M. Foster (with Jim Doucas and David B. Hadaway)

Why Headphones?

I am a devotee of using headphones as an aid to equalizing loudspeakers, because they provide me (1) a listening environment with reduced room interference, (2) no head-related transfer function (HRTF) problems to listen "through", (3) avoidance of the pitfalls introduced by multiple loudspeakers' mutual coupling, (4) excellent transient response, and (5) a phantom image that is free of the frequency response anomalies introduced by listening to a loudspeaker off-axis. Furthermore, headphones often sound better to me than speakers because they can produce sound with lower distortion and higher levels than most hifi loudspeakers, and finally they are free from the low-frequency variations introduced by the listening room.

Years ago, I incorporated headphones as another reference in my loudspeaker-EQ procedure because they often are smoother and flatter-sounding than loudspeakers. Unfortunately, headphones introduce a false reality; the experience is not the same as listening to loudspeakers in a room.

With training, however, the benefits of the headphone experience can contribute to a better loudspeaker-EQ approach, in my view, which when derived from the headphone experience is devoid of the negative influences of the room. Using headphones can therefore produce subjectively flatter sound.

Headphones have problems: 1) they remove the benefits of the listening room, 2) the sonic images seem to occur inside the head, 3) the sound lacks depth, 4) they bypass and distort some of our ears' built-in mechanisms for hearing in space, and 5) headphone EQ is not standardized; manufacturers determine how each model will sound. This personal voicing results in part from the lack of really accurate headphone simulation dummies or artificial heads (see Henrik Moller, Dorte Hammershoi, Clemen Boje Jensen, and Michael Friis Sorensen, "Evaluation of Artificial Heads in Listening Tests," in the March 1999 *AES Journal*). Without industry agreement on standards, manufacturers too often EQ for what they think will sell best. Some headphone companies consistently achieve fairly good results: Koss, Beyer, Sennheiser, Stax, AKG, Sony, among them.

A Headphone Clinic

The BAS is known for high-quality test clinics. During our 30-plus years, we have set standards at our clinics for testing amplifiers, preamplifiers, CD players, recording tape, and more. Some have been adopted industrywide.

On 23 March 2002 we held a headphone clinic, during which I made most of the measurements while David Hadaway determined sensitivity and impedance.

Our literature search had revealed that headphone testing is almost a black art. There are numerous standards but wide disagreement on how and what to measure. Therefore, we developed a set of traditional and nontraditional tests that we felt would adequately characterize each headphone's performance and correlate with subjective impressions.

Our tests included: (1) frequency response, (2) a scope trace of a 400Hz squarewave, (3) THD at 100dBspl, (4) IMD at 100dBspl, (5) spl at about 10% distortion level, (6) sensitivity, (7) impedance, and (8) subjective audio quality. The complete set of tests on each headphone took about 20 minutes.

Objective: A Flat Diffuse-Field Frequency Response

Gunther Theile ("On the Standardization of the Frequency Response of High-Quality Studio Headphones," December 1986 *AES Journal*, v34n12) concluded that headphone manufacturers were aware of the tone color defects of headphones that complied with equalization according to the then current standard: DIN 45 619 part 1, or IEC Publication 268-7(1981).

Theile and the authors of the artificial-heads evaluation study concluded that using a human was best, except for the problem of finding subjects who would permit a probe to be placed in their ear canal. Both studies concluded that the secondmost-reliable simulation was obtained through use of an accurate model of the individual's ear system being tested. Exact copies obtained by using the ear patterns of others were less reliable, but better than all the standard dummy heads then available irrespective of cost.

Theile suggested a new standard for high-quality headphones: It should require a flat diffuse-field transfer function and an ear probe measurement on test subjects. Theile described placing the microphone inside the human ear canal, just beyond the external ear.

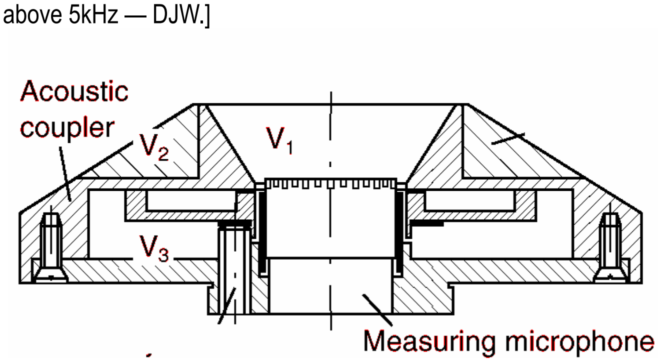
Thus we felt that a successful frequency-response result required a flat diffuse-field transfer function. However, we substituted a test jig for the human ear canal.

The Test Jig

For my first jig, I designed and built an 8"-wide U-shaped wood device with a ½" hole to hold the headphone and the ½" test microphone. However, Jim Doucas and I abandoned this jig for the B&K Telephone Test Head, because it held all the components (microphone and headphone) more securely and the results were more reliable. The B&K Test Head is metal and the headphone ear cups are held 8" apart, corresponding to the typical human head size after headphones are seated.

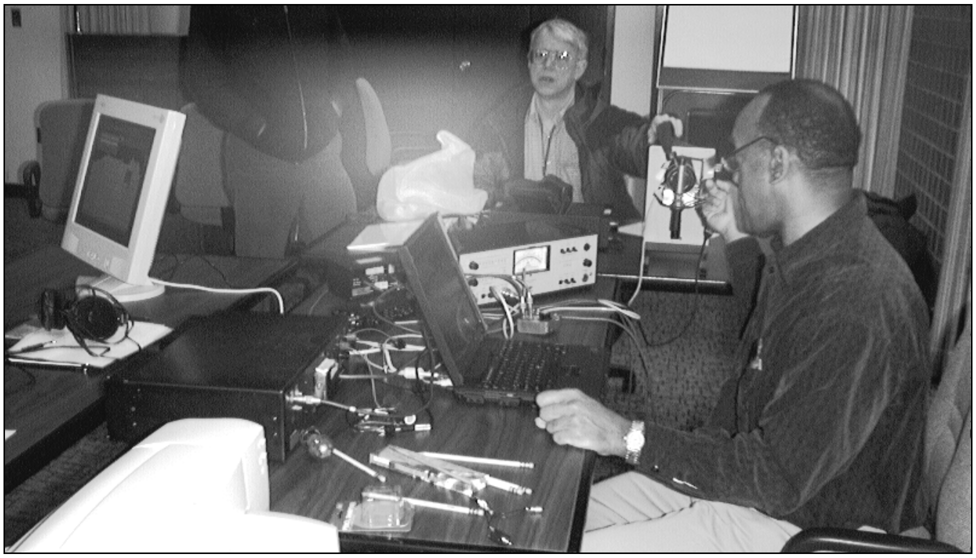
The complete measurement system included the B&K Type 4905 Telephone Test Head, a B&K pressure microphone (details discussed later), a spectrum analyzer and a voltmeter. Our early findings indicated good reliability and consistency, and our measurements correlated well with subjective determinations.

The tiny horn used to couple the microphone and the ear cup is about ¼" deep and 1" wide — small enough to prevent colorations caused by reflections, while providing a near-anechoic environment. None of the frequency-response anomalies exhibited in a particular headphone were repeated or duplicated in another ear cup. This hoped-for result indicated that a neutral, reflectionfree environment between microphone and ear cup had been created. [The flare of this horn might have caused colorations based on its dimensions and 'smoothness'. 20kHz has about a 2/3" wavelength; dimensions larger than this (meaning lower in frequency) might cause resonance effects between the fixture and the headphone transducer diaphragm, and transverse across the flare mouth, albeit probably at frequencies above 5kHz — DJW.]



The B&K test head held the microphone firmly and provided a solid platform on which to mount the ear cups. Test consistency required that the headphone ear cushion's diameter/size not exceed the test head's mounting plate, and that they were firmly seated, with no leaks. Once this was accomplished, a smooth frequency response was used to indicate the best alignment of the ear cup with the microphone. Moving the ear cup around resulted in very little response change, except for one oversized ear cup: the Koss ESP/950 required a thin cardboard cutout to accommodate its large foam-filled ear cushion (largest in the study: 5"x3.5"); however, only the bass was affected by the microphone plate being too small, which caused the ear cup and microphone to become unsealed. The high-frequency response was unchanged by the addition of the cardboard cutout.

The test head has an adjustable clamp that could apply pressure to hold the ear cup against the microphone assembly more tightly. By using the clamp, low frequencies could be improved on most headphones. It was decided not to employ the clamp because its use would likely not be duplicated in actual listening. The natural pressure of a unit's headband and the 8" spacing provided the requisite ear cup and microphone assembly tension. Only the Koss Pro/4AA headphones were extremely sensitive to the headphone pad's coupling pressure; the harder the ear cup was pressed against the B&K microphone plate the higher the output below 80Hz [I have found



Alvin Foster, measuring at the clinic, with J.K. Pollard (back)

this to be subjectively true with every non-Etymotic (an in-ear design) headphone I have used — DJW].

To make sure that all tests were fair and reliable, the first test performed on each headphone was frequency response with pink noise. The headphone's earpiece was moved around on the microphone mounting plate to get the flattest frequency response, thus optimizing the coupling.

Adjusting the spectrum analyzer to display 1/24th-octave frequency response did not accurately reflect what I [or anyone — DRM] hear through headphones with music; 1/3rd-octave measurement was more representative. The response test could be run in under three minutes with the Sound Technology SpectraLAB Analyzer. Test components: B&K 4134 microphone (recommended by the manufacturer especially for such close miking); B&K 2639 microphone preamplifier; B&K 2610 measuring amplifier; B&K 4905 Telephone Test Head; B&K 2426 electronic voltmeter; Sound Technology SpectraLAB FFT Spectral Analysis System v4.32 (includes the signal generator); the USP Pre 1.5 external sound card; an IBM 2628 ThinkPad laptop computer; and a Pioneer VSX-D509S multichannel receiver (whose headphone output impedance is 135 ohms; this model was used because its headphone output design is typical of consumer equipment).

[Such a relatively high output impedance, most likely a 125-ohm resistor in series with the output IC op amp, is likely to alter the frequency response of many headphones — AES.] [Southwick's point is well-taken, but it is notoriously difficult to design reasonable headphone amps (reasonable as to cost, size, and heat dissipation) with extremely low output impedance that can deliver sufficient power, and the Pioneer's impedance is not as high as some. Indeed, the well-regarded headphone preamps found in the Apt Holman, dbx CX1, and NAD preamps all had an output impedance around 100 ohms or a bit higher — DRM.]

Open or Closed

A closed headphone provides at least 12dB isolation from room noise. Only one side is open to the ear; the opposite side is enclosed and does not vent into the room. The closed headphone is typically circumaural (around the ear). It is popular with recording engineers because of their need to monitor the recording process onsite and be acoustically isolated from ambient sound.

The open-back headphone is typically lighter and offers only minimal room isolation. It often employs a foam pad that fits against the external ear while the ear cup's housing is perforated so the sound escapes out both sides.

The Tests

Frequency Response

We used pink noise to obtain a headphone's frequency response and set the playback level. The signal was fed to the left channel of the Pioneer's headphone jack [to save time, only the left ear cup was tested, as it was assumed that the headphone transducer design and manufacture are consistent; it would have been useful to verify this by repeating the test on the right earcup of at least some of the units — AES]. The microphone picked up the signal and the B&K measuring amplifier reported the spl throughout all the tests to the computer, which saved all test results in a format that preserved the running results of at least 10 seconds of each completed test. Each clinic participant was given a printout of his headphone's frequency response.

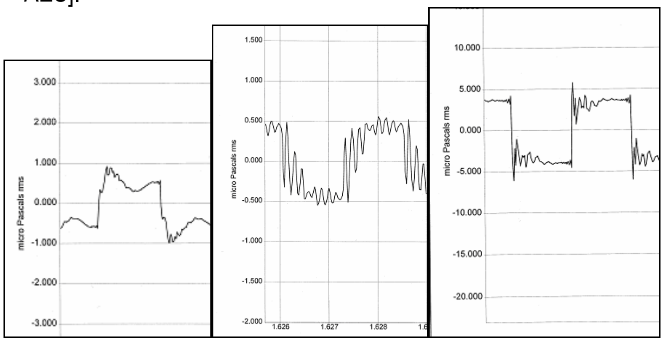
Attempts to use a 20-20,000Hz swept sinewave instead of noise were discontinued because the results were found to be less reliable.

Also because of time constraints, the graph for each model is a 'snapshot' of its frequency response with pink noise, selected as closely representative but not showing the pink-noise data continuously averaged.

Micha Schattner observed a pattern in many responses: he concluded that the 180-250Hz boost was introduced by headphone manufacturers to provide the listener with the loudspeaker experience. He pointed out that headphones are mechanically capable of being flat in the lower midrange. The Stax SRMT 1 and the SRX MK III do not have that hump, but Stax's newer 'improved' model does.

Squarewave

The 400Hz squarewave yielded data that are difficult to assign importance to because most headphones reproduced a reasonable reproduction of the signal. The Koss ESP/950 (first graph) is a good example of a typical result; however, AKG (figure 2) and Sennheiser (figure 3) units showed ringing [which could have been the result of interaction effects mentioned earlier, the comparatively high output impedance of the headphone amplifier, reactive behavior with some phones, or clipping problems due to headphone-amplifier limitations — AES].



Midrange squarewave: Koss electrostatic; AKG; Sennheiser

Several headphones exhibited a high-frequency peak. We did not post all the results because we are not sure how that misbehavior correlates with bad sound [or if it somehow again relates to a test anomaly — DJW].

Total Harmonic Distortion (THD)

A 1kHz sinewave was fed to the headphone. To obtain a THD value, the spectrum analyzer was used to compute the output harmonics and the power ratio between them and the input sinewave. The computer adds up the levels of all the harmonics (distortion products harmonically related to the tone). What was nontraditional about the BAS clinic test was the high playback level used for the measurement, 100dBspl (compared with the lower levels used by most manufacturers). [Clean 100dBspl does not seem to be asking a lot of a serious headphone design, though! — DRM] [But again, depending on the driving amplifier's output capability, THD could have been increased by its inability to provide adequate, that is undistorted, signal to the lower-impedance headphones, and indeed, some of the distortion data for the lower-impedance designs are higher — AES]

Intermodulation Distortion (IMD)

We settled on the traditional test: 60Hz plus 7kHz mixed 4:1 (the bass tone is 12dB higher than, or over twice as loud as, the treble tone).

The clinic was nontraditional in performing this test in the first place as well as in the level used. We chose a playback level for all the headphones of 100dBspl, and to divide the good from the bad further, we set an arbitrary cutoff point of 10% maximum distortion as we raised the input. When the 10%-distortion figure was reached, we recorded the playback level and discontinued the test [at such levels, amplifier output capabilities into very low-impedance or low-efficiency designs could also have worsened the results significantly — AES].

Attempts to use multi-tone test signals to separate out the units capable of low distortion and high playback levels were dropped because the results were too qualitative, in that we could not get a single number to represent performance.

Sensitivity & Impedance Measurement Station *(photo below)*
by David Hadaway

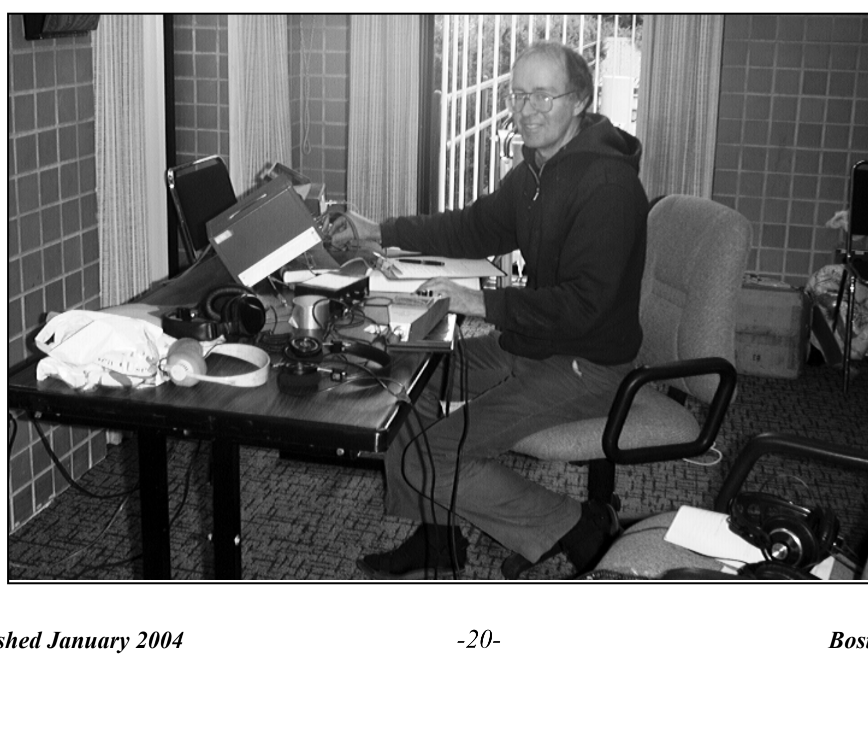
Sensitivity & Voltage for 100dBspl Playback

I elected to determine sensitivity with an open-air measurement to avoid any influence of a coupling device. I found that with the microphone of the Columbia Research Labs sound-level meter placed in the approximate location of the eardrum, the level gradually rose as the frequency increased, then steadied at a plateau in the 500-2000Hz region, then rose further to a resonance peak. Each headphone was frequency swept until the midrange plateau was reached; the average level there was recorded. My Columbia Research Labs sound level calibrator was sent to a metrology lab for calibration after the clinic and was found to be accurate (no adjustment required). I later compared my calibration with Foster's and found a 1.5dB discrepancy, which was judged not to be significant [and might have been due to the open vs closed environment — DJW].

The voltage required for 100dBspl was recorded for each headphone except for those with their own amplifier. The range was 0.36-9.6Vrms [an almost 30dB range of sensitivities — DJW].

Impedance Curve

Ideally, impedance should be measured by driving the headphones from a constant current source and measuring the voltage across them, using Ohm's law (Z = E/I) to calculate the impedance. In practice, it is sufficient to use a series resistor much larger than



David Hadaway measures sensitivity and impedance.

the impedance of any headphone, in our case a 50,000-ohm resistor. Thus:

I = E(fixed)/50,000 (neglecting headphone impedance, which is insignificant in this setup)

Therefore Z = 50,000E/E(fixed)

E was fixed at 5Vrms so the impedance in ohms was 10,000 x the voltage reading. Thus a 10mV reading corresponded to 100 ohms.

I used a Krohn-Hite 5800A continuously variable function generator with 20Hz-20kHz on a single variable dial and had verified that the combination of the Krohn-Hite and a Ballantine 303 voltmeter was flat to the thickness of a line on the meter face (within 0.02dB).

A typical impedance curve showed a dip below 50Hz, a resonant peak at a low frequency in the 70-200Hz range, a shallow depression in the 500-2000Hz region, and a rising response above 2kHz. Unlike loudspeakers, where the impedance curve has little correlation with the frequency response (since generally they are driven from voltage sources with an output impedance that is comparativeely very small), headphone amplifiers generally have an output impedance of 120 ohms or so, which is the same order of magnitude as that of most headphones [resulting in approximately half the headphone amp's power lost — DJW]. Why this is so is not clear — perhaps to minimize sensitivity variations among headphones or protect against short circuits? The result is that a headphone's frequency response can be readily affected by its impedance curve, sometimes strongly so. Headphones with the highest impedance are less affected as well as ones with the most constant impedance curve. [It seems to me that with a 120-ohm amplifier output impedance, a headphone would have to have either uniform impedance across the audio band, or an impedance at all audio frequencies over 1200 ohms or under 12 ohms, in order for it to be immune to the problem. Probably not many hifi headphones qualify — DJW.]

One of the worst offenders was my Radio Shack PRO 90 (which varies from 330ohms@20Hz to 600ohms@20kHz. With a typical headphone amp they would sound brighter than with a loiw- or zero-impedance source, i.e., a voltage source). In contrast, the Sony 7506 varied only from 73ohms@20Hz and 90 ohms@51Hz to 85.5ohms @20kHz (it would sound essentially the same with most headphone amps).

Listening Test

by Alvin Foster with J.K. Pollard

For the listening test, a station was set up to hold and connect the headphones supplied by members. Each member was asked to compare and rate all units on-hand, using either an available CD or their favorite CD. Multiple headphones were simultaneously fed by the output of an amplifier designed and built by Hadaway. It had a volume control for each headphone. Up to six headphones could be connected at one time. Its output impedance, too, is 120 ohms.

The subjective results have a built-in bias [apart from not being blind — DRM]: Headphones remaining on the table the longest were cited as a likely favorite. For example, the Sennheiser HD 600 and the HD 280 Pro, plus the Koss ESP/950, belonged to people administering the tests. As a result, they were available all day to listen to. Most headphones remained on the listening table only while their owner was present.

Another weakness of the listening test design is revealed by the fact that some owners preferred their headphone over the better ones available. On two occasions owners singly voted their unit best [the definition of subjective evaluation — DJW].

How Loud?

I have owned a Koss electrostatic headphone system (which includes its own amplifier, to be fed from a source's headphone output) for many years. Early on, I realized that they provided generally sufficient volume but were not capable of playing loud before gross distortion set in. This clinic's IMD test revealed that the Koss headphone system would go into clipping if any peak exceeded 105dBspl. Dynamic headphones, even those not capable of playing as loud cleanly, sounded louder. The distortion of the Koss below 105dBspl was among the best; however, when the clipping level was reached the sound was instantly and severely degraded. They cannot be recommended for onsite monitoring, where playback levels must exceed the noise of the room that leaks through the ear seal. In my experience, the average playback level of the Koss ESP/950 must not exceed 88dBspl or gross 'clipping' distortion will result.

Results

Given the number of units tested, the results in each category were reduced to the best three, to make the data easier to digest.

The listening station

Best / Worst in Selected Tests

THD Best

Sony MDR-V6	0.02%
Sennheiser HD 600	0.02%
Beyer DT 990	0.02%

THD Worst

Koss QZ/2000 Noise-Canceling	9.5%

[It's hard to believe that such distortion would be tolerable in headphones at normal listening levels; one wonders whether the noise-canceling circuitry played a part — AES.]

IMD (100 dBspl) Best

Stax SRMT1	0.04%
Stax SRX MX3	0.1%
Koss Pro/4AA Original	0.1%

IMD (100 dBspl) Worst

Kenwood portable DPC391	7.2%
Koss Pro/4AA refurbished	4.7%

[extremely curious that the same model would exhibit such drastically different behaviors; apart from a testing problem, one suspects a faulty unit or some drastic redesign by Koss for the refurbished unit — AES]

Sennheiser HD 580	4.7%

Lowest IMD at highest spl, Best

Koss Pro/4AA Original	0.7%@113.8dBspl
Sony MDR-CD850	7.4%@114.4dBspl
Beyer DT 990	6.7%@113.6dBspl

Highest IMD at lowest spl, Worst

Koss QZ/2000 Noise-Canceling	12.9%@86.0dBspl

Most Expensive Srp

Stax SR-Lambda Signature	$1900
Stax SRMT1	$1800
Koss ESL/950	$1000

Most Sensitive (100dBspl)

Radio Shack Pro-135 Optimus	0.43V
Sennheiser HD 280 Pro	0.47V
Sony MDR-CD850	0.55V

Least Sensitive

AKG K-240DF	9.6V
Koss Pro/4AA Original	6.3V
Beyer DT 990	4V

Impedance Highest (manufacturer data)

AKG K-240DF	600ohms
Beyer DT 990	600ohms

Impedance Lowest (manufacturer data)

Grado Prestige SR 60	32ohms
Grado Prestige SR 80	32ohms
Sennheiser 433	32ohms

Other Criteria

Open/Closed Back

20 units of 16 different models	Open
11 units of 9 different models	Closed

Subjective Winner

Koss ESP/950 Electrostatic	12 points
Sennheiser HD 600	15 points

Honorable Mention (selected by owners)

AKG K-240DF
Sony 7506

Notes

—The Koss QZ-2000 NR headset obviously had its NR circuitry switched on.

—Many IMD readings are below the "10%" reading for assigning dBspl because the amplifier used had a detented volume control, which caused undesirably large steps. The control was adjusted to the highest level available such that the IMD was no higher than 10%.

—Two model AKG-K-240DFs exhibit an spl @10% IMD differing by 12dB. A similar problem occurs with the Stax SR-Lambda Signatures. [The two AKG models report dramatically different spls; the net result is a distortion difference. The second AKG (lower at 98dBspl @6.2% IMD) was used to show how it performed at a lower spl, not at 10% IMD. The Stax is a classic example of an amplifier overdriving electrostatic headphones. If the maximum level is exceeded by as little as 4.2dB, the unit exhibits excessive distortion. This is the reason why I am never able to play my Koss electrostatic headphones near their maximum level, well below hearing damage, without irritation — AMF.]

—Sennheisers with different impedances exhibit similar spl at the 10% IMD test but appear to have inverse distortion readings at 100dBspl. This too is suggestive of some form of amplifier issue.

—It clearly would be useful to measure the frequency response using pink noise and, this time, the SoundTechnology SpectraLAB's 'forever' averaging of each of several headphones driven by a power amp with an exceptionally low output impedance; then repeating the process with a 20-ohm resistor, a 125-ohm resistor, and a 1000-ohm resistor in series with the headphone. This would plainly show any effect of output impedance on headphone response. If you make a cursory comparison of the frequency response of four models tested in this clinic with their FR curves displayed at www.headphone.com, a fascinating site worth the attention of all headphone-lovers, parts of the curves are similar, and other parts differ dramatically.

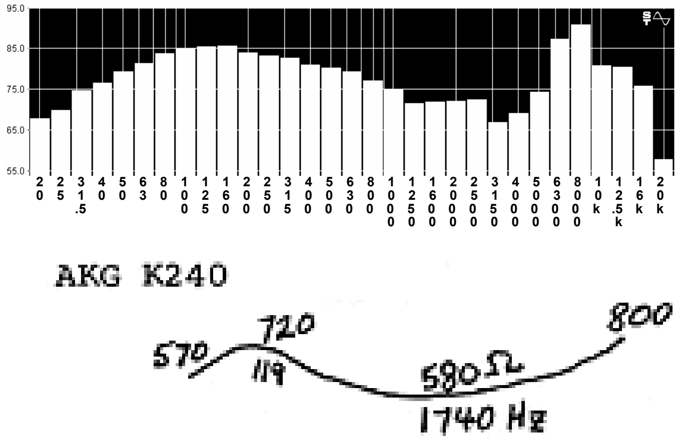
—Load issues aside, this clinic's data do show the variability in headphones that should relate to listener perception and preference. One also can see the wide range of sensitivity among models and thus how hard it is to make fair comparisons (since louder typically sounds better). And as always, a final important result of the clinic lies in showing how hard it is to set up and run tests of this complexity. Foster, Doucas, and Hadaway deserve our great thanks for taking on such tasks — DJW and DRM.

Results

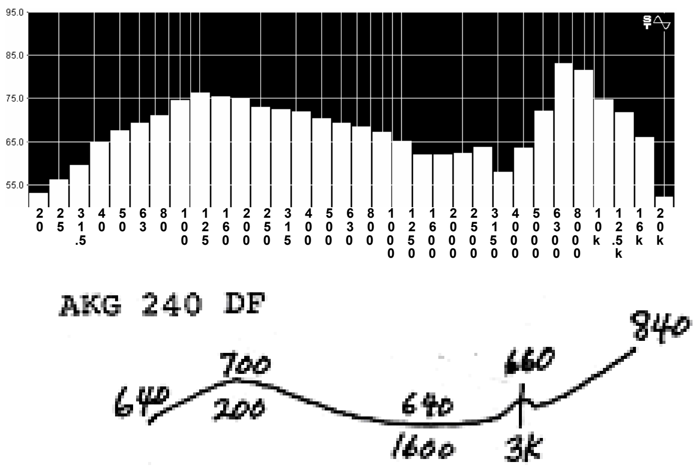
Headphone	Srp ($)	Open/ Closed	%THD @1kHz / 100dBspl	%IMD @100dBspl (±2 dB)	dBspl near 10%IMD (±1%)	Volts for 100dBspl	Impedance (mfr / BAS)	First, Second, Third
AKG K-240DF	150	O	0.10	3.4	110.1dBspl@11.4%		600/580	1
AKG K-240DF	150	O	0.10		97.9dBspl@6.2%	9.6	600/640	
Beyer DT 131	60	O	0.05	1.6	112.4dBspl@6.8%	1.12	40/40	
Beyer DT 990	200	O	0.02	1.4	113.6dBspl@6.7%	4.0	600/550	
Bose QC1 Noise-Cancel	300	C	0.20	1.2	106.5dBspl@1.3%	2.2	/	
Grado Prestige SR 60	70	O	0.20	3.3	105.5dBspl@10.1%	0.66	32/33	
Grado Prestige SR 80	95	O	0.12		98.4dBspl@7.0%	1.06	32/34	3
Kenwood Portable DPC-391		O	0.10	7.2	103.1dBspl@9.7%		16/	
Koss ESP/950 Electrostatic	1000	O	0.04	0.4	105.9dBspl@10.3%		/>1k-ohm*	2,1,2,1,3,2,1
Koss Pro/4AA (Original)	100	C	0.03	0.1	113.8dBspl@0.7%	6.3	230/86	
Koss Pro/4AA (Refurbished)	100	C	0.05	4.7	107.5dBspl@9.8%		230/	
Koss QZ/2000 Noise-Cancel	200	C	9.50		86.0dBspl@12.9%		5000	
RadioShack Pro25 Optimus	40	O	0.04		100.0dBspl@5.2%	0.78	/105	
RadioShack Pro135 Optimus	65	C	0.05		99.2dBspl@8.3%	0.43		
Sennheiser HD 280 Pro	280	C	0.07		106.3dBspl@10.6%	0.47	64/82	
Sennheiser HD 580	350	O	0.08	4.7	105.5dBspl@10.0%	1.95	300/315	S
Sennheiser HD 433	40	O	0.12	2.2	107.8dBspl@5.1%	1.3	32/35	
Sennheiser HD 600	450	O	0.02	3.8	107.6dBspl@9.8%	3.0	300/310	3,2,3,2,1,1,3
Sennheiser HD 600 (AMF)	450	O		4.4			300/	
Sony MDR-V6	100	C	0.02		99.3dBspl@6.4%	0.68	63/68	S
Sony MDR-V6	100	C	0.10		98.3dBspl@5.7%	0.57	63/72	
Sony MDR-V6	100	C	0.10	2.9	109.0dBspl@10.3%		63/	
Sony MDR-7506	180	C	0.08	2.2	112.5dBspl@10.0%		24/75	1
Sony MDR-CD850	200	C	0.07	1.5	114.4dBspl@7.4%	0.55	/	
Stax Lambda Nova Classic	1300	O	0.03	0.3	108.3dBspl@9.8%		/	
Stax SR-Lambda Original		O	0.03	0.5	104.3dBspl@3.4%		/	
Stax SR-Lambda Pro	750	O	0.05	0.5	105.9dBspl@0.9%		/	
Stax SR-Lambda Signature	1900	O	0.07	0.2	105.6dBspl@0.3%		/	
Stax SR-Lambda Signature	1900	O	0.03	0.2	109.8dBspl@8.3%		/	
Stax SRX MK III		O	0.05	0.1	104.0dBspl@0.2%		/	
Stax SRMT1	1800	O	0.04	0.04	105.9dBspl@1.9%		/	

* Koss headphone amp input impedance measured by Foster

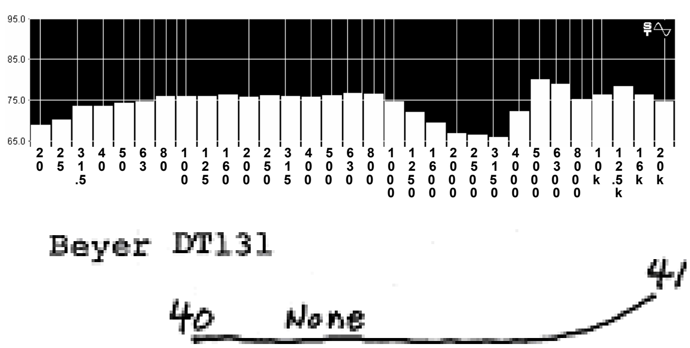
Frequency Response (top graph) and Impedance (bottom curve, with endpoints 20Hz & 20kHz and Z above curve, f below)



AKG K-240DF (Sean Mangan)



AKG K-240DF (John S. Allen)



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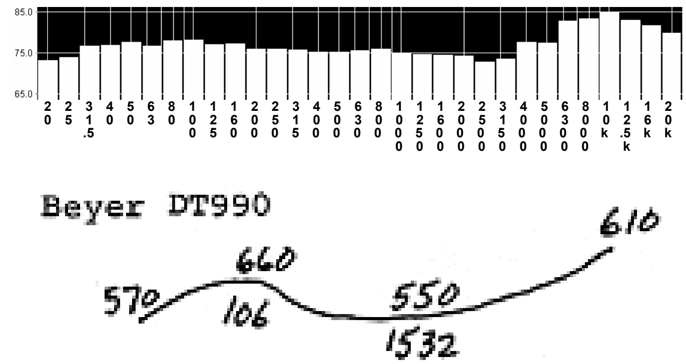

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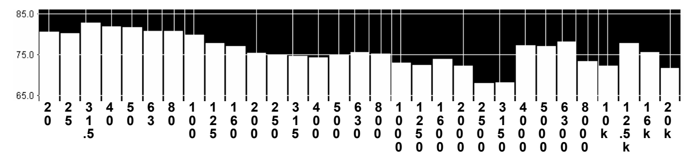
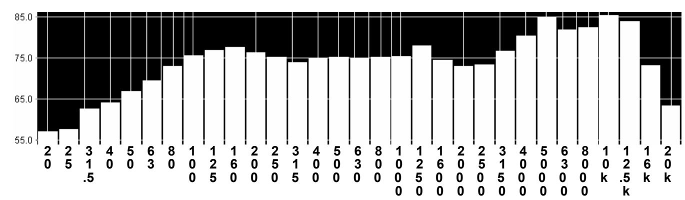


Bose QC1 Noise-Canceling



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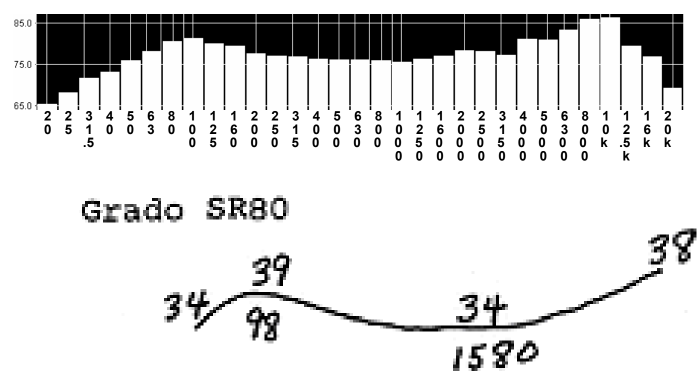
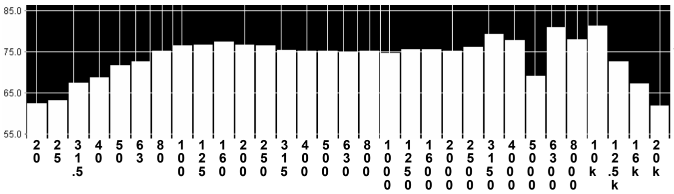
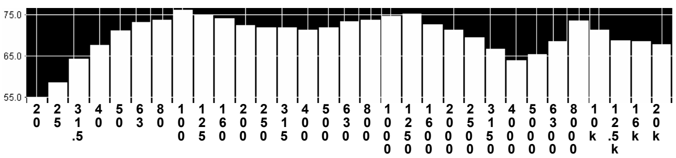


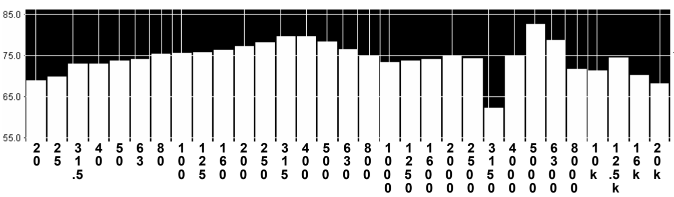
Grado Prestige SR 80

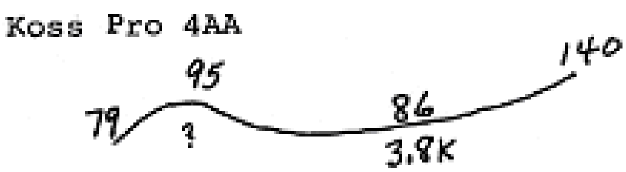


Kenwood Portable DPC-391

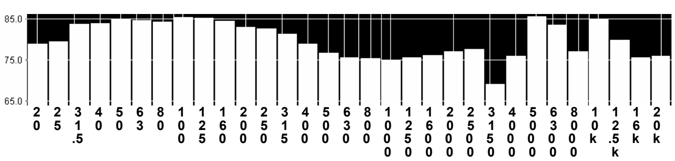


Koss ESP-950 Electrostatic (Alvin M. Foster)

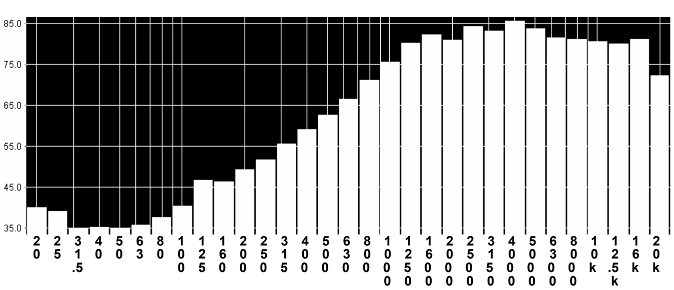




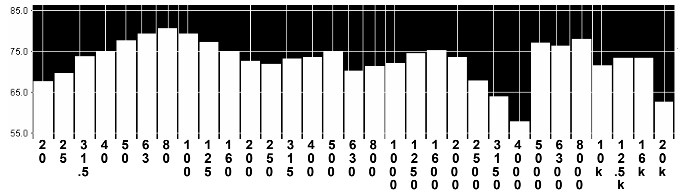
Koss Pro-4AA (original) (Alan E. Southwick)



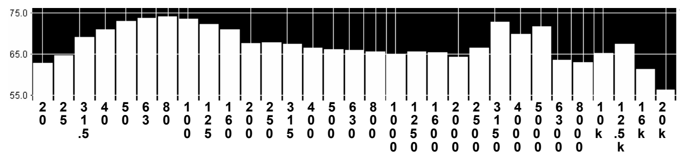
Koss Pro-4AA (refurbished)

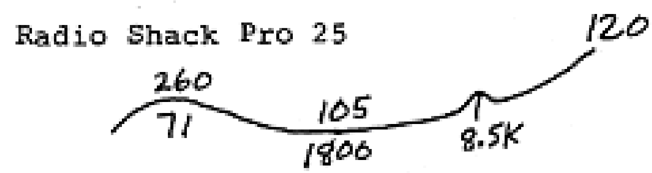


Koss QZ-2000 Noise-Cancel



Radio Shack Pro-135 Optimus





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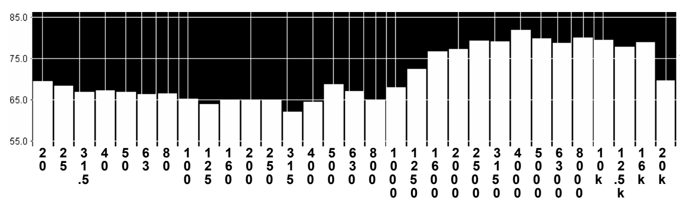
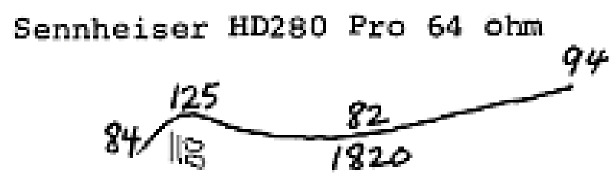
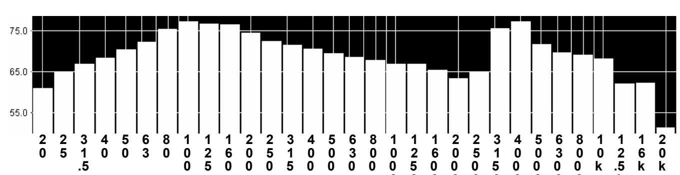
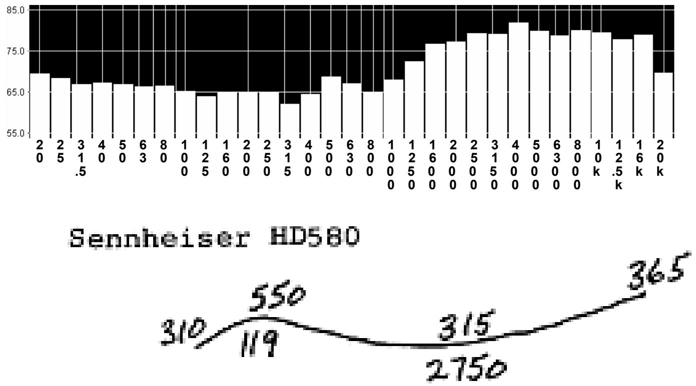




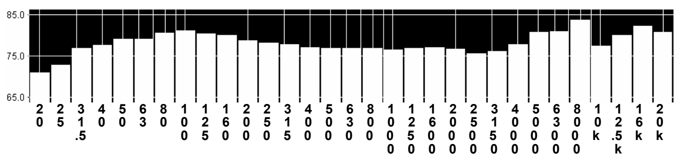
Sennheiser HD-280 Pro



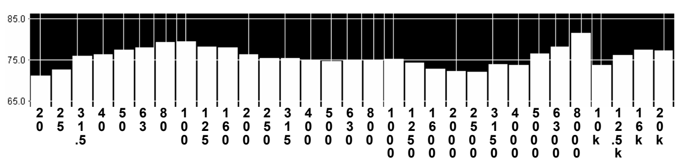
Sennheiser HD-433



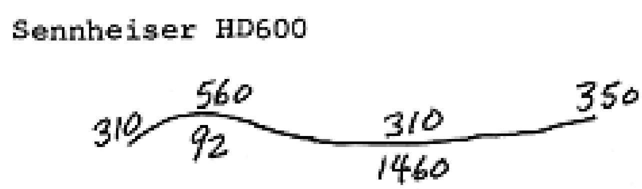
Sennheiser HD-580



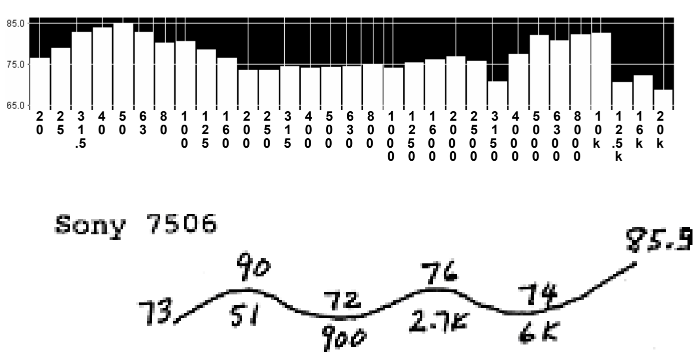
Sennheiser HD-600 (Alvin M. Foster)



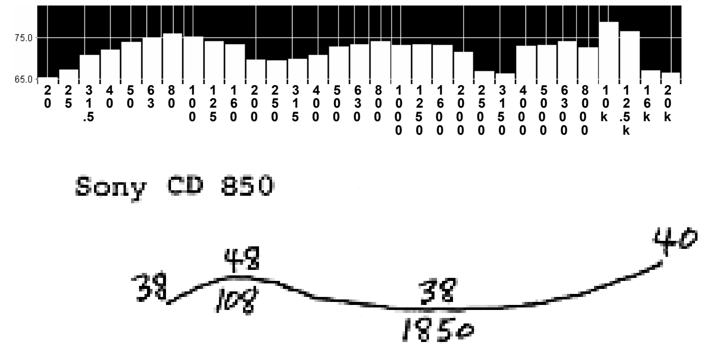
Sennheiser HD-600 (T. Brown)



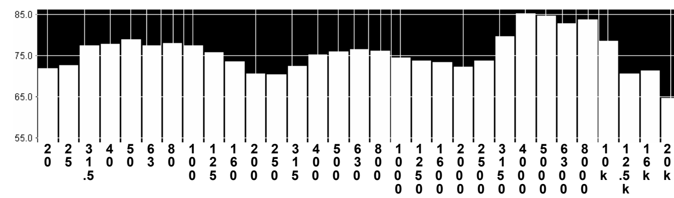
Sennheiser HD-600



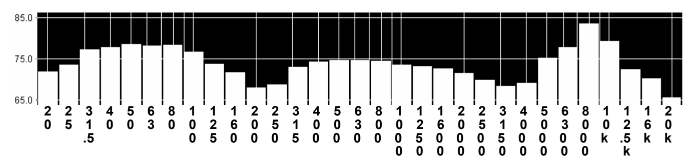
Sony MDR-7506



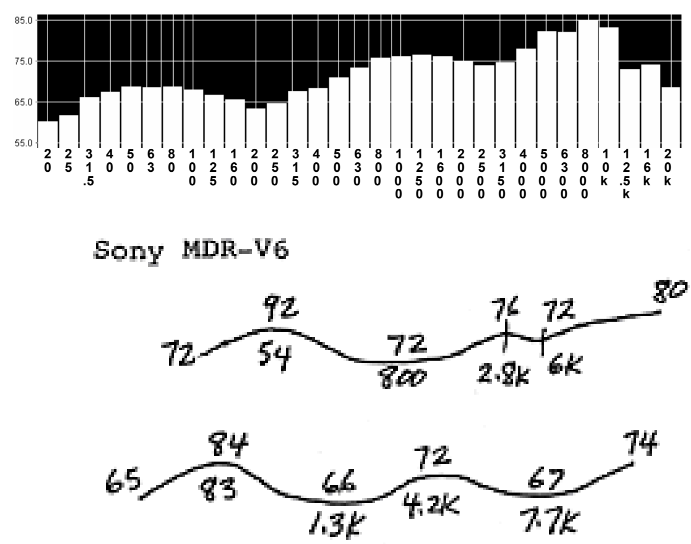
Sony MDR-CD850



Sony MDR-V6 (Alan E. Southwick)



Sony MDR-V6 (Stephen Owades)



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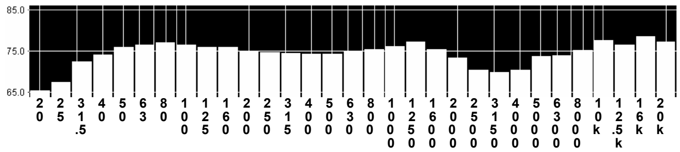
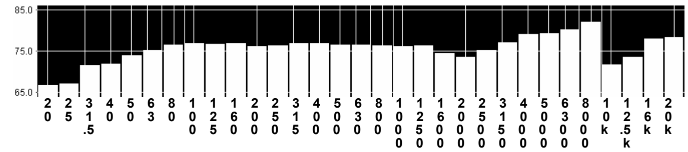
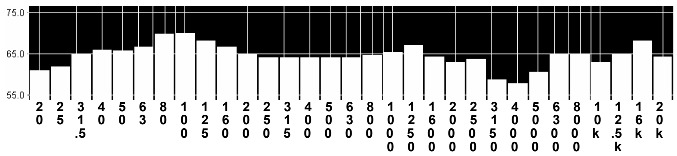


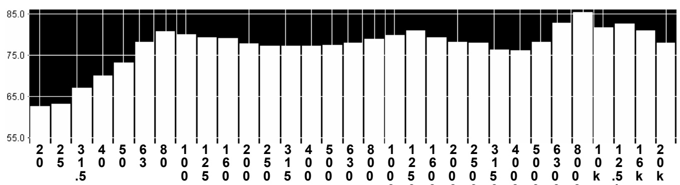
Stax Lambda Nova Classic Tube



Stax SRX Mk III



Stax SR-Lambda Original (Micha Schattner)



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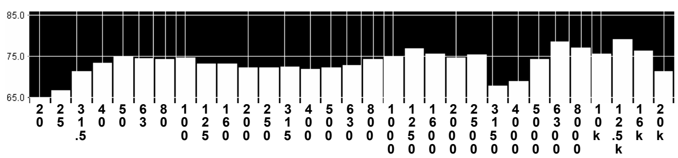
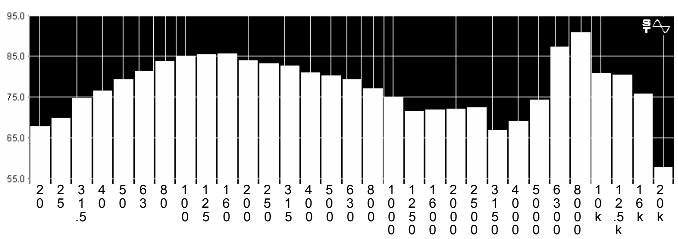
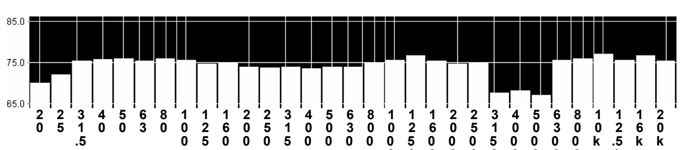


Stax SR-Lambda Signature (Bill Wolk)



Stax SR-Lambda Signature (J.K. Pollard)



Stax SRMT 1

March 2003 Meeting

by David B. Hadaway

Longtime BAS member Alan E. Southwick [the coincidence of his initials being AES is not lost, just typically ignored — DJW] has been making professional location audio and video recordings since the late 1960s. Building a house in Rhode Island offered him an opportunity to create a small dream studio complex in a portion of the basement, along with storage space. Longtime friend David Weinberg from Maryland was closely involved in the design and finishing processes.

DBH: Didn't you two go to school together?

AES: “Yes, we met in 1964 when I arrived at Worcester Polytechnic Institute from my home, in Lowell, Mass., with a Scott/Fisher/KLH stereo system and a Heathkit portable FM radio. I believe his remark that he had built the same radio was his greeting when we met and he helped me move into the dormitory. Later we learned that we were both born in Baltimore and that our mothers had the same obstetrician!” [Unclear whether these exchanges were accompanied by barking-seal guffaws à la *Revenge of the Nerds* — DRM.]

“My whole life has been involved with audio. My physician father was a competent cellist and classical organist; he owned several Hammond electric organs, ultimately replaced with a Rogers — in the living room. [Hearing his father play the organ as a Sunday morning call to a great pancake breakfast was very special — DJW.] My mother, a concert pianist who studied at the New England Conservatory, was a winner of the Beebe competition; the prize included a grant for a year’s study in Europe, performance of the Lizst Piano Concerto no. 1 with the Boston Symphony under Serge Koussevitzky, and a Mason-Hamlin Model A baby grand [now in Southwick’s living room — DJW]. Music was a critical part of their relationship: they originally met at a BSO-sponsored summer music camp in Maine, prior to the founding of the Tanglewood Music Center.

“When I was an infant, my mother would put me in a crib under the piano so she could keep an eye on me while she practiced. To this day I still don't fully enjoy listening to a piano unless I am under it!”

Q: Don't they object at concerts?

The fact that AES’s wife is a realtor helped with the location and design of their house, and she has been very understanding of his audiophilia.

Putting the studio complex in the basement was logical since the first concern was acoustic isolation from the rest of the house and the neighbors. Sound transmission is restricted by mass, as in poured-concrete walls and floor.

The next issues were to size the studio and the production room for good distribution of room resonances and to treat it for other acoustical parameters.

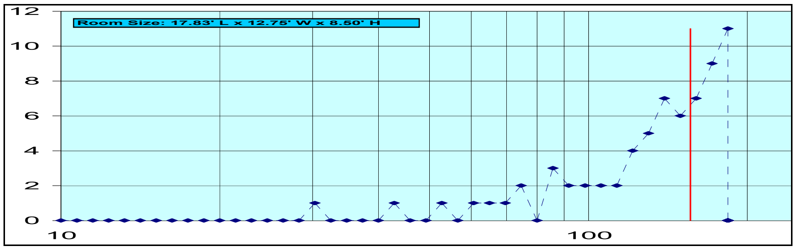
Reviewing *Acoustic Techniques for Home & Studio* by F. Alton Everest led to investigating the criteria developed by Richard H. Bolt (of Bolt, Beranek and Newman) in 1944 (with Philip M. Morse; "Sound Waves in Rooms," *Reviews in Modern Physics*, v16n2, April 1944, pp69-150), polished by L.W. Sepmeyer in 1965 and definitively refined by M.M. Louden ("Dimension-Ratios of Rectangular Rooms with Good Distribution of Eigentones," *Acustica*, v24, 1971, pp101-4). Their findings were that certain height, width, and length dimensional ratios lead to a more even distribution of bass resonances throughout a domestic-sized rectangular space. Based on Louden's conclusions, optimal domestic-room dimensions should have the ratio of 1:1.4:1.9 for smoothest distribution. [See BASS v17n6 p5 for a shorthand formula — DRM.]

Because most poured-concrete foundation forms come in 4'-height increments, the foundation wall is 8'h. Four 2"x10" planks were laminated together in a knee-wall atop the poured foundation wall. Thus the available finished ceiling height was 8.5 feet. Because of this and the space constraints within the house plans, the studio room ended up at 8.5'x12.75'x17.83' (1:1.5:2.1, which, while not fully optimal, is Louden's third listing and near the geometric center of Bolt's kidney-shaped area of acceptable dimensional ratios. The smaller production room width was set by the studio size, which limited the remaining foundation area available) so the dimensions ended up at 6.8'x8.8'x12.9' (1:1.3:1.9, corresponding with Louden's 2nd listing, higher on Louden's preference list than the studio) The catch is that the production room is small and very low, so relatively even distribution of low-frequency room resonances over the bottom octaves (20-80Hz) is harder to obtain than in the larger room, resulting in somewhat rougher bass room response. Playback is reinforced (boosted) at each resonance as well as over some bandwidth (Q) around its frequency. The smaller the spacing between adjacent resonances (the closer they are together), the more these bands of reinforcement overlap and potentially smooth out the in-room frequency response. As shown in graphs 1 and 3, the lowest resonance in the smaller room is at a higher frequency (44Hz vs 30Hz) than in the larger room, in the bottom two octaves the smaller room has fewer resonances (four vs eight), and their spacing is wider. (These are computed, not measured.) Thus it is harder to get smooth bass response in the smaller room than in the larger room.

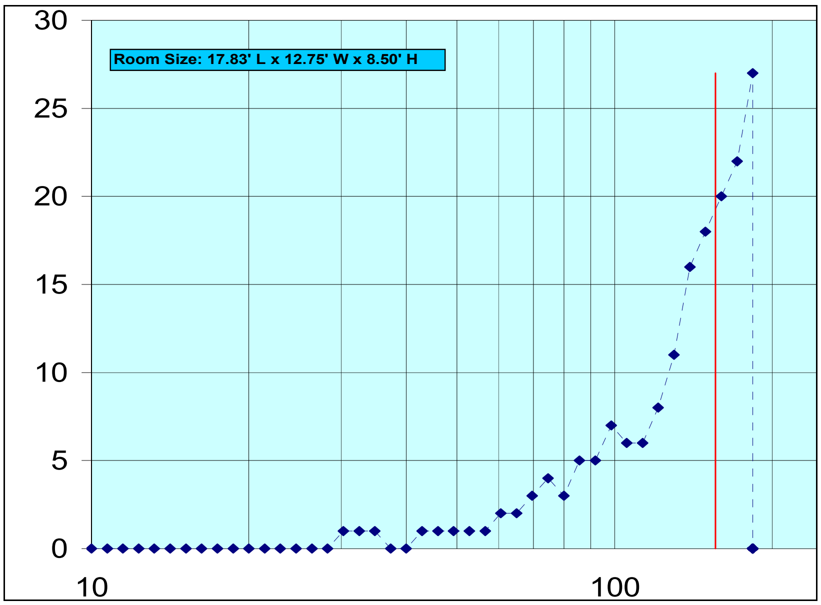
Graphs 1 (studio) and 3 (production room; all from David Weinberg) show the theoretically computed number of dimensional resonances vs frequency. Ideally, resonances should be uniformly distributed; these rooms are pretty good. (The frequency above which this parameter becomes audibly inconsequential is called the Schroeder frequency, f_s., which here is around 150Hz in the larger room and around 250Hz, a shade below middle c, in the smaller; it is marked in the four graphs by the extra vertical rule >100Hz that is not a grid line.)

Graphs 2 (studio) and 4 (production room) are more bundled, meaning they show resonance cluster monotonicity. This is based on the Bonello criterion that for best bass acoustics the number of dimension-based resonances in each succeeding third-octave should increase monotonically, never going down (Oscar Juan Bonello, "A New Criterion for the Distribution of Normal Room Modes," September 1981 *Audio Engineering Society Journal*). These graphs show the clustering in tenth-octave steps of a sliding third-octave band.

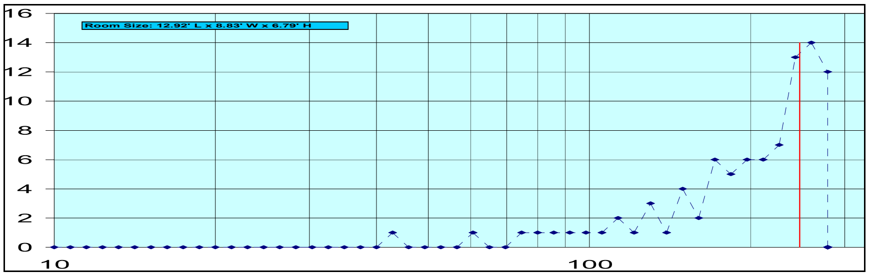
[The spreadsheet requires entering only the three room dimensions; the charts and the Schroeder frequency are automatically computed. It is available from me with a Word readme — DJW (WeinbergDa@cs.com)] [Also see http://www.rpginc.com/products/roomsizer/rs_compareo.htm to get some sense of the range of subtleties involved in computations like these in real rooms — DRM.]



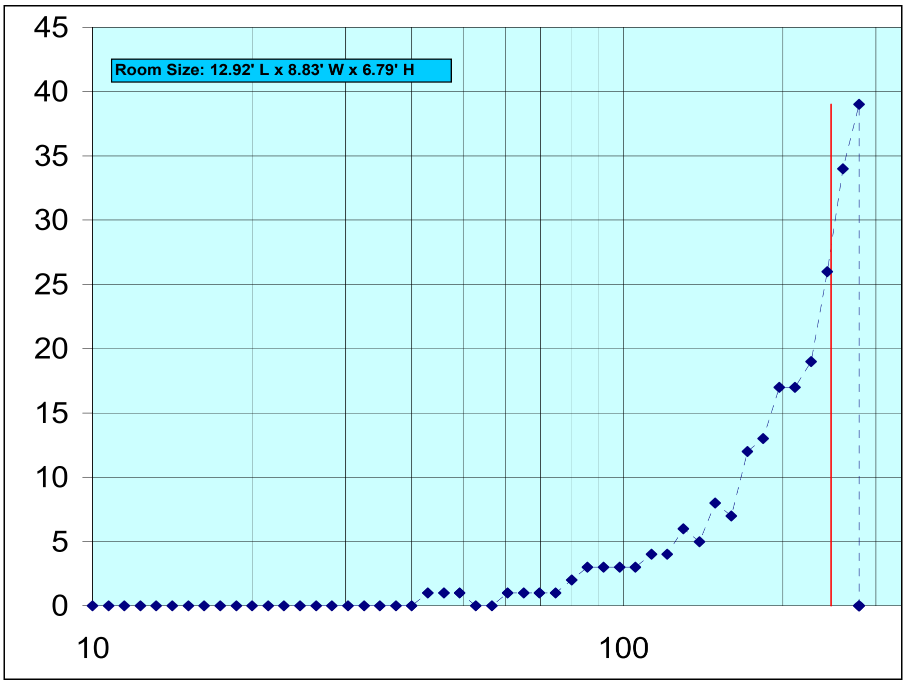
Graph 1: Larger room's resonances, no. vs f (computed; tenth-octave bands)



Graph 2: Larger room's resonances, clustered no. vs f (computed; third-octave bands)



Graph 3: Smaller room's resonances, no. vs f (computed; tenth-octave bands)



Graph 4: Smaller room's resonances, clustered no. vs f (computed; third-octave bands)

The room layout (Figure 1, below) shows the four areas, studio, control/production, and two storage spaces.

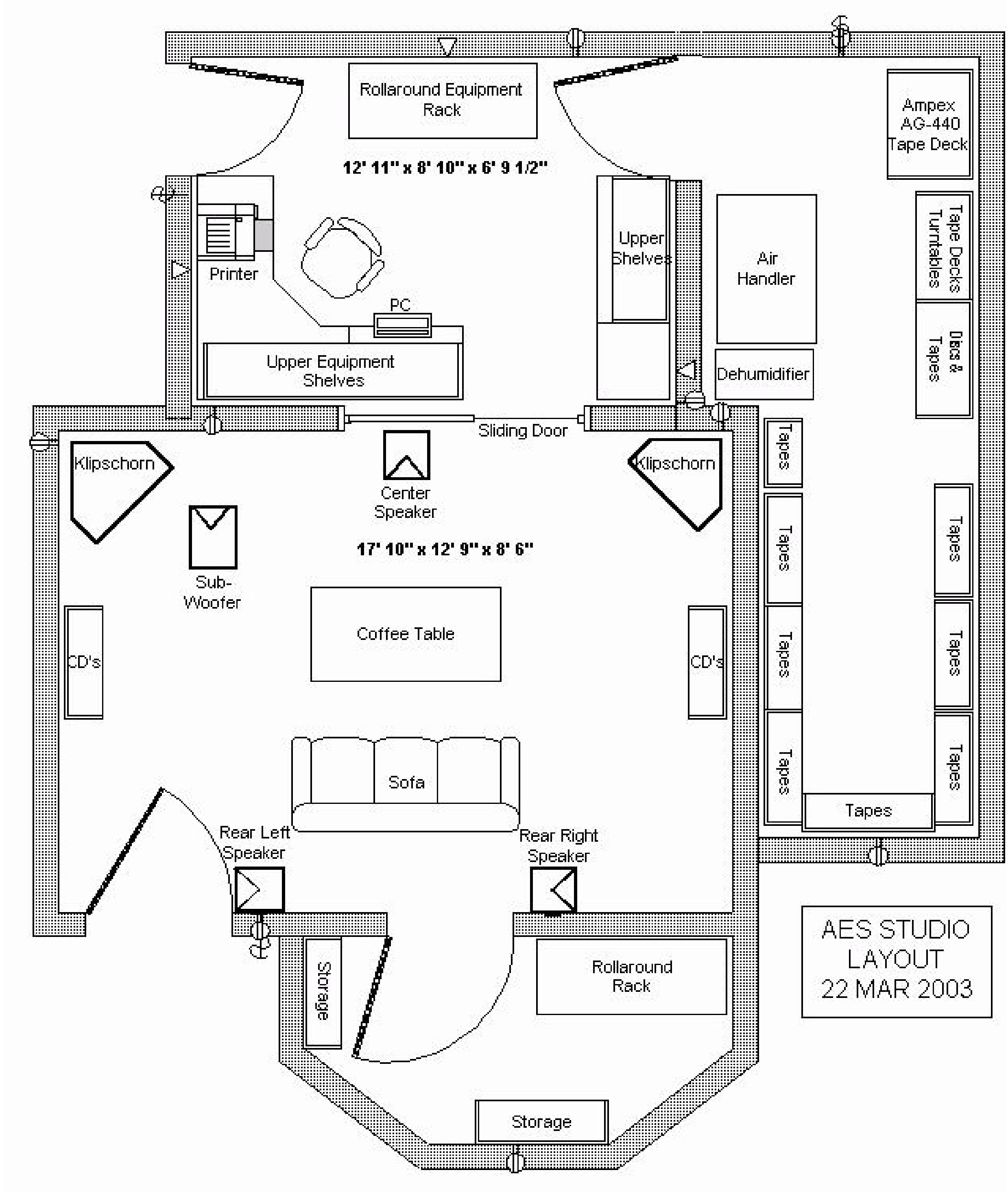


Figure 1

Three of the studio walls are sandfilled cinderblock, the fourth poured-concrete. The smaller production room has two sandfilled-cinderblock walls and two of poured concrete. The flooring for both rooms consists of a 3” layer of wire-reinforced poured concrete over a 6-8” washed-stone base below grade. Sound is contained.

While all this mass means low bass loss, such loss as there is does occur upward, through the wooden joists and flooring to the kitchen above. There was no structure or budget for concrete up there. After consultation with several experts in architectural acoustics, the most cost-effective ceiling design turned out to be a layer of lead sheet above two layers of wallboard, all screwed into floating joists for support and acoustic isolation.

Floating joists? The kitchen’s floor joists are 2”x10” boards laid on edge across the concrete foundation walls. Interleaved with them are trimmed 2”x8” ceiling joists resting only at each end atop the studio’s concrete walls, such that their bottom is about an inch lower than the bottom of the kitchen floor joists. All ceiling materials are attached to these isolated ceiling joists. The intent was that none of the studio ceiling materials touch the kitchen floor joists, to minimize conduction of sound into the kitchen. About 400lbs of lead (in 36lb 3'x3'x1/16" sheets) was installed with the seams caulked to limit sound leakage. The whole area above the lead was filled with fiberglass for further soundproofing. The smaller 6.8’-high production room ceiling was isolated in similar fashion, with the ceiling joists supported at each end by 2”x8” boards attached to the concrete walls. There is some very deep bass heard in the kitchen, but it is quite subdued (estimated to be -25dB or less); higher-frequency sounds are inaudible. Kitchen footfalls and the trash compactor are minimally audible in the studio; due to nearly 20” of dead space above the production room, kitchen noise is even quieter there.

All the doors are solid-core with gaskets and sealed thresholds. The passageway to the outdoors (studio rear left) is through a 42"-wide steel door to allow easy transport of the recording gear (and to get the Klipschorns in). There is also an oversized storm door, which was an afterthought, but one day while the lawn was being mowed, Southwick closed the storm door and the outside noise level dropped about 20dB. The doors and intermediate dead air space make a significant difference in external-noise isolation.

The interior studio doors, walls and ceilings have been covered with black concrete stain, plus the floors have been coated with Rustoleum’s Epoxy Garage Floor sealer; all to limit concrete dust. [Black was chosen to prevent reflected light from changing the color of the projected image and to keep the image’s contrast ratio as high as possible. For video, a dark D6500 gray is the ideal, but difficult to get — DJW.]

In the ceiling between the front and rear upper corners of the studio and the production room are 3" pvc sewer pipes to run cables through. Next to the sliding glass door between the rooms is a cable-feedthrough port near the floor. End caps have been installed to limit sound leakage.

All ac power wiring is 10-gauge, which Southwick said is difficult to work with but provides 30A capacity through outlet strips located on the underside of 2”x4” chair rails around the perimeter of each room. [The outlets face down so plugs don’t stick out so far, and the outlets are essentially invisible for anyone not lying on the floor — DJW]

The room is heated and air-conditioned by a special Trane unit. It's not acoustically designed but is pretty quiet due to the variable-speed fan and use of flexible, fiberglass-insulated ducts above the leadlined ceilings. As the rooms are caulked and sealed, outside fresh-air intake and exhaust vents were integrated into the HVAC system.

Around a dozen acoustic foam panels (roughly 2’x4’; some Sonex and some pretenders, mostly wedged on both sides) are scattered about the walls. Some are suspended from the ceiling with cup hooks and fishing line, about 6” from the walls, while others are attached with hook-and-loop fastener strips. AES verified, with a propane torch, that they were fireproof: the foam melted but did not ignite.

The $850srp Hsu Research VTF-3 powered subwoofer has been set for its 22Hz rolloff to provide higher output (in accordance with company instructions). The subwoofer was positioned for smoothest response along the couch — the primary listening area — using a Phonic PAA2 third-octave rta. The subwoofer output is quite impressive.

Klipschorns (modified with John F. Allen’s midrange and tweeter drivers and crossovers) are the left and right front speakers, with an Allison CD9 center channel speaker and Allison CD6es in the rear. All are fed from a Lexicon DC-1 through a McIntosh 2205 (for the Klipschorns) and two Apt 1 amplifiers (driving the Allisons).

A PowerPoint presentation including photos served to highlight construction details.

AES: I would like feedback and comments.

Victor Campos: Do you find the lead affects the midrange?

AES: No. It's installed against the floating joists, with the seams caulked, above two layers of wallboard, so it's not exposed. However, recent auditions have detected reflections off the ceiling that most likely will need some further acoustic treatment.

Attendees were entertained with the DD5.1 soundtrack from the Balrog section, scene 36: *The Bridge of Khazad-dum* from the Extended Edition DVD *Lord of the Rings — Fellowship of the Ring* (scene 30 in the regular widescreen-edition DVD), and listening to CDs.

AES: I’ve noticed that folks auditioning the room really get into the movie and are loath to leave. As John F. Allen has shown, full-range quality sound-with-pictures does enhance the viewing experience, even on my 13” professional monitor.

Alan Southwick may be reached at:
PO 577, Newport RI 02840-0500; 401.683.7767

The JVC D-ILA projector

Ken Freed continued the meeting with a presentation of JVC’s D-ILA video projector.

JVC (formerly Japan Victor Company) was originally a subsidiary of the Victor Talking Machine Company, and uses the Nipper logo where it doesn't conflict with RCA.

Freed treated us to high-definition video from a D-VHS D-Theater demonstration videotape (MPEG2, 28 megabits/second — mbps, compared with DVDs which deliver about 3-8 mbps video, plus the audio). One D-VHS tape will hold a three-hour movie at 1920x1080i30. Around 80 D-VHS D-Theater titles are available for $25 to $30, playable only in certain $1000 vcrs.

The projector, based on 25 years’ development, was a JVC DLA-G150CL ($16.5k srp plus $2k srp for an appropriate lens). This projector uses three 0.9"-panel D-ILAs (Direct-Drive Image-Light Amplifiers, an LCOS [liquid crystal on silicon) technology), at WSXGA 1365x1024 pixels. [The picture was excellent with no dropouts or visible noise — DBH] [The exceptionally close spacing of the pixels (about a 92% aperture or fill ratio) on the 6'-diagonal screen resulted in no obvious demonstration of the pixelated nature of the technology, a problem that bothers me on almost all other pixelated projectors and displays. In addition, its color saturation and grayscale detail were excellent. Its only weakness was inadequate black levels, compared with my Runco CRT projector — DJW.]

There are three kinds of competing pixilated-projector technologies:

LCD has a reasonably high pixel count but a low aperture ratio (a measure of what percentage of the light actually reaches the screen.

Texas Instrument's DLP (Digital Light Processing, based on Digital Micromirror Devices, or DMD), which was developed for PowerPoint presentations, has a finer spacing between the pixels, but there must still be room so the pivoting mirrors don't collide. [The new HD2+ DMD has a higher aperture ratio and better black levels than its CAREFUL predecessors — DJW.]

JVC's D-ILA has the finest interpixel spacing (92% aperture ratio), the mirrors are stationary, and color is produced by bending the light in the liquid-crystal panel. The area between the pixels is not black, but takes on the luminance of the adjacent pixel, so you don't see the chickenwire effect that other displays have. Contrast ratio is rated at 600 [conservatively measured relative to many other manufacturers' techniques, but still optimistic compared with Joe Kane's findings across the industry — DJW]. D-ILA is the only non-film format that is approved by the motion picture industry for viewing dailies (the previous day's shooting).

Stephen Owades: My problem with all of these [pixelated-projector technologies] is that the blacks are not really black. I saw the same thing when I made the trek to New Jersey to see the DLP system in a theater [the d-cinema presentation of George Lucas' *Star Wars: The Phantom Menace* — DJW].

Outside the widescreen image area, but within the area projected by the D-ILA chips, the screen was slightly lit.

Freed: That is normally masked in an installation. In addition, the screen can be designed with negative gain. There is no problem pumping more light through the system. [Lossy screens help, but do not solve, the problem — DJW.]

[More recently, Joe Kane seems to have solved the issue of inadequate black with the new $10k Samsung SP-H700A DLP projector. I saw it at CEDIA 2003 demonstrate quite convincing blacks on a 1.3-gain screen — DJW.]

W. Kenneth Freed is at:

JVC Professional Products Company, Digital Broadcast and Professional Systems Division; 800.526.5308; kfreed@jvc.com.

Conclusion

The meeting concluded with the playing of further audio highlights from attendees' CDs and video highlights from several of Weinberg's DVDs, which are commonly used for audio and video demonstrations.

Southwick's studio complex exhibits much forethought and excellent results on a limited budget. The video and audio playbacks were most impressive.

The BAS executive committee has agreed to cross-advertising with the Audiophile Voice. *In exchange for our running their ad in the* BASS *(below),* Audiophile Voice *editor Eugene Pitts has agreed to promote the BAS in his publication.*

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Fame

The *BASS* can always use articles on audio matters. Send them to the editor, David J. Weinberg, *BASS*, 10705 E. Nolcrest Drive, Silver Spring, Maryland 20903-1006; 301 593-3230; **WeinbergDA@cs.com**.

Your byline will appear in good company, as a number of our contributors have gone on to eminence as audio and video writers and editors. Note that for meeting summaries there is compensation.

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