THE B A S SPEAKER

Journal of the Boston Audio Society

This is the second — Summer 2014 — issue of volume 36.

BASSv36n2

Please pass along the membership form, inside back cover, to interested friends, and use it when you renew!

ABOUT THE BOSTON AUDIO SOCIETY

POBox 260211
Boston, Massachusetts 02126 USA
603.899.5121; fax 603.899.6415
Email: **AMFMail@gis.net**
Web site: **www.BostonAudioSociety.org**

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

More than just a local society, the BAS attempts to speak to the worldwide community of audio and video enthusiasts. Founded in 1972, the BAS meets monthly to hear and discuss developments in audio, video and related fields. Guest speakers have included prominent engineers, designers, researchers, editors and reviewers, musicians and critics, broadcasters and recording producers. On occasion we hold joint meetings with the Boston chapters of the AES (Audio Engineering Society), SMPTE (Society of Motion Picture and Television Engineers), and the ASA (Acoustical Society of America). Our noncommercial journal, the B A S 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 video topics.

BAS members range from the novice enthusiast to the technically sophisticated. Consumers and producers of audio and video equipment are both represented. Members include freelance journalists, reviewers, and editors at audio and home theater magazines, as well as design engineers, recording engineers, consultants, and researchers who influence product development and the course of the industry. Some members work for manufacturers (as technician, design engineer, or marketing manager), others for dealers. All are devotees — audiophiles and videophiles 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 and video.

For these reasons, the BAS and the Speaker are a vital forum. As someone involved in audio or video, you might well find the group an interesting, helpful resource. Our meetings and journal might help shape the future of consumer and pro audio/video even while clarifying its past. Manufacturers, for example, can use the BAS to follow trends and developments, and to gather informed reactions to products and events. At the least we attempt to be a clearinghouse for ideas, helping various parts of the industry and experienced consumers keep in touch with one another.

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

THE B A S SPEAKER

Journal of the Boston Audio Society

Publisher: Robert L. Miller
Editor: David J. Weinberg
Associate Editor: David R. Moran

Contents

The *B A S Speaker* (USN 0195-0908) is published by the Boston Audio Society. A subscription is included with membership. Dues in the US are $40 a year; see the application form for rates outside the US. $38 of the US dues are for one volume of the *BASS*. For further information, please visit www.BostonAudioSociety.org or write the BAS, POBox 260211, Boston, MA 02126 USA. Editorial content should be sent to editor David J. Weinberg, *BASS*, 10705 E. Nolcrest Drive, Silver Spring, MD 20903-1006 USA; or email WeinbergDJ@BostonAudioSociety.org.

From the Editor

BAS members listen to music through their loudspeakers, so in this issue David Moran not only reviews ways of measuring speaker performance, but measured one currently highly successful design. In addition, following much discussion among members about the varied sonic quality of BSO broadcasts heard over various media, several members studied the signals. Plus our usual features. Enjoy.

To the Editor

"Finding concerts … " (Frederick J. Ampel; Kansas) Responding to Jack Reed's comments relating to my suggestion of Bachtrack ("To the Editor"; *BASS*v36n1p4), the site's FAQ and info describe how to use it, as further explained by the site's founder and managing director David Karlin: Bachtrack is a self-input site. Anyone who wants to can input their own events — free — as long as they register with us and are happy to learn our input forms, which concentrate on getting high levels of accuracy and searchability. If they don't want to do the work themselves, they can pay us a modest fee and we'll input the data they provide. A larger fee gets "premium listings" with pictures, red buy-tickets links and the opportunity for fuller descriptions, which is one of the main ways in which we make enough revenue to keep the site going. Our coverage is therefore a result of who has and who hasn't decided to work with us. Naturally, we'd love to have all events from everyone everywhere, and we're getting better at spreading the word. But we're a small office and haven't yet got anything close to complete coverage. If organizations such as the Carolina Ballet get in touch, we'll be only too happy to get them a login so they can post their events. As regards the representation of cities, different people like seeing things differently. We encourage people to use the largest local conurbation (it makes sense, for example, to label events in Beverly Hills as being Los Angeles, even although BH is a separate city), but some inputters feel strong civic pride and are insistent on inputting what we would consider quite small cities. For this reason, the search works first by "Region", which in the case of the US means a state. Having yet another level of granularity between state and city, in my opinion, would overcomplicate the user interface.

Home theater math errors? (Frederick J. Ampel; Kansas) Regarding Ethan Winer's "The Ultimate Living Room Home Theater" (February 2014 *audioXpress*; synopsized in *BASS*v36n1p6), I find the comments/information (?) mentioned somewhat concerning and I think seriously flawed conceptually. First, about his statement: "the more bass traps you have, the closer you get to a flat response, [and] note that a rectangle room has 12 corners" — as questioned by Weinberg, I also fail to see how he arrived at 12 corners. Second, all published information I have read does not recommend a goal of a flat room response, which would sound uncomfortable and abnormal to most listeners; further, few loudspeaker systems are capable of producing such a room response without extensive equalization, which could lead to overdriven amps and speakers as attempts are made to fill in acoustic nulls. Third, the number of acoustic panels he indicates as having deployed in his room would lead the reader to believe he has created an extremely dead space, which will make listening a strange experience, since the ear expects some ambiance and envelopment, which he seems intent on removing. Finally, the large number of bass traps might make the low-frequency room response weak. there seems to be a disconnect between his use of acoustic materials and the physics. Loudspeakers and the room integrate into a blended system, with the resultant spatial effect, frequency response, and acoustic signature being a combination of speaker performance and room acoustics (anomalies included). Looking at only part of the total picture likely will produce a distorted view.

Vintage & vinyl listening club. (13 September 2013 M. Duval email to Allison-Speaker Yahoo group) Laura Pope has started the Vintage & Vinyl Listening Club to raise funds for the nonprofit performance hall known as the Dance Hall, in Kittery, Maine. It is a 1920s Grange hall in immaculate condition, including its original stage. Pope borrowed vintage equipment for the kickoff event on Saturday, 16 November 2013. The 1977 David Bowie album *Heroes* was the vinyl of choice for the event. The evening was hosted by WBCN DJ Bradley Jay and Boston Globe arts writer Steve Morse (who also is a teacher at Berklee College of Music). There have been similar since; see www.Facebook.com/vvlisteningclub. Further expressions of interest will foster additional events. There will be good wine and a couple of tables of vintage audio equipment, all for sale.

Channel Master DVR+: more limited than advertised. (John Sunier, www.AudAud.com) Referring to "Channel Master's dual-tuner DVR+" (*BASS*v36n1; "CES 2014: Integration Among Technologies and Life"; item in middle of left column, page 16), although the manufacturer's website claims it is possible, I tried to record two channels at the same time and got the message: "You cannot record two simultaneous program at once — or watch one while recording another." Also, the report claims that the unit "requires" a user-provided external drive. [The DVR+ website does state that the device "requires USB external hard drive for full DVR functionality". DJW] Channel Master's tech support said that an external drive ("hopefully a Seagate") must be hooked up in order to simultaneously record two programs. However, when I press OK or PLAY to view a program I've recorded, absolutely nothing happens; I am still awaiting a response from tech support about why this occurs. However, the device has a small drive built in, which is able to record two hours HD if you record at 720p instead of 1080i. I like the idea that that is all it can do and I don't use an external drive. I just delete the programs so I can record others. Now I don't have shelves full of tapes that I haven't had time to watch. Some programs might use a higher datarate, so you could only record 1½ hours instead of two. In my experience PBS uses the lowest rate, so supposedly you can record two of their one-hour programs. At 480i you can record at least five hours of programs. My review of the DVD+ is at http://AudAud.com/?p=36681.

Open Forum

"Echoes of the Past: Rescued from Silence" (Mitch Steiner; Massachusetts) In the 6 April 2014 online *Boston Globe* is classical editor Jeremy Eichler's report that "the tools of physics give new life to recordings far too delicate to touch", specifically mentioning lost Woody Guthrie sessions and "a snatch of a ballad from 1860". Experimental physicist Carl Haber has won a MacArthur Fellowship "for a revolutionary image-scanning technology that has the power to pull sound from rare and fragile recordings without touching them". The LIbrary of Congress's National AudioVisual Conservation Center has been using his technology for several years, and this year "a large mill in Andover [Massachusetts; the Northeast Document Conservation Center; www.NEDEC.org] has become home to the fourth groove-scanning system in the country". Haber and colleagues named the system Image Reconstruct Erase Noise Etcetera (IRENE; a tribute to the first 78rpm disc Haber examined). Eichler explains the technology and gives examples of its application. "In the end, scientists and preservationists are at least now joined in working toward the same goal of pushing back against the approaching ocean of silence."

"Sound bite: Despite Pono's promise, experts pan HD audio." (Mitch Steiner; Massachusetts) Stephen Shankland (www.CNET.com/news/sound-bite-despite-ponos-promise-experts-pan-hd-audio/) reports that "by raising $4.3 million on Kickstarter, Neil Young's startup shows an appetite for better sound quality. The only hitch: experts say there's little point going beyond CD quality." Fraunhofer's Bernhard Grill is quoted as saying that "people should worry much more about speakers and room acoustics" than higher-resolution audio files. "A prominent part of the case against high-resolution audio is a 2007 study by E. Brad Meyer and David R. Moran of the Boston Audio Society that concluded listeners couldn't tell the difference between SACD and DVD-A music on the one hand and CD-quality versions of the same recordings on the other. In that experiment's 554 tests, listeners correctly identified an SACD or DVD-A recording compared to a CD only 49.8 percent of the time — in other words, they didn't do better than randomly guessing." [The Meyer/Moran paper "Audibility of a CD-Standard A/DA/A Loop inserted Into High-Resolution Audio Playback" is available at www.AES.org/e-lib/browse.cfm?elib=14195 (free to AES members, $20 to others; for a free *.pdf, email DRMoran@aol.com).] The article cites claims on each side of the argument. However, "one thing everyone agrees on is that the debate is mostly irrelevant for the mass market for music". In a separate email E. Brad Mayer wrote that "the Monty Montgomery/Xiph Audio demo [mentioned in the article, and available at http://XIPH.org/video/vid2.shtml] is fantastically good — every 'intuitively correct' digital myth is neatly exploded and the demos are brilliantly clear and straightforward. No one should write a word about digital audio who does not understand (and accept) the contents of this Montgomery demo."

"Sound Overexposure: It's More Dangerous Than We Thought!" (David B. Hadaway; New Hampshire) From the announcement of an 8 April 2014 lecture by Massachusetts Eye and Ear Infirmary's Director of Audiology Sharon Kujawa, part of the "Ears Looking At You, Kid" seminar: "Exposure to loud sound can cause temporary or permanent hearing loss and injures delicate structures of the inner ear. When sensitivity recovers after noise, it has been assumed that this indicates reversal of damage and no persistent or delayed consequences for auditory function. In contrast, new research has shown that loud sound exposures, even those that result in completely reversible sensitivity losses, can cause ongoing degeneration of the cochlear nerve. This neurodegeneration alters how ears age after noise, and likely contributes to speech-in-noise difficulties other perceptual anomalies commonly associated with inner ear damage. The clear message from these studies is that noise is much more dangerous than has been assumed."

Commentary and News

by David J. Weinberg

"The High-Rez Roadmap — II" (March 2014 *Audiophile Voice*) is George Witterschein's view that "there's obviously a tidal wave of interest rolling in the direction of high-resolution audio" as he describes some of the websites he "stumbles across": David and Norman Chesky's www.HDTracks.com "is the granddaddy of high-resolution musical sources" (the site's email notices of upcoming offers and specials include discounts); www.LinnReords.com is "vast, and the selection of material an formats is deep"; Cookie Marenco's www.BlueCoastRecords.com (DSD-encoded content); Todd Garfinkle's MA Recordings (www.MARecordings.com) offers discs containing high-resolution files; Reference Recordings' www.ReferenceRecordings.com offers downloads (most are available at HDTracks) and data discs (bit copies of the 24-bit/176.4kbps recordings); Cisco Systems' founders Sandy Lerner and Leonard Bosack started www.SonoLuminus.com (includes the Dorian Records catalog , plus their own and others' recordings; some on audio Blu-ray discs); Robert Witrak's www.HighDefinitionTapeTransfers.com (includes some DSD-encoded files); content from Everest Records is available at HDTracks, Amazon and iTunes; Gimell Records (www.Gimell.com) specializes in its recordings of the Tallis Scholars (some on audio Blu-ray discs); and Germany-based www.Katzenberger-Music.com ("billing themselves as offering Das Original vom Tonmeister, or the original from the recording engineer"; a substantial portion of the site's text is only in German).

Sound On Sound (March 2014):

- **Dolby and dbx in software.** Paul Nagle reports on his evaluation of the U-he (www.U-He.com) Satin ($130) tape emulation plugin for MacOS X and Windows, which he lauds as "highly tweakable … with a wide range of applications from subtle warming to multitrack glueing, … [having a] clean informative interface, useful presets", and more. The feature that attracted my attention was inclusion of Dolby A and B, plus dbx types I and II. I twice sent an email to the company to find out if the decoders could be used to decode Dolby- and dbx-encoded recordings (such as my complete collection of dbx-encoded LPs, which I could import into my Mac through my Davis-Brinton phono preamp), but have not received a response.
- **"Monitor Wizard: Establishing Project Studio Reference Monitoring Levels."** Hugh Robjohns agrees with Bob Katz, and others who perform mixing and mastering, that establishing a consistent and reasonable standardized listening level for their work is beneficial to their hearing health and to facilitate reliable prediction of the high-and low-frequency effects of their adjustments to the tracks. Robjohns starts by referencing Matt Houghton's article on gain staging (http://SOSM.ag/sep13-gainstagingDAW) and his own article on ITU-R BS.1770 loudness standards (http://SOSM.ag/feb14-endofloudness), explaining that a reference level provides a reliable base for aural decisions and doesn't require major investment in new equipment. Proper gain staging sets a nominal level with consistent and appropriate headroom and adequate S/N ratio throughout the equipment chain; [it is as important in a home playback system as in a recording chain]. Experience has shown that "our ears quickly become used to a 'standard' volume, and we can then judge levels, loudness and dynamics by ear. … Mixes end up being more consistent, and working to the new loudness standards becomes easier because we know intuitively when something is 'too loud' or 'too quiet'." He acknowledges the cinema industry's adoption of this concept in the 1970s, and that as broadcast TV sound balancers adopt appropriate reference monitoring levels, "their mixes fall in line with the [EBU R-128; http://Tech.EBU.ch/docs/r/r128.pdf] requirements almost automatically". Robjohns further points out that a consistent reference level enables working "in an optimal part of the Fletcher-Munson equal-loudness curves", minimizing "problems with too much or too little perceived bass". He provides a seven-step procedure that includes suggested nominal listening levels and notes that Bob Katz has an 'honor roll' of well-mastered pop CDs on his website (www.DigiDo.com/media/honor-roll.html), listed with the appropriate monitoring level settings for equal perceived loudness.

ARSC Journal (Spring 2014):

[As I have previously noted, I believe the organization (www.ARSC-Audio.org) deserves support, plus its journal and newsletters are well worth the nominal cost. DJW]

- **"Copyright & Fair Use"** is Tim Brooks' regular section. In one report ("Don't Mess With Lessig") Brooks notes the numerous "complaints about abuse of the takedown procedure established by the Digital Millennium Copyright Act, whereby any copyright owner can demand that a website take down material it alleges is infringing or face huge fines. Organizations like the RIAA have automated this procedure with robots that scour the Internet for fingerprints of songs they want suppressed, then fire off notices to YouTube and others. … Recently Google reportedly received its 100 millionth takedown notice this way (according to one report, it was its 235 millionth). Problem is, not all of these uses are infringing. Fair use makes it perfectly legal to use copyrighted works without permission in some circumstances, but rights-holders routinely ignore this.” The burden of proof of legal use is on the user. “The RIAA can continue to fire off masses of sometimes dubious notices without any consequences.” The rock-band Phoenix’s label Liberation Music submitted a takedown notice to YouTube for “a posted lecture by [Harvard law professor and] copyright reform advocate Lawrence Lessig, because it included a portion of the band’s recording *Lisztomania*”. Lessig responded that scholarly use was “classic fair use”. Liberation Music threatened to sue Lessig unless he recalled his complaint. He and the Electronic Frontier Foundation sued Liberation for its false takedown notice. Liberation soon settled, admitted their error, agreed to change their procedures and pay unspecified damages. In another report (“Copyright & Fair Use Update”), Bruce Epperson discussed the status of several cases involving Sirius XM Radio, which revolve around the vagaries among federal and states royalties laws of who can collect and who gets paid.

- **“Let’s Go Surfing”** does not refer to the Beach Boys’ songs but identifies some lesser-known website sources for music downloads: www.QuBoz.com (French megastore; deep catalog; plenty of bargains, but can be hard to find on the “rather busy site”; blocks US access to some titles); www.HDTracks.com (has “high-definition recordings you won’t find elsewhere, especially remasterings of classic ‘50s and ‘60s jazz, rock and classical”); former BBC engineer Andrew Rose’s www.PristineDigital.com (“passionate about his transfer work”; primarily historic classical and some jazz); www.eClassical.com (in Sweden; nice selection of classical downloads); www.PrestoClassical.co.uk (some downloads are blocked from US downloading); www.HighDefTapeTransfers.com (specializes “in reasonably priced historic recordings transferred directly from commercially released reel-to-reel tapes; many rarities; excellent sound quality; limited selection); www.HyperionRecords.co.uk (specialist UK label); Andrew Wardle’s www.SonoMinstrels.co.uk (“unusual site established to celebrate the 10 recordings made by the ‘Zono MInstrels’ for British Zonophone in 1913”; also other early recordings); Judaica Sound Archives of Florida Atlantic University’s http://FauJsa.FAU.edu/jsa/home.php (goal to “collect, preserve, and digitize Judaica sound recordings; to create educational programs highlighting the contents of this rich cultural legacy; and to encourage the use of this unique scholarly resource by students, scholars and the general public”; >4000 78rpm recordings; some downloads limited to 45-second snippets; others restricted to listening on site or at “qualified institutions”); and Dartmouth Jewish Sound Archive http://DJSA.Dartmouth.edu (with a scholar’s reasonable request, professor Alex Hartov — Alexander.Hartov@Dartmouth.edu — can grant access from the user’s location). All this should keep you listening for quite a while.

“Streaming with the MPEG HE-AAC Audio Codec” (April 2014 *Radio Magazine*; www.RadioMagOnline.com) is discussed by Fraunhofer IIS’s Matthias Rose, who reminds us that the “first Internet radio broadcast took place just 20 years ago”. [High-Efficiency Advanced Audio Coding (HE-AAC) was developed by a group of companies (one of which was Fraunhofer IIS) and is part of MPEG-4 Audio and the US’s ATSC-M/H (digital TV broadcasting mobile/handheld) standard. More at http://EN.Wikipedia.org/wiki/Advanced_Audio_Coding.] Rose reports that “during 2013 the number of Internet radio listeners grew to ~147 million, up 11% from 2012. … There are several audio codecs in use for streaming but they vary significantly in their ability to provide and maintain reliable audio quality at low bitrates.” The BBC, NPR, and Cienradios, Pandora, and iTunes Radio mostly use HE-AAC. NPR tested six codecs (HE-AAC, LAME, MP3, AAC-LC, AMR-WB+, and xHE-AAC [extended HE-AAC]) “over a wide range of bitrates and programming format, and found that HE-AAC was the top performer, citing high-quality audio, low bitrates and broad compatibility with target device platforms such as Android, Apple iOS, Widows, Mac, [plus] Adobe Flash and leading HTML5 browsers. … HE-AAC does not require content distribution fees and provides support for audio-specific metadata for loudness normalization.”

“2160p and UHDTV Update.” (April/May 2014 *Widescreen Review*) Joe Kane synopsizes the state of UHDTV, including the relevant international standards such as ITU-R BT.2020, which “specifies UHDTV image system parameters for production and international program exchange. The implication of the document is that the enhancements in it, over our current HDTV system, are higher resolution(s) and a larger color space.” Kane continues to strongly hold that the color space in BT.2020 “shouldn’t be implemented in displays … [as] there are strong arguments being put forth by organizations such as Munsell that even if it were implemented it wouldn’t work from a human factor point of view.” The problem is that in contrast with the primary colors in NTSC (SMPTE C) and HDTV (ITU-R BT.709), each of the UHDTV primaries is defined at a point on the CIE chromaticity diagram such that it is almost a single frequency (each primary has an extremely narrow bandwidth), leading to widely varying color-perception among viewers, including those who oversee program color quality (see M.D. Fairchild and D.R. Wyble, “Mean Observer Metamerism and the Selection of Display Primaries”; www.Cis.RIT.edu/fairchild/PDFs/PRO30.pdf). In addition, the narrow bandwidth of each primary color will make it more difficult to generate high brightness from the display. Kane’s article includes most-illuminating spectra showing white as generated by the BT.2020 primaries, a CRT, a UHP lamp-driven DLP projector, an OLED display, and an LED-backlit LCD display. Most consumer UHDTV displays increase resolution but keep HDTV’s ITU-R BT.709 color space, and thus are not offering all of the enhancements UHDTV can offer. There also is evidence that “an increase in resolution over 1080p HD isn’t enough by itself to sell the idea of UHDTV to consumers”; taking visual acuity into account, the higher resolution will require the viewer to sit relatively closer to the screen (which history has shown is not likely) or get a much larger display to benefit fully from the increased resolution. Kane goes on to recommend redefining the specifications and standards for the future of video in a manner that does not quickly limit content, distribution and display capabilities.

audioXpress (June 2014):

- **“NAD D3020 Hybrid Digital Amplifier”.** Gary Galo reviews this redesigned and partially digital upgrade of an amplifier that some of us remember from CES in Chicago more than 30 years ago, when the original was demonstrated simultaneously driving several then-expensive large full-range loudspeakers, delivering acceptable sound quality. The original amp’s performance belied its low power rating (20Wpc), while introducing the concept of soft clipping and high dynamic power output capability. Galo found that this hybrid analog/digital design “is a resounding success, … [offering] a level of sonic performance that will be difficult to match in its price range. This amplifier will cause Class-D skeptics like me to rethink their views on this power amplifier topology. … I highly recommend it.” [There is also a positive review by Robert Archer in the June 2014 *CE Pro*. DJW]

- **"Psychoacoustics (Part 1)"** begins Ron Tipton's series on "audio enhancing, spatial location, noise reduction, and sound masking". This chapter focuses on virtual bass generation, showing how we can be fooled into 'hearing' low frequencies that aren't actually radiated by our loudspeakers — "an advantage for apartment dwellers and anyone with thin walls". Tipton reports that "perhaps the oldest application is that of the 'missing fundamental', although it seems that term was not used until Leo Beranek discussed it in [an updated revision of his] book *Acoustics*. But the application itself was discovered in the early 1700s by some composers who found they could trick the listener into hearing bass notes that weren't really there by playing a combination of higher frequencies. The need to do this arose because 32' organ pipes produced enough sonic energy to cause structural damage to the European churches in which they were installed. These composers found that the perceived pitch of a combination of tones equally spaced in frequency is the constant difference in frequency (the missing fundamental)."

New Old Thoughts On Measuring Home Speakers

by David R. Moran (Massachusetts)

Based on the "Lies, Damn Lies, and Audio Gear Specs" panel presentation at the October 2013 Audio Engineering Society's 135th convention, in New York.

Prologue: The Problem

This ramble through what to measure and how best to measure it (anechoic, and installed in a domestic situation) covers loudspeaker goals past and future. It includes old news, settled issues, straw charges, circular reasoning, won battles, sermons to choir, minor horn-tooting, and hopefully few errors and cheapshots. Its summary is that modern measuring technologies typically can mislead if not spatially/temporally averaged.

Referring to the measurements in Figures 1 and 2: Which are helpful/useful/practical to try and comprehend and address? Which correlate with what is heard in a room? What do they mean? Are they really audible?

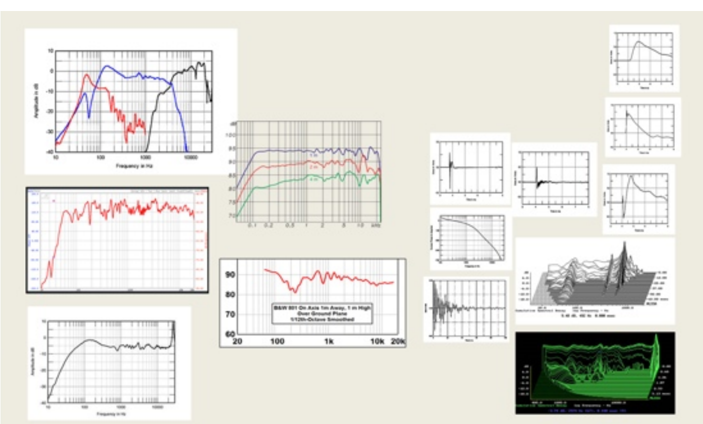


Figure 1. These are snapshots or slices of ultrahigh-rez on-axis frequency responses, and/or of 'time behaviors'. Most of these measurements, certainly the time-related ones (right group), are by John Atkinson, who has measured more speakers than anyone else. The frequency-related ones (left group) are by him and also by Don B. Keele, Alvin M. Foster, or Brent Butterworth. Some of the left group might be averaged a bit.

Now this batch:

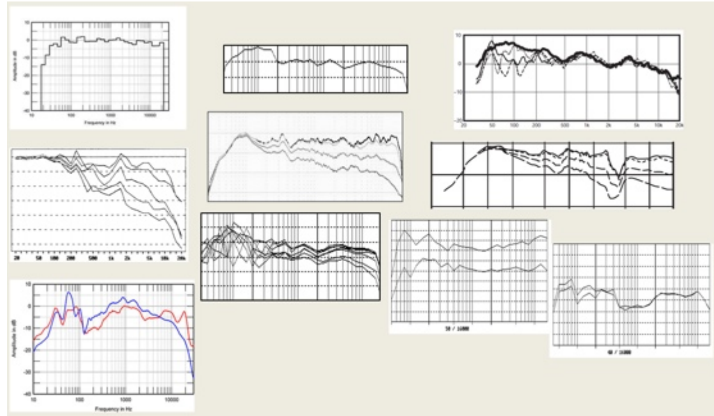


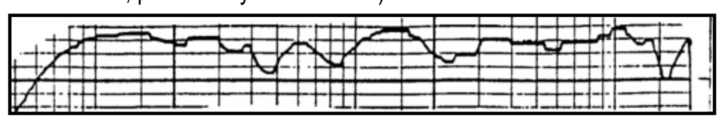
Figure 2. All are averaged responses, even if not always at high resolution (all show frequency-response measurement, and are by Atkinson, Floyd Toole and Sean Olive, or the author).

Some History

Turning to 'modern' audio history (which means omitting Institute of Radio Engineers and Western Electric work by Chester Rice, Edward Kellogg, Irving Wolff, Abraham Ringel and others from the 1920s and 1930s), we see that in 1942 these Acoustical Society of America (and later RETMA — Radio-Electronics-Television Manufacturers Association, predecessor to CEMA, which became CEA) measurement standards existed:

- Situating a speaker system anechoically or outdoors,
- sticking a mike on some front axis, and
- running a curve on a chart recorder.

In accord with the last point, here from the later 1980s is the manufacturer's measurement of the Celestion SL12 Si on axis (presumably tweeter axis, presumably full anechoic):



But all the way back in 1944 Jensen Radio had presciently written that " … significant measurements representative of performance in live rooms require a determination of the total-radiation-frequency characteristics (as contrasted with the simple axial pressure-response frequency characteristic)." (For this reference, many thanks to audio historian Tom Tyson.)

By the mid-'50s, from the writings of Jensen, Harry Olson, Leo Beranek, Edgar Villchur, Stewart Hegeman, Arthur Janszen and probably others, the sonic importance of dispersion within the listening space became widely appreciated, at least in the US. Designs changed.

In 1958 Villchur conceived dome drivers to go with his revolutionary acoustic-suspension woofer system. AR published frontward dispersion curves (Figure 4), drivers smoothly flushmount.

In 1966 Roy F. Allison, who'd joined AR some years earlier, undertook (with Chuck McShane) to redesign the huge-selling AR-3 into the huger-selling AR-3a. It employed smoother and wider-dispersing midrange and tweeter; the latter is, mostly, still unsurpassed today in its dispersion. As important, maybe more, Allison lowered the woofer/midrange (WM) crossover almost an octave, from 1kHz to under 600Hz, to smooth the beamy and rough off-axis response (audible, and visible in Figure 5), which typically results from taking a 12" woofer all the way up to 1kHz. In the midrange and elsewhere, the AR-3a sounded notably better than the AR-3, and for many listeners set a new playback standard. (It still had

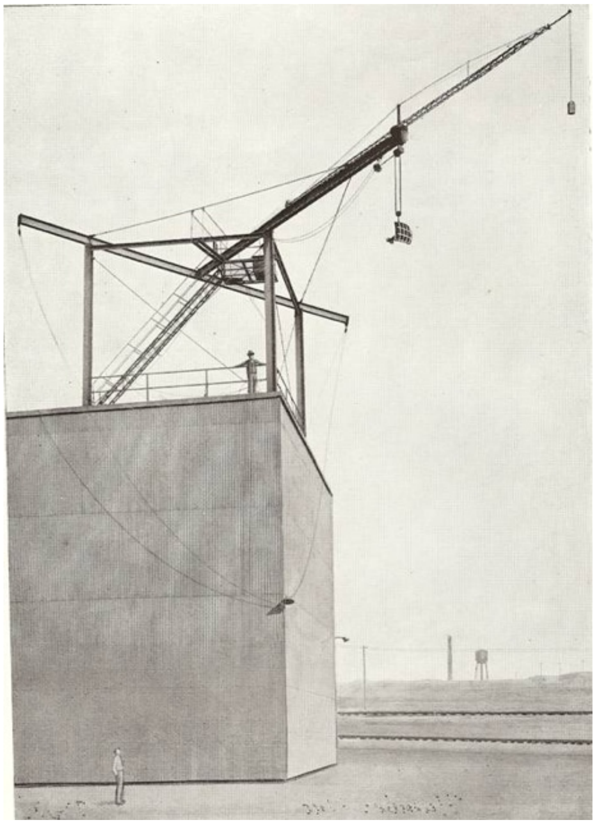


Figure 3. Jensen photo shows tower equipment and 40'-cube "enclosed free space" room

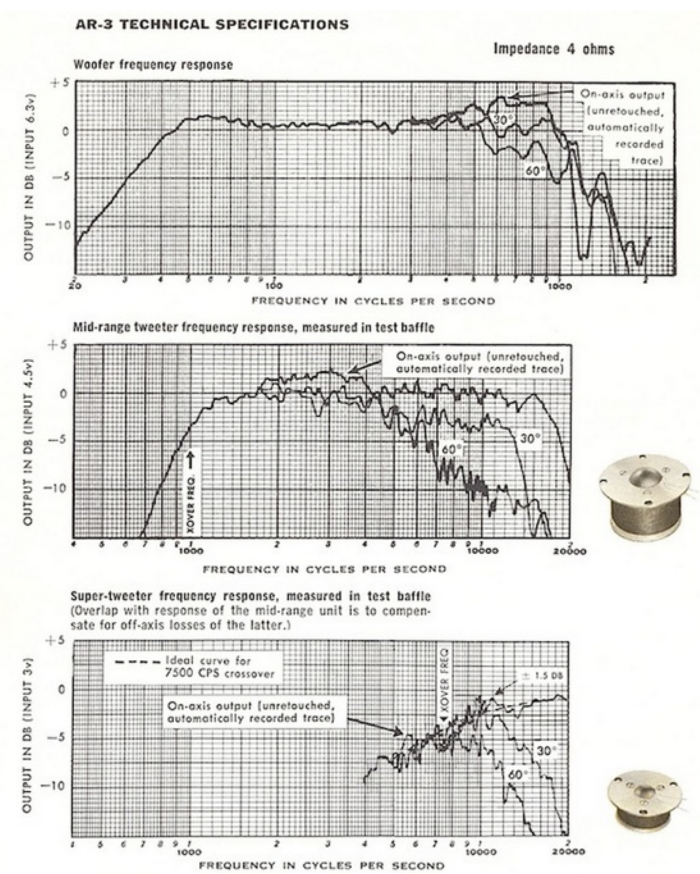


Figure 4. Late-1950s Acoustic Research AR-3 on- and off-axis frequency-response curves.

generated cabinet-edge reflections.) Not widely appreciated was how it had improved not just from newly designed drivers but by crucially dropping the WM crossover point.

The goal of playback that sounded natural, meaning widely dispersed, spread further. Dome tweeters became ubiquitous.

Even though in a competing worthy technology, Paul W. Klipsch espoused use of horns for control of directivity, Olson in 1967 specified a performance goal (Figure 6) — a strict one, often unmet today.

A great leap forward in distribution of midrange and treble into the room came in 1968 with the Bose 901. Nine midranges on three cabinet surfaces, eight facing rearward, toward the wall, although they fully radiate forward also, up to the treble. It included an outboard EQ-controller. The 901 radiated sound in comparatively uncontrolled fashion throughout the front of the room. Explicitly designed to not sound like anything prior, it was a huge and longlived success. But for some reason it did not spur intensive investigation into nonconventional radiation patterns using dynamic drivers.

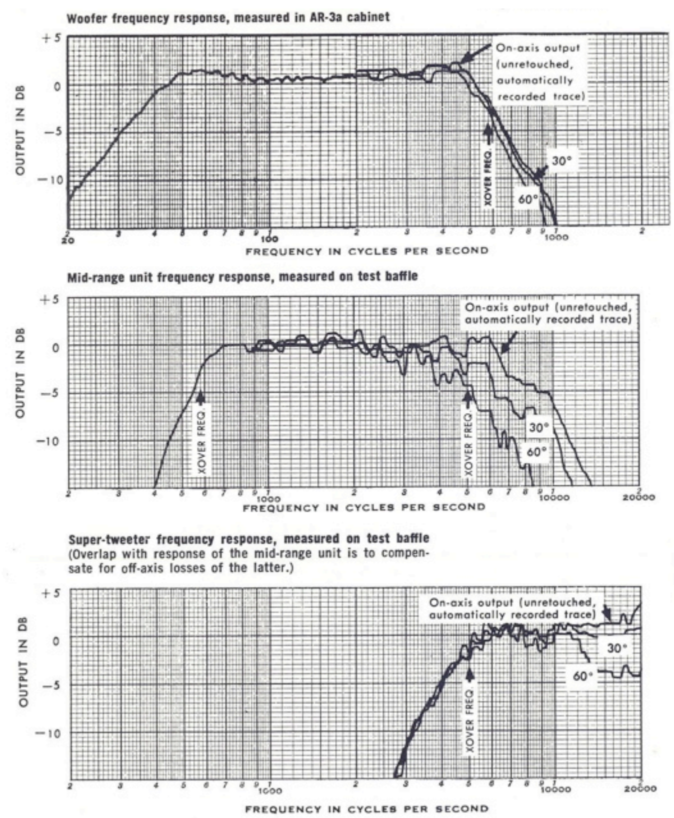


Figure 5. AR-3a showing the benefits of an improved midrange and tweeter, and lowered woofer/midrange crossover frequency.

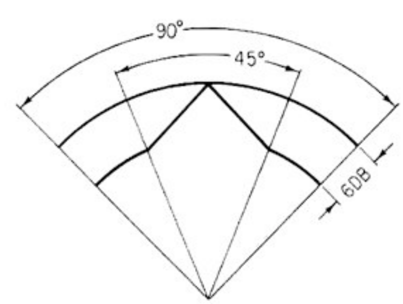


Fig. 2. The directivity pattern of a good speaker system should fall between these extremes over the 30 to 15,000 Hz range.

Figure 6. Olson's directivity performance goal.

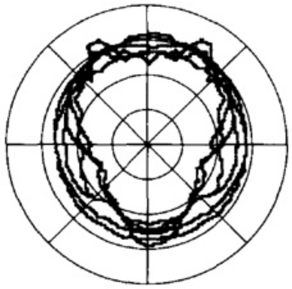


Figure 7. Bose 901 polar plot 20Hz-20kHz, octave-bundled (by Mark F. Davis).

Planars were gaining serious notice with their airier, wide-range bidirectional front/back radiation. Quad and KLH initially set the high standard, followed by several others. Decades later, Linkwitz's dynamic dipoles have kept it up.

Variety of radiation pattern aside, there were battles over the details of frequency response measurement, angular and/or in toto. The significantly sales-influencing Consumer Reports repeatedly took heat from many sides, and not because they graphed in sones.

Casting Dispersions

Following Beranek, Allison was emphasizing extreme forward treble dispersion (the first designs were multipanel as well) to try and re-create performance ambience, a closer simulation of what we hear in terms of spaciousness: "For envelopment, you need widespread energy generation … reverberant energy broadcast at very wide angles from the loudspeakers, so the bulk of the energy has a chance to do multiple reflections before it reaches the ear."

English loudspeakers in the 1970s developed mostly oppositely, at least in the treble, toward favoring wider-diameter and therefore beamy tweeters. Airless highs, but with sharp treble-image focus, was espoused and achieved.

Some US investigators, and some companies, came to pay attention not just to smooth direct sound and treble control but also, slowly, to overall horizontal radiation pattern. In 1972 and again in '78, MIT doctoral researcher Davis wrote, from experimentation and also crediting others prior, that what we hear in an enclosed space *is* a source's frequency response as a function of radiation pattern. If you vary it and only it, you clearly change the sound heard.

In the early 1970s Allison cut a nonhard dome tweeter in half, inverted it and, placing a second little dome above it (Figure 8), made a tweeter system that puts out nearly as much treble sound sideways as forward. Unequaled but unpatented, it hasn't even been copied.

Bose designs aside, for some leading US companies, generally wide dispersion went without saying. In 1976 Peter W. Mitchell and I gathered a newspaper discussion-panel of Boston-area designers, and radiation pattern was never once mentioned (Allison, Victor Brociner (Scott, Avid), Winslow Burhoe (EPI), C. Victor Campos (AR, KLH), Henry Kloss (KLH), Andrew Petite [Kotsatos] (KLH, Advent, Boston Acoustics), and Daniel von Recklinghausen (Scott, KLH) were the participants; Amar Bose refused).

By the late 1970s and early 1980s everyone sensible seemed to be slowly converging on the idea that what sonically matters most are frequency responses on- and off-axis, along with their integrated power response: total reflections in the room. It took years, but the zeal for 'timing' and phase behaviors as chief determinants of natural sound — a zeal abetted if not caused by new measuring technologies — slowly began to wane.

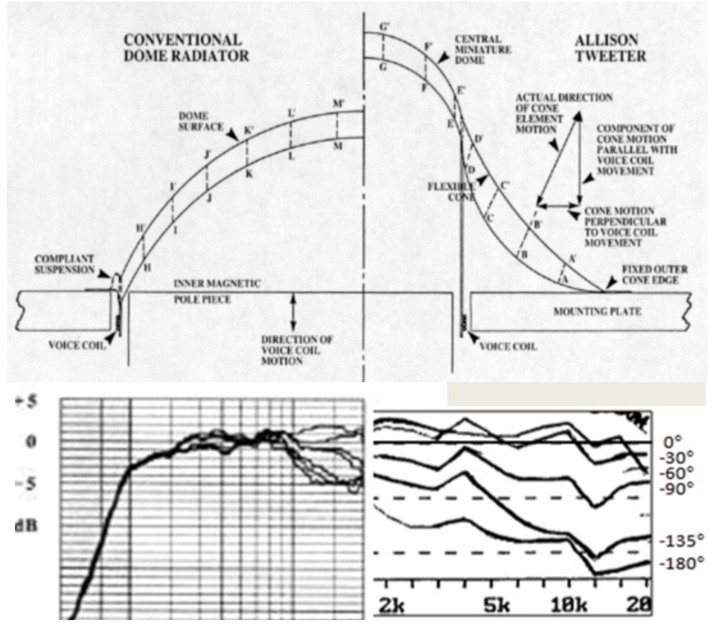


Figure 8. Allison tweeter diagram, plus measurements by manufacturer (left) and author (right).

Shown next (Figure 9), as traditional frequency by horizontal angle front to back of cabinet, are a range of theoretical horizontal radiation patterns: conventional at top (meaning frequency response falls as a function of angle), very-wide dispersion in the middle, very beamy at bottom. (Theoretical bidirectional, balanced off-centerline, and some other unusual radiation-pattern goals are not shown, such as designs with another tweeter on the side(s) or back.)

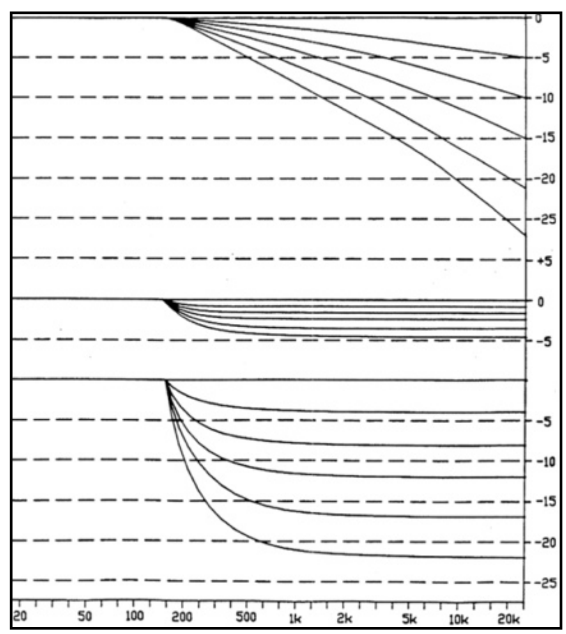


Figure 9. Theoretical horizontal radiation patterns.
In each of the three groups of curves, top-to-bottom:
0º (on-axis), -30º, -60º, -90º (sideways), -135º, -180º (directly behind).
Top group: Conventional narrowing with frequency, aimed for by most every company
Middle group: Very widely dispersing 'omni' design, a la Allison, Ohm, Beolab, dbx, Keele CBT, a few others. Some are 'equi-omni', while others beam but do so very widely.
Bottom group: Narrow, beamy dispersion down into the lower midrange; very difficult to achieve.

Figure 10. Conventional radiation pattern as polar plot, KEF105.x, octave-bundled (left) and narrower (right) (by Mark Davis).

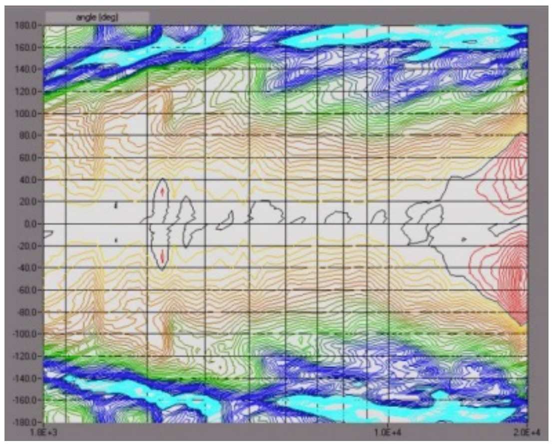


Figure 11. More-omnidirectional radiation pattern as polar map: Beolab treble "acoustic lens". Color = level. (by David Moulton, Manny LaCarrubba).

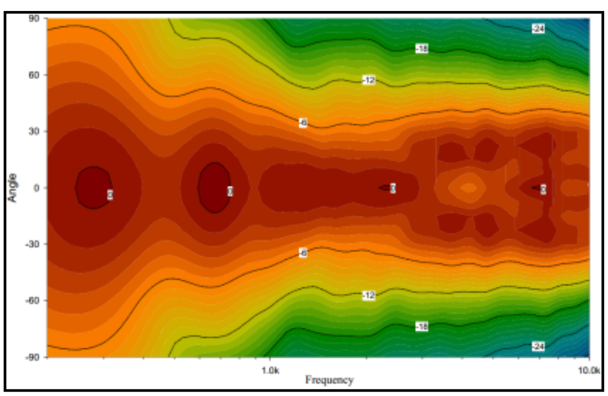


Figure 12. Narrow controlled-directivity broadband: Summa polar map (by Earl Geddes).

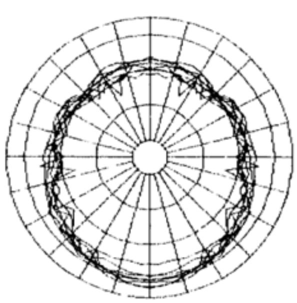


Figure 13. Angle-invariant frequency response except for level: wide-beaming dbx Soundfield 1A, forward orientation (by Mark Davis).

Beyond Davis' previously mentioned experimentation, it was in the 1980s that Floyd Toole and colleagues began to test for and quantify listener preferences in loudspeaker playback. Radiation-pattern smoothness mattered. This did not always represent new thinking or discoveries, but because it was rigorous and scientific, the work was extremely important.

How best to measure frequency response?

In the late 1970s Ivie introduced a handheld third-octave RTA with pink noise and a slow setting but no temporal (continuous) averaging. Regardless, it was easier to use than a big GenRad, as had been employed by some speaker manufacturers, and others, for frequency measurements both aggregated in-room and sometimes angular. B&K, Altec-Lansing, Crown, HP and others had expensive products for this application. McIntosh's Roger Russell, now audio historian, opted for this approach. To give reliable on-paper results, these RTAs typically entailed some intensive manual averaging.

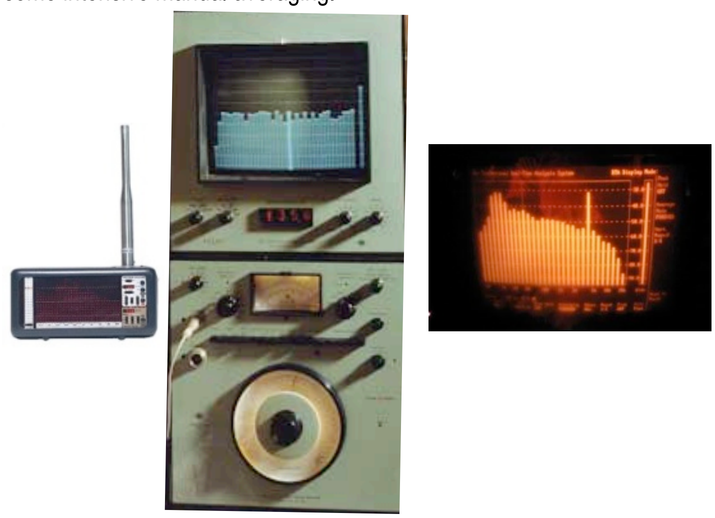


Figure 14. Older RTAs, from Ivie to dbx.

In 1986 dbx pro introduced the first RTA with temporal averaging, a precision instrument permitting easy spatial averaging and quickly showing settlement of its pink noise. Speaker measurement veteran Tomlinson M. Holman promoted this averaging noise-based approach for theaters (soon advocating use of a new RTA similar to dbx's), and ANSI/SMPTE standard 202M-1991 informative supplement A.4.6 codified it.

Many other handheld RTAs came into being afterward: Audio Control, Goldline, TerraSonde, Coustic, and more (the Ivie has been updated), also a half-dozen laptop (TrueRTA is the leader) and smartphone apps (S6D AudioTools is the leader; this last program discussed later). Many but not all still lack temporal averaging.

The Heyser Era: Toward Better Resolution?

Starting in the latter 1970s and dominating today, measurement technologies initially using swept sinewave with bandpass post-filtering and, later, pulse-based stimuli, typically with FFT processing, produced much higher-resolution frequency data. They aspired to produce anechoic-like results in any environment. Some point to HP as an early leader. Richard Heyser's pioneering 'energy/time' work soon generally came to stand for the new methods. Presently there were, and still are, dozens of such 'FFT' approaches, and hi-rez testers have variously embraced TDS/TEF, MLSSA, SIM, AP, LMS, SATlive, Smaart, Clio, Listen Soundcheck, and Smith & Larson products, along with numerous laptop 'digital EQ' implementations. Looking at their snapshot results, as shown at the start of this article, we see vast amounts of frequency and time/phase detail, slices in time, and even granular pistonic misbehaviors captured via laser interferometry. Because it could be measured, it was.

But are the results accurate, useful, repeatable? Most important, are they audible?

Early on there was skepticism. A leading researcher wrote privately that he would not trust these new systems below 1kHz. A leading speaker designer said privately that everything Heyser wrote was either incomprehensible or wrong. (At least one early Heyser *Audio* review was riddled with measurement errors, including being off by 1.5 octaves about a crossover point, and more.) A third expert sweepingly labeled these approaches 'useless seeing'. Nonetheless, the leading magazines completely adopted the technologies, and for decades have been printing the results.

It seemed a powerful advance in tools, except attention now turned from frequency response by angle and in aggregate, to microscopic analyses, in superhigh resolution, of a system's step, impulse, and 'transient' responses; cumulative decay, delayed resonances, nonlinear distortions. (Many if not most of which show up in frequency response, loudspeaker drivers being minimum-phase.) Year after year, magazine after magazine, consumers puzzled over speaker reviews, effusive text accompanied by unfathomable graphs.

In recent years, 'timeslice' measurements have thankfully appeared to wane in popularity, probably because they're inexplicable, also probably because we are so demonstrably deaf to much phase/timing and waveform information. More than once has John Atkinson commented on their lack of correlation with what is heard, as opposed to his averaged in-room frequency measurements, which do correlate. Martin Colloms is another who eventually came around to concur in the importance of room response. So today most testers appear to be back taking measurements of frequency response, usually via pulses, not noise, and seldom suitably averaged. The FFT products have improved, and reviewers measuring with them have gotten savvier, although not as much as design engineers. Still, it's essential not to rely on snapshots.

Problems from not averaging

1. By definition it's a limited look, which can mislead.
2. Your technical understanding can be deceived and your sense of terms skewed (e.g., 'nearfield').
3. You might think and write wacky things.
4. You might miss what is demonstrably audibly important.

At worst, modern measurement snapshot technologies, when not averaged, make audio consumers convinced that the human ear can hear, sense, sort, parse, and detect events and sounds that it cannot — in frequency and bandwidth resolution, in time/timing, and in other ways. They have made the eye dominant over the ear.

For example, Keele, Heyser's successor reviewer at *Audio* magazine, once wrote for publication that direct sound is what's responsible for our sense of spectral timbre and tonal balance. Of course the opposite of the case, the ear being a detector that needs time to make its judgments; first arrivals help with localization. Only a few months ago, a Cambridge physics PhD with a long career in audio design asserted privately that it's the first 8ms, which is, as the quip goes, not even wrong. Most recently in an interview, speaker 'guru' David Smith opined that "we judge frequency balance with largely a time-windowed approach. Late-arriving sound is ignored. Also, this time window is long for low frequencies and short for high frequencies. In effect it is the steady state or room response for low frequencies and typically just the direct (anechoic) response for high frequencies. At midfrequencies it might contain the first floor or back-wall bounce, but later reflections are generally under the level required for audibility". This will come as major news to any designer (or customer) choosing between the dispersions of 1.5"- vs a 0.75"-diameter tweeters, changing baffle width or rounding cabinet edges, comparing lower-midrange playback with woofer placements close to or far from the floor, or indeed to any of us who treat any livingroom surfaces.

I submit that it's likely that such completely backward thoughts would never have formed without the modern measurement snapshot capabilities.

In the mid-'90s a science-oriented audio magazine decided not to review a respected new speaker because their Audio Precision system showed it had serious 'problems'. The reviewer was moreover greatly bothered by the design's supposedly antiquated first-order crossover filters. Figure 15 shows that actual unit's averaged 30Hz-20kHz room response using noise and a temporal-averaging system (the average of optimal positionings, meaning that this degree of goodness is not typical):

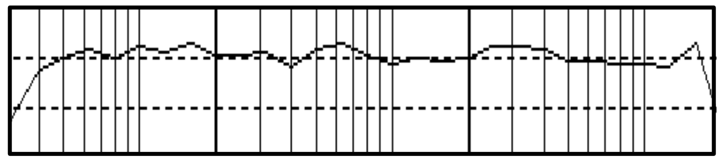


Figure 15. Allison AL125 temporally averaged room response.

But forget measurements of either sort. To listeners experienced and inexperienced alike, does this 'AP-rejected' design (Allison AL125) sound natural: smooth and accurate? Absolutely, unusually so.

More and more, the new hi-rez 'anechoic' frequency responses, when taken singly, have proved to be of iffy repeatability and unclear validity. Of course one could not see dispersion effects clearly, but also much else could not be properly interpreted: too much 'grass', meaning narrowband-data chatter.

Those wavelengths below middle C

Let us turn to the crucial area below 400Hz or so, the lower midrange, where so much of music and speech reside, where the oboe tunes the orchestra. It is here that the modern measurement systems are not just weakest but most misleading. Yes, they're by definition anechoic. Acquiring 'anechoic' data in domestic listening rooms means the data must be time-windowed to eliminate reflections, which typically limits frequency resolution, compromising it up to midfrequencies. Either way, what end does it serve to always directly show speaker smoothness across the <100-400Hz midrange and lower-midrange octaves (red arrow)? Drivers on their own are always smooth there.

***Figure 16. Recent KEF curve from* Sound&Vision.**

But not in use in typical consumer listening rooms.

Often you can predict this problem, to some degree, just by looking at the design, as with the woofers up high on the cabinet, which will be examined presently. Note that these octaves are not bass, and do not sound like bass, as anyone knows who listens to the preconcert oboe or tries to sing up to middle C, =~260Hz.

Perhaps modern measurement systems' curves below 500Hz-1kHz should be accompanied by "Warning: Does not apply when product is played in rooms." To put it most politely, it's confusing to contemplate the curves expressed in most speaker reviews by graphs that display direct sound from the lower midrange on down. To the ear in a listening room, there's no such thing; domestic direct-sound effectively doesn't exist on its own. Take the A below middle C — 220Hz. The wavelength is ~5', the period ~4.5ms. Given typical room and speaker layouts, even in large rooms, and the many-millisecond integration time of the ear, how can we hear or sense 220Hz directly, or even as predominant?

In his amazingly thorough must-read 2008 book Sound Reproduction: The Acoustics and Psychoacoustics of Loudspeakers and Rooms, Toole writes of this graph (Figure 17): " … the [audible effect of the dip] will depend on the relative contributions of direct and reflected sounds in the room. In a very dead room, the audible impression may be more influenced by the frequency response on the prime listening axis, such as the on-axis curve… ."

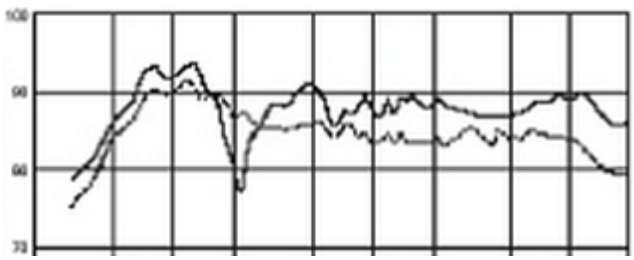


Figure 17. Solid line is 'axial frequency response', with notch at 220Hz (from Toole).

But what listening room doesn't reflect, meaning is unable to reflect, 220Hz? What room has floor, front wall, and sidewall so far away that a wavelength of 5' and a period of 4.5ms would ever be heard directly on its own?

No normal domestic listening room.

Toole of course well knows that total output is what matters here, and says as much, though he has added that the direct sound "preconditions" the ear, which I don't buy, nor did any of the experts I queried. Still, why do reviewers imply the opposite, why show a confusing axial output? I suggest the answer might arise from use of misleading, nonaveraging technologies that permit it.

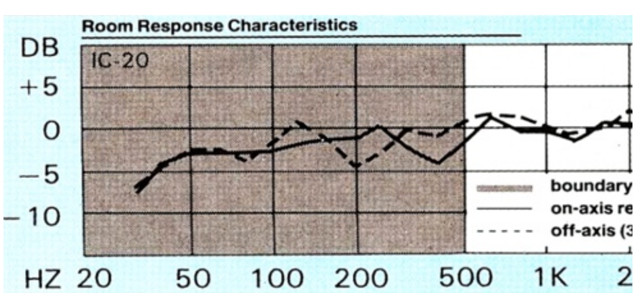


Figure 18. More implausible lumps.

Ed Foster similarly published (long ago) the above curves (Figure 18) purporting to distinguish 125Hz sounds on-axis and off-axis. What use is that? And this was of an against-the-wall Allison design, to boot. He too knew better, thus shaded the range as 'boundary-dependent'.

Closely related, but worse to my thinking, is mismeasurement of lower-midrange adjacent-boundary (aka 'Allison') effects. Altogether audible, but not shown in modern wall-free graphs, are any dips in lower-midrange response somewhere around or below middle C. The cause? The driver covering this range is positioned high up on its cabinet, or on a stand, such that its energy as reflected from the three near boundaries — front wall, sidewall, floor — all arrives back at the driver partly or wholly in opposite polarity ('phase') with its direct output.

This is not just the so-called floor bounce. Gravity has nothing to do with it. The floor effect isn't stronger or more important than any other boundary's. From every near corner there are seven reflections (Figure 19), equally influential:

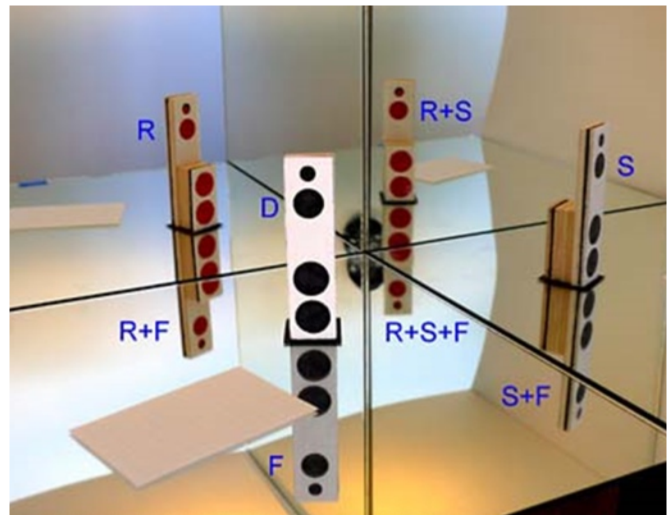


Figure 19. Linkwitz model, after Olson and Moran (rear, side, floor reflections).

Figure 20 shows magazine measurements printed for expensive speakers: smooth performance 50Hz-2kHz — supposedly.

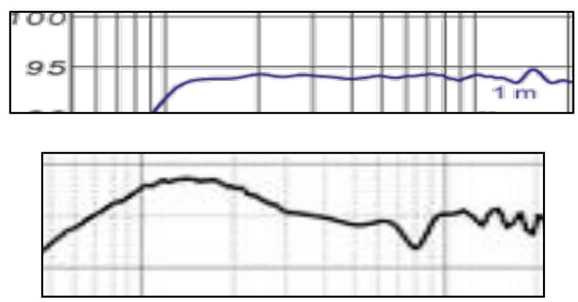


Figure 20. At top is a Revel design (Brent Butterworth measurement); the lower graph is unidentified (John Atkinson measurement).

No one will experience such smooth performance as this in domestic listening rooms. There will be ripple, typically dips. These dips result from the designs' having similar, that is unstaggered, distances to the three near boundaries, and while technically not classifiable as direct sound, it might as well be, since the reflected energy loads the driver and changes its frequency response; throughout the room the dips are therefore more constant than not.

Thus are the ineluctable bundling and nonseverability of direct and reflected energy at these lower-midrange frequencies manifested.

Figure 21 shows some measurements of the effect, 70-400Hz. In each graph, the upper curve is theory, lower curve(s) reality.

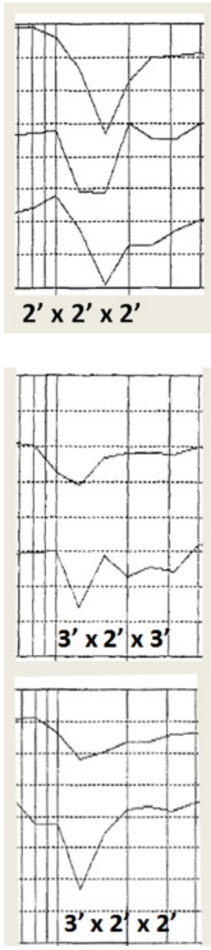


Figure 21 Top of each graph shows predicted theoretical woofer output, 70-400Hz, as loaded by those distances to the near corner; below which are actual in-room pink-noise averaging-RTA measurements (by Moran).

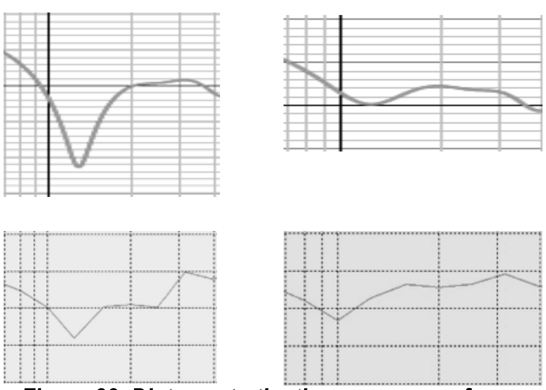


Figure 22. Distance to the three corner-surfaces: Left graphs — 30"x30"x30" Right graphs — 39+"x29+"x9".

Completely by coincidence (Figure 22), the bottom curve in each of the above two pairs recently came by email from Europe from a boundary-effect denier. At top is theory, below which is measurement, technique unknown (as is the speaker). This was happy accidental confirmation from an outside source. Note that staggering the distances, as at right, smooths things out much better.

What that means is that if, for whatever reason, the driver covering below middle C simply cannot be near the floor, then we as users should try for approximately 2'x3'x4' or 2'x3'x5' distances. The improvement (fill, smoothing) will be audible.

To see the theoretical boundary-distance calculations and get a feel for the variables, the free Jeff Bagby Excel program is available at http://Audio.Claub.net/software/jbabgy/BDBS.html.

Toole has written that "adjacent-boundary effects are broad trends, not highly frequency-specific". Depending on many variables, this is not altogether untrue. It certainly is the case that the dip and ripples are modified by ceiling height, furnishings and, to a somewhat lesser extent, mike position. The best recent measurements of the problem probably appeared in *Stereophile* a few years back, taken by Atkinson (Figure 23). As noted, snapshot measurements show none of this.

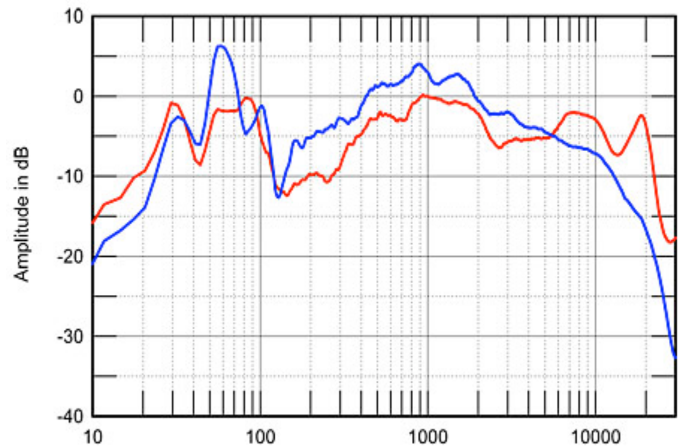


Figure 23. Most-dramatic averaged room-measurement of adjacent-boundary-caused dips: kilobuck systems from Wilson (red) and AudioNote (blue), whose woofers are well up off the floor.

Having double-drivers covering the 90-400Hz range seems to ameliorate the roughened and reduced output, partly anyway; Allison and I are investigating the phenomenon. See the Infinity P362 review, next.

Illuminations

As mentioned, many testers, like some designers, using the new technologies have learned to average their results, and are now more readily able to do so. In any enclosed space, using noise and temporal/spatial averaging, you can get a good picture of the reverberant field and total speaker output, choosing closer for proportionally more direct sound, and farther for proportionally more diffused sound. The crucial development is that some of the FFT-technology systems now include temporal averaging. One app even labels the feature 'pink noise settlement'. The best appears to be the AudioTools FFT (not called an RTA) from S6D (Studio Six Digital; Figures 24 and 25), although using its narrowest-resolution settings (1/12th- or 1/24th-octave) gives the usual results of 'what does this mean, what is one supposed to do with this?', quite apart from the arguable inaudibility of such narrowband detail. I have casually compared the built-in iPhone 4S mikes with the good Dayton Audio iMM6 outboard plugin one, widely available for under $20 including calibration file, and they appear a plenty good-enough match below 10kHz. (Other inexpensive flat-response omni — resulting from their being of small diameter, less than a half-inch — mikes also have become available.) I haven't yet tried the S6D programs (there's a conventional third-octave RTA as well with temporal averaging, though its filters are class I, not steep) on an iPhone 5, nor have I experimented enough with the FFT acquisition-windowing settings (Hamm, Hanning, Blackman) to say anything intelligent about their differences, and I haven't yet performed a thorough and detailed comparison with my gold-standard dbx RTA plus Earthworks M30 mike.

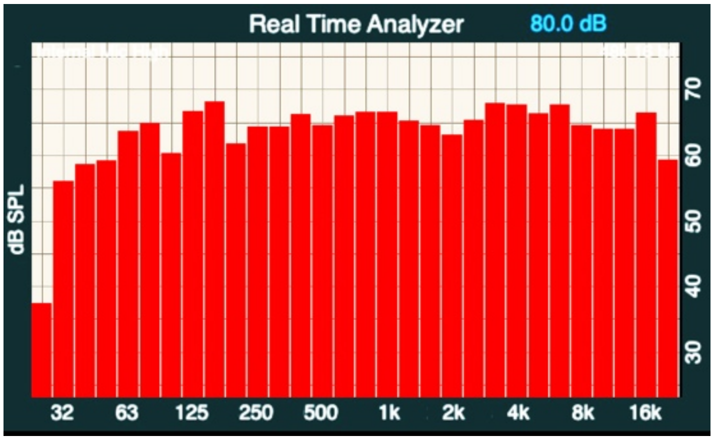


Figure 24. S6D Audio tools pink-noise measurement, temporally averaged third-octave conventional RTA mode, taken with iPhone 4S built-in mikes.

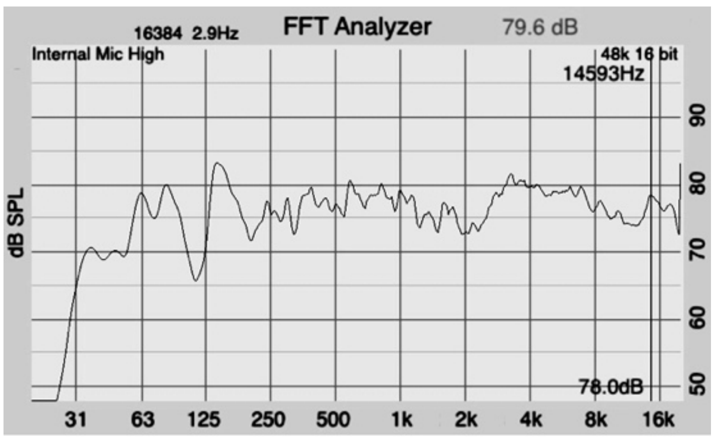


Figure 25. S6D Audio Tools pink-noise measurement of a different system than in Figure 24, temporally averaged sixth-octave FFT in RTA mode, taken with iPhone 4S built-in mikes.

But since the Boston Audio Society war cry from 40 years ago was 'audio nirvana for $49.95', the ~$10 S6D AudioTools suite (it comprises numerous other functions; Andrew Smith is one fertile audio-apps developer) is well worth investigating for anyone with an iPhone. Moreover, there are many competitors, as googling iPhone RTA will uncover; RTA Audio and Octave (which has much narrower-resolution capability despite the name) have the slowest display settling options, although Octave does not yet bypass the iPhone's built-in low-frequency highpass filter.

The new attention to averaging capability might partly be the influence of Toole's masterful book, a highly substantiated reference promoting careful frequency response and dispersion measurements, of course along with much else. Around the same time the book was published, Infinity designed an utterly conventional inexpensive tower, executed with care, which Harman's Sean Olive has regularly employed in blind preference-testing, that beats all comers for all listeners. Toole suggests that the P362 (now succeeded by the 363) might be as good as needed — that is, at the point of diminishing returns. One can readily think of a couple of improvements, but it's well worth examining its simple details and performance, and the upcoming review does so.

In conclusion

Let us end this ramble by quoting Toole's book, closing the loudspeaker measurements circle back to the thinking of Jensen, Villchur, Olson, Allison, and Mark Davis:

" … it is possible to process the right set of anechoic data and to predict with impressive precision what listeners will think of a loudspeaker when they hear it. … the sets of curves that provide useful information for our eyes, and the predictions that suggest how the product might be judged by listeners in a room, are simply not widely available. It is long past the time in this mature industry that manufacturers of loudspeakers … need to provide comprehensive anechoic data on their products. … The descriptions of acoustical performance offered by many of the significant players in the loudspeaker business are simply insulting in their inadequacy. … Ask manufacturers for real high-resolution (1/20th-octave) anechoic measurement data on their products … . A minimum of on-axis and several off-axis curves extending to 60° or more off axis, vertically and horizontally. … Sound power and directivity index are useful additions. … None of this is difficult or mysterious."

Toole being so influential, perhaps this criteria insistence of his — along with savvier driver placement for smoother boundary augmentation — will spread further. May it be so.

And Olive's blind preference-testing (US patent 8,311,232 B2) has design significance that can hardly be overstated: Smooth horizontal radiation pattern, with smooth room response, really sounds better — invariably.

Design goals/steps

1. Decide on uniform radiation pattern: conventional, figure-8, all wide, or all beamy.
2. Design for all crossover seams and stitches to be less lumpy than, say, ±3dB above 150Hz as measured third-octave. Smoother is better. Check at higher resolution. (Figure 26 is a graph of decent speaker performance.)

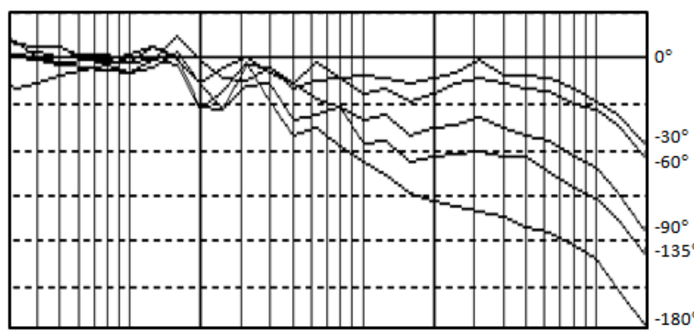


Figure 26. Graph of decent speaker performance.

3. Ensure that driver(s) covering below 300Hz are near a boundary; or force staggered placement in situ with respect to the three adjacent boundaries; or employ some sort of peaking filter (preferably switchable) for the 100-200Hz octave.
4. Balance low-frequency quantity and extension.
5. Check room response in a range of typical domestic listening venues. Temporally and spatially averaged measurements all.

Imagine where things would be if speaker designers worldwide had decided from the getgo, ~60 years ago, to heed the Jensen monograph, the Villchur work, the Allison work, the Olson spec, the Bose marketplace impact, the Mark Davis papers, and now the Toole/Olive research?

Imagine tomorrow's consumers being able to choose among different radpats, different but all similarly smooth, from, say, Infinity, Genelec, JBL, Linkwitz, and perhaps many others, plus the leading loudspeaker companies from Japan, England, Denmark, and Germany.

I close with more new old news from the last country: as I was preparing this AES convention presentation, a leading US audio investigator emailed me that "directivity … was found, by the East Germans just before the fall of the wall, to be one of the two things that count: 45 loudspeakers, 40 listeners, three rooms, 32,000 A/Bs. Put results of semantic differentials like near/far into a statistical hopper and compare with physical measurements. Only two count: Listening window response and directivity vs frequency. Period. Full stop."

A Winning, Blind-Preferred Design: The Infinity P362 Loudspeaker

by David R. Moran (Massachusetts)

[Reprinted, in slightly modified form, with permission from Linear Audio *magazine (v6; order at www.LinearAudio.net; €23.50/~$32.75). Note that the BAS has recently purchased this very pair for meeting use. DRM]*

Photo 1. Infinity P362 (grille removed).

Over the last several decades, that is, before, during and after their stints at audio giant Harman, the investigations by Floyd Toole, Sean Olive and colleagues into audibly preferable loudspeaker performance have set a welcome goal, welcome at least for consumers and do-it-yourselfers if not necessarily or always for manufacturers. These Harman blind tests rigorously quantified and thus reconfirmed, again and again, much historical conventional wisdom, but did so in newly detailed, hence irrefutable ways. Smooth broadband horizontal response more or less uniformly dispersed, frontally wide if not wider, has been advocated for a half-century or more. The eminent acoustical engineer Harry Olson in 1967 wrote a prescriptive paper on speaker performance goals, and his specification and advocacy were not totally original even then.

But many loudspeaker manufacturers, year after year and design after design, have sometimes espoused other, unsubstantiated goals, having to do with the supposedly crucial importance of direct sound, or with "time" and/or "phase" alignment, the unimportance of total room response, the need to reduce lateral reflections through wall treatment and/or reduction of sideways output, and more. For some of them, the Olive, Toole et al. research, with its emphasis on constancy and uniformity of frontal horizontal radiation, will — hopefully — prove instructive and, more important, influential.

As a veteran loudspeaker tester on the lower tiers of audio journalism (*CD Review*, *BASS*, *Digital Audio*, *Speaker Builder*, *$ensible Sound*), I thought it would be interesting to acquire and independently measure the Harman Infinity design that has been preferred by listeners of all sorts in these blind comparisons throughout recent years. The Primus 362 (or perhaps a similar predecessor) has handily beaten more-expensive Polks, Martin-Logans, Klipsches, the B&W 80x, and others unnamed. All of whom measure more lumpily in their horizontal radiation pattern than the Infinity tower. (It is dumbfounding how the B&Ws, with their decades-old unpleasant and altogether audible kink at the midrange/tweeter crossover, has become a classical-recording standard.)

My own 25-year-old measurement approach has always had much in common with Harman's, although the third-octave dbx professional temporally averaging RTA isn't as high-resolution in the frequency domain. I too have long been interested chiefly in horizontal radiation pattern. I measure first outdoors, at seated ear height, all the way round the cabinet, 7'-9' away. With some spatial averaging, I capture pink noise response at 0º (typically on the front tweeter axis), then -30º, -60º, -90º or directly sideways, and then -135º and finally directly behind, -180º. The outdoor measurements are half-anechoic, so to speak, but floors are of course a constant in listening rooms, and it's always useful to see how front and sidewall reflections are going to have to fill in that lower-midrange dip somewhere around or below middle C which so many designs exhibit (about which more later). Since my RTA (made by dbx, where I worked through the 1980s) does feature temporal averaging, which is crucial for reliable, repeatable measurement, I move my Earthworks M30 mike within approximately a cubic foot or a little more at any given datapoint — up/down, left/right, in/out.

Then I return indoors and take room responses similarly, closer listening positions to get more speaker and less room, and farther away to see more room effect. The RTA permits memory averaging of curves and other arithmetic manipulation including subtraction for normalization (making the axis response a straight line so that response differences by angle become stark). As I mentioned, I don't weight frontal measurements; anyone who's drawn drapes to cover hard bare front wall or picture window behind a stereo loudspeaker system knows the striking difference it makes, damping rearward output and changing the sound and imaging significantly.

Naturally, before any measuring, I listen to a single speaker and to the pair. Now I wade into subjectivist terminology. On well-recorded jazz trio, the P362 sounded just fine: smooth, accurate, natural, all that good stuff, including for the most part the lower midrange and upper bass, where I always listen hard for lumps and, worse, for missing oomph. The Infinitys let you easily hear recording differences among the excellent Brad Mehldau, Greg Reitan, Roger Davidson, and Brazilian trios, sensing which is always fun. The imaging was extraordinary in spatial "float" and stability. Treble playback (brushwork, cymbal) was not the last word as to air and depth and allowing the ability to hear into the recording venue, which undoubtedly results from middling high-frequency dispersion (better than the norm, though, with a ¾" tweeter, albeit slightly sunk) and consequent lack of highs delivered out into the reverberant field of the room. (This is compared with my broadly dispersive references.) Bass was rich enough, and almost low enough, on its own — the vent appears to be tuned around 50Hz, maybe a little below, as these are pretty big tower cabinets — and the speakers played plenty loud.

Speaking of higher-SPL assessment, I didn't do a crank-it-up distortion comparison with my reference Allison AL125 system, which is a similar (driver complement, baffle width) tower configuration, although somewhat shorter. That company tested their drivers severely for distortion out to the 5th or 7th harmonic or something, although never claiming (I believe) that they were audibly better in that regard than other competent cone and dome designs. The P362 drivers feature MMD (metal matrix diaphragm) construction, which claim superior pistonic, nonresonant behavior. A few years ago famous listening researcher David Griesinger, then employed by Harman Lexicon, told me how impressed he was by this new technology as employed in high-end Revel speakers, especially on taxing choral playback (which often exhibits lots of IMD and related problems). So I did play the P362s at healthy, realistic levels on the hoary Edwin Hawkins Singers "O Happy Day," which has very clean loud chorus, and the playback was pristine, perhaps more than usual on such difficult material.

Then I performed an informal blind comparison that was so striking I initially felt something must be wrong and probably discountable. I should describe it withal, as it's an area I'm perhaps a little more alert to than other audiophiles. I listen to stereo more than multichannel, but I audition all material over unusually dispersive designs. Without particularly having set spatiality as a matter of my focus, I placed a single Infinity snugly side by side with an Allison, therefore similarly affected by the room, and switched back and forth between them while listening to TV sports announcers. I didn't know how I hooked things up and which speaker was playing, although the P362 is more sensitive, hence louder, than the AL125. Over one of the single speakers, these voices sounded perfectly natural and accurate but also flat and uninteresting — uninvolving. Over the other single speaker there were marked depth and space and sense of the booth, mikes, announcer movement and interaction, etc. It was the damndest thing, and I was immediately suspicious of prejudice and predisposition on my part, given such loose blindness to my setup. I mean, I know full well that the unique Allison midrange and, more important, Allison tweeter put out nearly as much sound sideways as forward. Still, the difference was extremely surprising in degree, and upon checking by moving close enough to tell which cabinet was playing at a given moment, I did confirm that it was the Allison which was spacious and airy and deep and the Infinity comparatively one-dimensional. I recalled Toole's strong comments, in his recent magnum opus but moreso in online forums, about how much he dislikes conventional stereo and mono, that is, nonmultichannel playback. And the Infinity P362 is far from being some old beamy early-'80s English design, or even close. Its ¾" tweeter is actually quite wide in coverage compared with today's ubiquitous one-inchers and much wider-diameter tweeter designs back then. Although, per the Harman preferability testing, this is notionally a speaker as good as is needed, period, I too would rather not live with with a single pair alone.

Enough of my perceptions and judgments. I trust my measurements more, indeed any good dependable measurements, and so should you.

Figure 1 shows in-room response from 30Hz to 20kHz (5dB/vertical division) of a single Infinity P362 loudspeaker at numerous typical listener and cabinet locations, meaning the speaker was placed at different spots along or somewhat out from the long wall, and pink noise measurements were taken at a few listening positions across the room at seated ear height on a couch. This living room has three entryways, a fireplace, and a large TV; the room dimensions are ~16'x22'x7.5'.

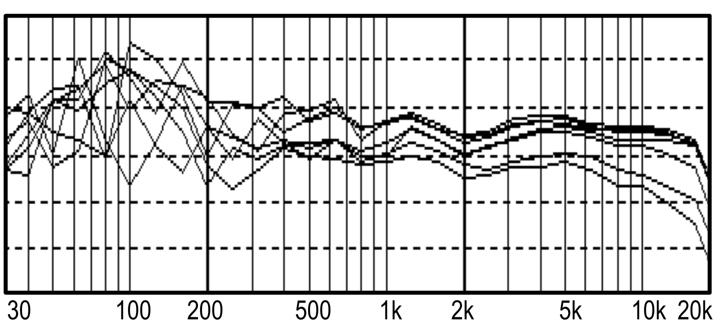


Figure 1. Room response of a single Infinity P362 loudspeaker at numerous typical listener and cabinet locations. 5dB/div.

The variation below 400Hz is typical of a design where, as with most, the woofers are high up on the cabinet rather than down near the floor, although as I say the double woofers appear to ameliorate the suboptimal loading situation.

Figure 2 shows the same measurements, spread out for clarity and plotted at 10dB/div.

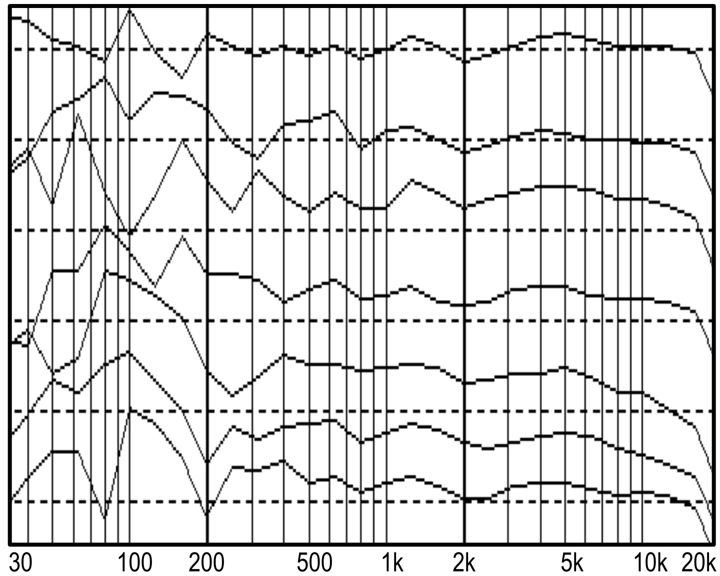


Figure 2. Same measurements as in Figure 1, spread out for clarity and for better observation of response variability below ~400Hz. 10dB/div.

Figure 3 is the careful best positioning of the Infinity unit, at three different locations where results were close to identical. As it happens, and don't you just love when it does, it's also virtually identical to the average of all those room responses in Figure 1. The verdict: The P362 is a well-voiced and balanced design with, in my particular room, welcome and euphonious and smooth bass richness below 200Hz, if you're lucky or skilled enough to place it properly. (Harman's instructions, like almost every speaker company's, are wholly, and frankly surprisingly, inadequate in this regard.)

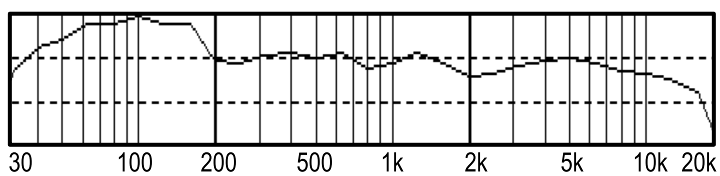


Figure 3. Average of the measurements in Figure 1, and also the room response of an Infinity P362 in a few optimal cabinet and listener locations. 5dB/div.

To achieve smoothest response below ~400Hz (which lies in the octave above middle C), whether or not you're able to achieve the euphonious bass rise shown in Figure 3, the P362 cabinets should be close to or against the front wall, not pulled out into the room.

Figure 4 is the room response of one of my references, in this venue the Allison AL125, positioned per the manufacturer. A very close match in tonal balance. It is too lean-sounding for me in this open listening room, so I usually boost the bass a bit. This curve also is an average of a few fairly close listening positions; AL125 response is seldom so exemplary at farther, less-scrutinizing listening seats.

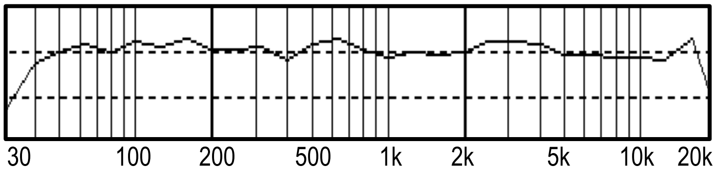


Figure 4. Allison AL125, averaged room response at a few nonfar listening positions, placement per user manual.

Figure 5 is Harman's measurement of the frontal and total output of the P362.

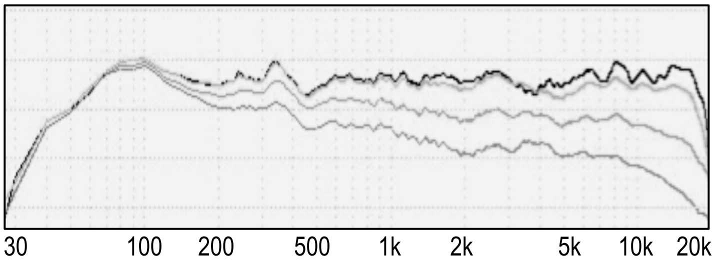


Figure 5. Harman's measurements of the Infinity P362. The dark top line is direct sound, that is, on axis. The gray curve right below it is ±30º horizontal plus ±10º vertical; Harman calls this the average listening window. The third curve is what the company calls average response estimated for first reflections in typical domestic listening rooms. The bottom curve is sound power: an average of 70 anechoic responses measured at 10º intervals on horizontal and vertical orbits at 6', each response weighted according to the proportional area of the sphere represented by each measurement. 1/20-octave resolution.

Figure 6 is my on-axis measurement outdoors, averaged, 7'-9' away and ±5º or more horizontally, also a few inches vertically above and below the tweeter axis.

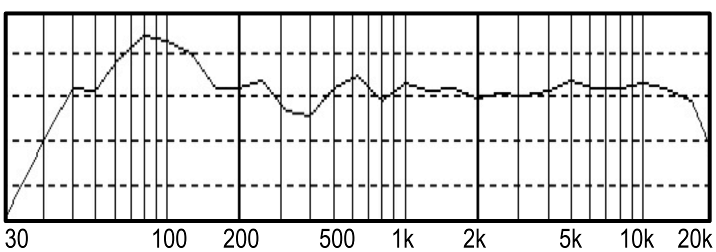


Figure 6. On-axis response measurement outdoors, averaged, 7'-9' away and ±5º or more horizontally, also a few inches vertically above and below the tweeter axis.

Figure 7 continues my examination of horizontal radiation, outdoors, from a single Infinity P362 on a concrete surface far from any other surfaces. The first curve is the on-axis direct response per Figure 6 above, with the successive spreading angles following: -30º, -60º, sideways, -135º, and directly behind.

Not to get all nerd about it, but please note the amazing congruent striation of the responses at -30º and -60º and then the same, at a lower level, at -90º (sideways) and the likely reflection from a corner as represented by the -135º output. This is not common. No wonder the image steadiness was rocklike. And my listening, while not casual, did not entail weeks of comparing John Eargle cuts (say) and similar splendidly recorded ensemble recordings.

Figure 8 is the data in Figure 7 normalized, meaning the on-axis response has been made ruler-flat in order to emphasize the *difference* in output by horizontal angle.

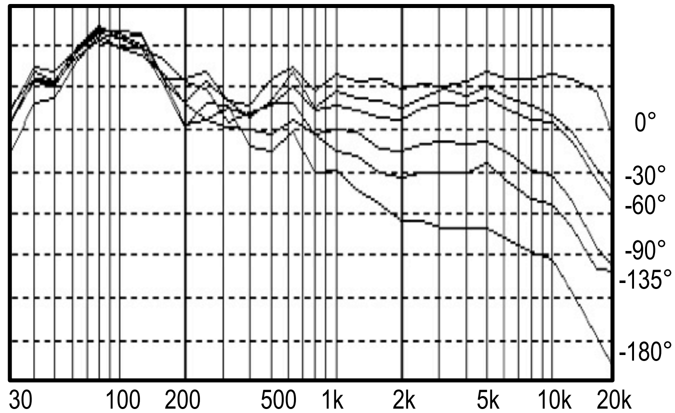


Figure 7. Horizontal radiation of Infinity P362, omni, by angle, outdoors, cabinet upright on ground.

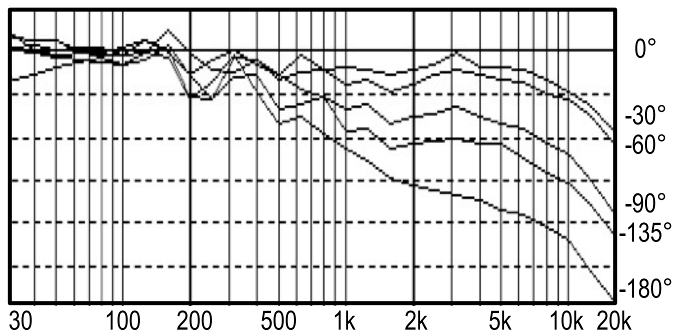


Figure 8. Normalized horizontal radiation of Infinity P362, omni, by angle, outdoors, cabinet on ground.

Graphed normalization simply makes visually clearer what the ears perceive: that constant directivity is a very good characteristic, because it permits mostly proper registration of the spatial information within the recording.

And this Infinity tower is an inexpensive loudspeaker, available now in its successor incarnation, the P363, $200 each as of this writing.

Good work, Harman, Olive, Toole, et alia!

Now a question is, for me anyway: Why was this fine design sonically clobbered, to my ear in my unrigorous comparison, by a 20+-year-old design with ultrawide-dispersion drivers? Figure 9, an old normalized measurement of the Allison AL125 horizontal radiation taken the same way as for Figure 8, holds the answer: extremely wide upper-midrange and treble radiation. It is perhaps not quite as smoothly uniform by angle, but shows much louder sound heading out into the room, reflecting off the sidewalls and front corners. Deliciously airy on jazz trio, and in side by side mono with the TV sports announcers.

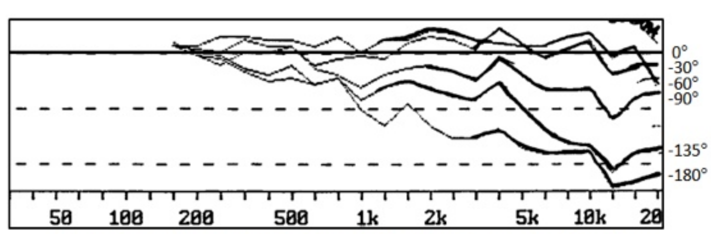


Figure 9. Normalized horizontal radiation of Allison AL125, standing, omni, by angle, outdoors, cabinet on ground.

Photo 2. Allison AL125 (grille removed).

Whether or not such spacious playback quality is desirable, can such dispersion be achieved using conventional tweeters and midranges and baffle widths? No. But from the topline Snell decades ago to recent topline Boston Acoustics and Revel (another Harman company) models, designers have occasionally stuck a tweeter on the cabinet back to just that end. Doing so helps overcome conventional P362-style radiation pattern and comparatively dimensionally flat sound.

I hope that this examination of an inexpensive, fine-performing, preference-test-winning Harman design, plus an earlier fine-sounding alternative, proves stimulating to loudspeaker aficionados. I urge all who are interested in speaker design variables to immerse themselves in the sea of scholarly and scientific detail in Floyd Toole's essential magnum opus, *Sound Reproduction: The Acoustics and Psychoacoustics of Loudspeakers and Rooms*, available from the usual places.

December 2012 Meeting — Auditioning and Examining BSO Broadcast Streams

by John S. Allen and Stephen H. Owades (Massachusetts)

The meeting, serendipitously occurring on the 242nd anniversary of Beethoven's birth, was held in member Richard Goldwater's home.

Open forum

David B. Hadaway (DBH): Congress has passed a law that broadcasters have to control the volume of commercials [the Commercial Advertisement Loudness Mitigation CALM) Act (https://EN.Wikipedia.org/wiki/Commercial_Advertisement_Loudness_Mitigation_Act).

Robert L. Miller (RLM): And as a result, they seem to be increasing the volume of the programs.

Joel Goldberg: There has been more than one deadline [for the act to take effect].

RLM: There is also controversy about DVRs that let you skip commercials.

Nick Noiseux (NN): PBS stations [sound levels are lower] than commercial stations.

Stephen H. Owades (SHO): Commercial stations are loud so the ads don't jump out of the programs.

F. Lee Eiseman (FLE; publisher, *Boston Music Intelligencer — BMint*; www.Classical-Scene.com): My son plays music through laptop speakers. What could I get for $100-150 that sounds good?

David R. Moran (DRM): Boston Acoustics, Polk — look on Craigslist.

SHO: The Bose store at Jordan's, or Audyssey speakers (Tomlinson Holman [was involved in their design]. Holman is now at Apple, where he is involved in improving the sound of laptops.

DBH: Old CD-Rs are having problems near the outer edge — they were not rated for the highest burning speed. [I have heard from several members who are having problems reading the original BAS Test CD-1, which were CD-Rs. Contact David Hadaway (dbsys2@att.biz) if this is true of yours. DJW]

SHO: Basically, there are two constraints in writing optical discs: the linear speed of the track passing the laser can't be too high or the pits will be less than cleanly burned, and the rotational speed of the disc can't be too high or the physical media might fly apart. At very high recording speeds, discs are recorded using constant angular velocity (CAV; a constant rpm, which results in faster linear speed toward the outer edge of the disc). At lower speeds, the drive runs at constant linear velocity (CLV), so it spins faster when writing the inner portion of the disc, where the circumference of the track is shorter. As the writing speed increases, the drive mechanism transitions from CLV to CAV to keep the rotational speed at the center of the disc below the danger point. In CAV mode, the linear writing speed in the latter (outer) part of the disc is higher and the likelihood of playback errors is increased. With a disc that isn't recorded all the way out to the edge, the problem doesn't occur, but the rated overall speed is also not achieved. For best results, the solution is to record at the highest speed at which the recorder runs in CLV mode — usually 16x. (The situation is actually a bit more complicated, since modern drives can work in a hybrid "zoned" mode that is safer than pure CAV, but CLV is still the most reliable way to burn discs.)

FLE: Can anyone recommend a good flash-drive audio recorder with balanced inputs? Marantz 661?

RLM: The Korg MR-1 (http://reviews.CNET.com/voice-recorders/korg-mr-1/4505-11314_7-32563619.html) is good. It is discontinued, and needs an external power supply for condenser microphones.

SHO: I have the small Sony PCM-D50, with internal mikes, that has inputs for external analog and digital signals. [Update: Sony has introduced a higher-end model, the PCM-D100, that also supports DSD and 192ksps PCM, but does not have mike preamps or balanced line inputs. SHO]

NN recommends the Fostex.

Goldwater's system

Goldwater's extremely complicated home-theater system has several screens including a 14' front-projection one, a tall equipment rack, and hundreds of cables hanging over a low rack near the door. The audio portion boasts 19.2 channels and includes a Denon receiver that includes Audyssey circuitry.

Goldwater and Owades tried one thing after another, and for a while it was not clear there would be any audio or video display for the meeting feature. There was a problem with connections between Goldwater's new computer, which does not have an internal disc player, and his system. A USB-connected CD drive saved the day.

Richard P. Goldwater (RPG): We're not going to listen to the full system, only to two-channel stereo over the front speakers: Goldstein model 10s.

BSO broadcasts over the years

Moran on how this meeting came to occur: Over the last few years the Boston area has seen two or three new classical-emphasizing arts-review websites, with the premier classical-only one being the *Boston Music Intelligencer*. Publisher F. Lee Eiseman has noted that WCRB has a panoply of ways it makes content available, and asked Moran [who is an assisting editor to *BMint*] if he would like to write about it. "Absolutely not," Moran replied, but did suggest that the BAS had members who can address this in other ways — attending concerts, listening over the radio, etc. BAS volunteers Stephen Owades, David Hadaway and John S. Allen (JSA) took up the challenge, recording the 13 October 2012 concert from 11 different sources (live and delayed), including over-the-air and via Internet streaming.

FLE: One reason to address this is that people have been quite disappointed with discontinuation of the Friday-afternoon broadcast, replacing that with WGBH's offering multiple ways to listen to the same broadcast.

RLM lives in the shadow of the Arlington-Belmont hill and gets a 35-40dBSNR on FM. Listening over the Internet is thus a practical alternative.

DRM: The BAS has a long and productive relationship with WGBH. Also, many in the BAS are working musicians, the premier one being Owades, who has sung in the Boston Symphony's Tanglewood Festival Chorus for decades. "There has been a lot of screaming and moaning from *BMint* readers about the quality of the broadcasts."

RLM works in the BSO archives as a volunteer, frequently dubbing older recordings from the BSO Transcription Trust. He was asked about the technical quality of the source recordings, learning that in the 1960s and '70s it met a high standard. Some people thought they were better than the RCA commercial releases — above and beyond compression and miking issues; these were purist recordings, using two mikes over row D.

E. Brad Meyer (EBM): Even the hall mikes now are not that far from the stage.

FLE: Soloists are now as loud as the orchestra.

EBM: Ben Roe (then Managing Director of Classical Services, WGBH) doesn't think that two spaced omnis are suitable for symphonic music. Discussion with him will be interesting. At an earlier job, Roe listened to tapes coming in over the transom with all kinds of miking. A center mike vs two-channel stereo has always been an issue; Europeans have been cognizant of this.

FLE: Multimiking lets them solve balance issues in postproduction.

EBM: In the 1970s, BSO concerts weren't dynamically compressed, and there was a consistent stream of complaint from most listeners — most people listened via table radios and in cars. Maybe five or six times in a piece, WGBH would reduce level by 5-6dB.

JSA: It appears that look-ahead compression is being used now, because all the peaks come to the same level, but usually without an audible sudden jump in level.

RLM brought a 1974 recording for comparison, but there might not be time to play it. Also, the Berlioz *Les Nuits d'été* can't be played. Old acetate tapes have to be transferred to digital before they are lost; then there is the problem with certain more-recent tapes' polyurethane binder, which is hygroscopic. The tape becomes hard and the binder sheds onto the guides of the tape recorder — "sticky shed". Another problem is now developing with tapes from between those periods.

JSA: Production values have changed. In the 1970s the impression given was that the radio audience was considered uninvited guests. There was no intermission feature: hall microphones were left open through the intermission, and as the musicians began to filter back onto the stage William Pierce would deliver an announcement that was usually cut short by the arrival of the conductor. I considered this horrible showmanship. Now we're at the opposite extreme, with blabbing during intermissions.

FLE: WCRB has one-third the listenership it had before WGBH took over. [Station management is] scared. They had a 3% share even when broadcasting from Lowell; the decline happened later.

SHO: WGBH, with the new talk format, has a much lower rating than WBUR.

Feature: Stephen Owades, comparing BSO broadcast streams

SHO: The 13 October 2012 BSO concert included the Mendelssohn *Violin Concerto in E minor, Op.64* (https://EN.Wikipedia.org/wiki/Violin_Concerto_%28Mendelssohn%29) with Arabella Steinbacher (www.Arabella-Steinbacher.com/), and the Shostakovich *Symphony No. 4 in C minor, Op.43* (https://EN.Wikipedia.org/wiki/Symphony_No._4_%28Shostakovich%29). The conductor was Vladimir Jurowski (many would like him to become the next conductor of the BSO; https://EN.Wikipedia.org/wiki/Vladimir_Jurowski). Three BAS volunteers recorded all of the broadcast versions — live, and when rebroadcast on 21 October 2012: WCRB analog and WCRB-HD (96kbps), plus WGBH-HD2 (48kbps). WGBH has the strongest RF signal in this region, with its high transmitter power grandfathered during the recent FCC modifications to FM transmitter-power allowances. [During the 1960s I received WGBH-FM from time to time in Middlebury, VT, on the far side of the Green Mountains. But does the grandfather clause apply to the recently authorized HD Radio signal? JSA] All versions were recorded to computers using 16- or 24-bit samples.

Owades also captured the 128kbps MP3 Internet stream, live and during rebroadcast, plus the on-demand 128kbps MP3 stream and the 192ksps BSO Concert Channel (www.WGBH.org/995/bsoConcertChannel.cfm) stream.

The comparison included a CD copy of the live two-channel mix direct from the engineering room in Symphony Hall.

WGBH captures all of the microphone signals as fed into the mixing board, storing this multitrack recording on a hard drive, which is archived for future use, as is the live two-channel mix.

WGBH has moved into the basement room originally set up by Deutsche Grammophon with 220Vrms power. The signal has been sent from that room over several types of connections. Now it is sent over the Internet as 16-bit LPCM audio (CD quality samples), which should be aurally transparent. The rebroadcast is taken from the 24-bit/48ksps LPCM stereo master recording made at Symphony Hall. All of these versions have gone through the radio station's mixing board. The remaining versions: an on-demand MP3 stream at 128kbps, and the BSO Concert Channel MP3 at 192kbps, are derived from the Symphony Hall master recording.

Jim Donohue [no longer at WCRB] has been in charge of recordings for these broadcasts for >20 years, and has not changed his philosophy. To prepare the BSO Concert Channel and on-demand versions, he runs the master recording through a Apple Macintosh application called Max, which converts the signal to the formats needed. [It was not clear at meeting time, but the Concert Channel version is actually saved to WGBH's server as a 320kbps MP3; conversion to the 192kbps stream is performed on the fly. The version heard at the meeting was saved directly at 192kbps, and thus is not identical with the actual Concert Channel stream. SHO]

The on-demand version is convenient to listen to. The BSO Concert Channel is inconvenient, because it is a continuous loop of all the concerts from the prior year-long period for which WGBH has rebroadcast permission. (It used to be two weeks of concerts.) The concerts are presented one after another with no information available as to what concert or music is being played, or when a particular concert recording will play. This stream does include the announcements. WGBH's Brian McCreath (www.WGBH.org/listen/mccreath_brian.cfm?) has said that the playbacks are in performance sequence and the entire sequence repeats in a continuous loop. With an application (such as WinAmp) that displays the metadata in the Internet stream, you should be able to determine what is playing. At the time of the BAS meeting the loop was of fewer than 30 concerts — less than 60 hours — but now contains a full year of Symphony Hall and Tanglewood programs. It sounds unlike what goes out over the air.

Donohue asked Owades not to use the Mendelssohn violin concerto performance for the demonstration at this meeting, because Steinbacher was moving around and wasn't where he wanted her to stand relative to the microphones' positions.

WGBH has heard complaints from every angle; there is nothing new that will affect their approach to broadcasting the BSO. They fired Brian Bell, who was punctilious about miking and wanted to be at the Friday concerts so he could anticipate issues for the Saturday broadcasts.

So, all in all, there were 11 broadcast streams: two analog FM, four HD Radio, two real-time streams, on-demand and Concert Channel streams, and a direct CD copy. Due to an interconnect problem, Owades recorded only one channel of the WCRB HD broadcasts, sufficient to make level comparisons but not for comparative listening. Its dynamics were similar to those of the WGBH HD2 broadcast.

Owades displayed a screen image (Slide 1; all slide images are on page 22) of a section of the Mendelssohn violin concerto including a few seconds with no music. This level is -57dB in the broadcast master; 16-bit resolution is ample for this dynamic range. Owades' graphics were generated using iZotope RX 2 Advanced (www.iZotope.com/products/audio/RX/), which shows a waveform envelope (revealing waveform details when expanded) and a frequency spectrum. Owades compared silent-interval levels and peak levels across the 11 broadcast streams.

Levels were identical among the broadcast master, on-demand stream, and BSO Concert Channel. The live Internet stream is very different: compression is not what you think of as decilinear (linear in decibels). The quiet sounds are still quiet; most compression occurs at higher levels. The 21 October rebroadcast Internet stream is even louder; it hit zero — it clipped!

Over-the-air WCRB FM did not show as much compression as WGBH HD2, although HD Radio has a potential dynamic range of 107dB. The compression is intended for people listening in cars.

An HD1 channel and the analog channel should show the same compression, because they carry the same program and radios need to be able to switch between them without changing level. [This is not uniformly so with WCRB. I've noted significant level changes when my tuner switches from one to the other, borne out by Owades' measurements. Also, on many stations the analog and digital streams are not synchronized perfectly, so there will be an echo when they switch. JSA]

An HD2 channel, which carries a different program, has no analog channel it needs to match. When WGBH HD2 first broadcast the BSO concerts, WGBH was quite proud to state that this stream was uncompressed. Whether the HD2 stream is still broadcast with no compression has yet to be determined.

For the graphics and listening comparison, Owades gathered samples of the last few minutes of the Shostakovich Fourth Symphony (Slide 2), with the applause and announcement that followed, and matched applause level.

"The Shostakovich sounds louder than the Mendelssohn, except that the stream reveals that they have hit the limiter, hard."

The dynamics of the on-demand stream (Slide 3) and the BSO Concert Channel (Slide 4) are similar to those of the master recording.

The live stream (Slide 5) is very different from the master.

FLE: Quiet is gated; everything else is hitting the max. When you listen to it without turning it down, it sounds horrendous. When you turn it down, it sounds pleasant. There are a lot of places where people are listening. This is slightly worse than the broadcast.

JSA remarked on the glitches he heard in the rebroadcast recording.

SHO: The 'gritty scritch' (near the start of the Mendelssohn) was in every version including the download, but not in the broadcast master.

JSA: There were a couple of other glitches, the same in both the WCRB FM and WGBH HD2 (Slide 6) rebroadcasts, including a complete dropout for about a half-second when Della Chiesa is naming orchestra members over the applause following the Shostakovich.

The Internet streams all cut off at 15kHz, although that is only necessary for analog stereo FM broadcasting. [Actually, with today's high-quality digital filtering, HD Radio and streaming audio could extend to ~18kHz, unless a station is transmitting a subsidiary communications authority signal (SCA; http://EN.Wikipedia.org/wiki/Subsidiary_Communications_Authority) (depending on FCC regulations). JSA] [But there is virtually no listener to this material capable of detecting even 15kHz. DRM] The WGBH on-demand stream looks "patchy", evidently due to deletion of subaudible frequency bands in the digital data compression. The live web stream exhibits a much narrower dynamic range, apparently by raising midlevel material volume and limiting high levels. The "hump" in the compressor curve is pushed up in the replay web stream, and there is probably some hard clipping.

On WCRB analog (Slide 7), processing is similar to that of the live stream, and no peaks are visible.

Compression is even greater in the WCRB analog replay (Slide 8).

HD Radio suppresses all input signals above 8kHz and regenerates them in the decoder through use of harmonic synthesis (distortion). The WCRB 96ksps stream looks essentially the same as the WGBH HD2 48ksps stream (Slide 9). In both HD Radio versions (Slides 9 and 10), the 8kHz transition from actual to synthesized audio can be seen in these slides, but it's not audibly evident.

JSA noted that polarity appeared to differ for some of the talk segments, and also between the live and rebroadcast recordings, as shown by waveform asymmetry.

There was a 48-second delay between the WCRB and WGBH HD2 signals, with WGBH HD2 heard later. The delay can be quite annoying when switching from one station to the other. This is apparently a feature of the standard software used for preparing HD1 and HD2 broadcasts.

Owades displayed a table (Table 1) showing the relative levels of the various versions.

Measured Levels

Q = silence within first–movement cadenza (2.5 seconds)
A = level of end of movement 1 (last 30 seconds)
B = level of passage linking movements 1 and 2 (16 seconds)

	Mendelssohn end of mvt 1					Shostakovich End		
	Q	*A*	*B*	*A–B*	*B–Q*	*max*	*applause*	*match*
broadcast master	−56.87	−6.55	−28.67	22.12	28.20	−1.17	−13.58	
128kbps download	−56.68	−6.84	−29.03	22.19	27.65	−0.48	−12.94	−0.64
192kbps BSOCC	−57.42	−6.73	−28.92	22.19	28.50	−1.44	−13.77	0.19
live stream	−46.90	−0.31	−3.72	3.41	43.18	−0.31	−0.31	−13.27
replay stream	−39.63	−0.23	−5.12	4.89	34.51	0	0	−13.58
live WCRB-FM	−55.82	−11.58	−27.26	15.68	28.56	−11.18	−12.94	−0.64
replay WCRB-FM	*	−11.69	−25.18	13.49	*	−11.04	−12.86	−0.72
live WGBH-HD2	−40.97	−4.02	−11.72	7.70	29.25	−3.47	−3.77	−9.81
replay WGBH-HD2	−39.41	−2.39	−10.24	7.85	29.17	−1.93	−2.16	−10.11
live WCRB-HD†	−45.82	−3.80	−13.64	9.84	32.18			
replay WCRB-HD	−45.85	−4.14	−12.61	8.47	33.24			

** recording of replay WCRB-FM missed the "Q" passage*
† recordings of live and replay WCRB-HD were missing the left channel

Table 1. Measured levels.

Of note is that levels are substantially the same for the master, the 128kbps download and the 192kbps BSO Concert Channel. The other versions showed effects of compression, particularly at peaks. Compression was similar for the live and replay HD versions on each station, but differed between the stations: WGBH HD2 apparently applying more. Oddly, the WCRB analog stream appears to suffer the least compression of all the live and replay streams, considering that analog stereo FM broadcasting can be noisy if the signal is weak.

Owades concluded the meeting by playing the same excerpts from the Shostakovich with the ensuing applause and announcement, in each of the nine versions in which both channels had been recorded. He asked attendees to take notes. Except for the first version played — the master recording — he did not identify the others until after all had been played.

A table [not reproduced here] records attendee observations.

[Since the meeting, there have been significant changes in the BSO broadcasts, some dictated by new personnel handling this work at WGBH and some inspired by the observations made in this comparison study. Producer Brian McCreath, who has taken over from Brian Bell, is similarly devoted to BSO broadcast quality. At the end of the 2012–'13 season, Jim Donohue was let go by WGBH; broadcast engineering is now handled by Antonio Oliart. Each broadcast now has a single announcer (usually Ron Della Chiesa); the tag team is gone. Various changes at the station facility mean that the signal path for the live broadcast (analog and HD) is cleaner and less subject to processing and compression, but the on-demand and BSO Concert Channel versions are still better than the broadcasts. The delayed rebroadcast of each BSO program now takes place on Monday night instead of Sunday afternoon. All concerts (except for a few with licensing restrictions, such as the *West Side Story* film-with-orchestra performances) are made available after a few days for on-demand and BSO Concert Channel listening. SHO]

Notes about the slide images

SHO: I used the then-current version of iZotope's RX 2 Advanced audio repair software (RX2) to work with the audio selections for this meeting, and to create images to make the differences visually apparent. The main RX2 screen shows two channels of audio (left-channel above, right-channel below). When you open an audio file, the overall length of that file fills the screen from left to right, with a time scale displayed beneath. You can zoom in to whatever region of the file you choose, and the time scale (and some additional numerical information) adjusts accordingly. For this presentation, all the Shostakovich excerpts were about eight minutes long, and that entire span is shown in the screen images. The Mendelssohn excerpts used for dynamic comparisons were about 6:35 long.

RX2 shows two types of information superimposed in the main window. There's a dynamic envelope centered around the negative-infinity level (infinity below digital full-scale); this is shown in a light bluish color and, if you zoom in far enough, it displays the audio waveform in detail. The dynamic scale can be zoomed as well, for examining very soft audio files, but all these images are referenced to digital full-scale. The legend (Figure 0) normally is just to the right of the main window. The left-hand scale shows level in dB from negative-infinity (in the vertical middle) to digital full-scale (at the top and bottom of the legend scale). Plus, there's a spectral analysis in the background, where brighter yellows mean higher levels at a given frequency; the frequency scale is at the far right of the legend.

By looking at these RX2 displays, we can see how dynamic compression affects the different versions of the Shostakovich excerpt, with the WGBH streams having almost none of the dynamic variation shown in the original master. Careful examination of the background spectral-analysis information shows which of the versions have a hard cutoff at 15kHz, and that a subtle change in the pattern above 8kHz is present in the HD Radio recordings (because HD Radio synthesizes all the information above 8kHz).

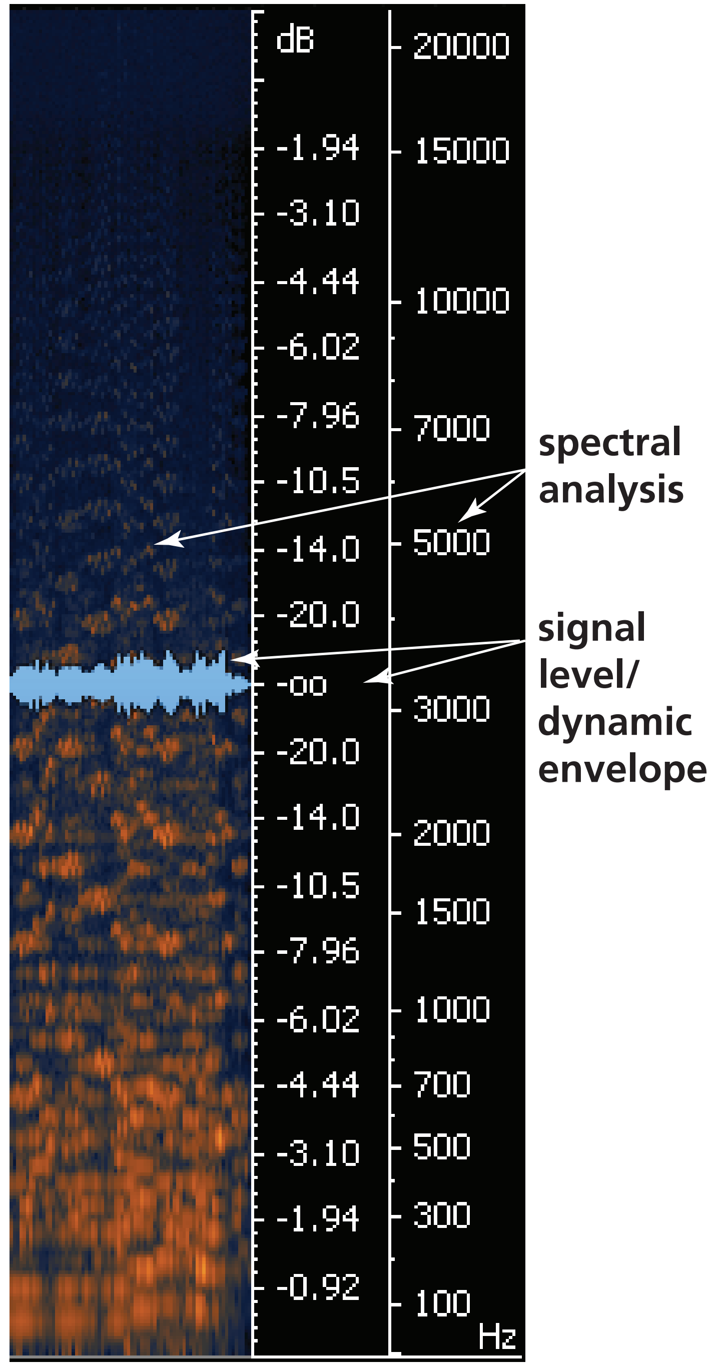
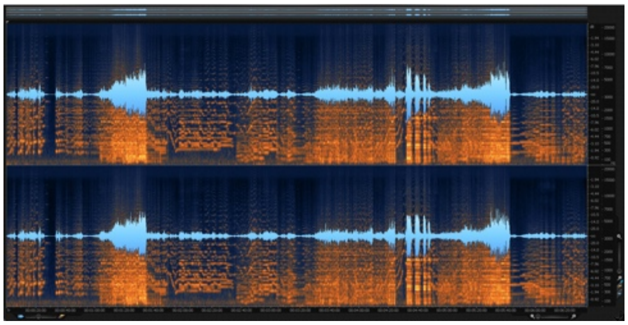
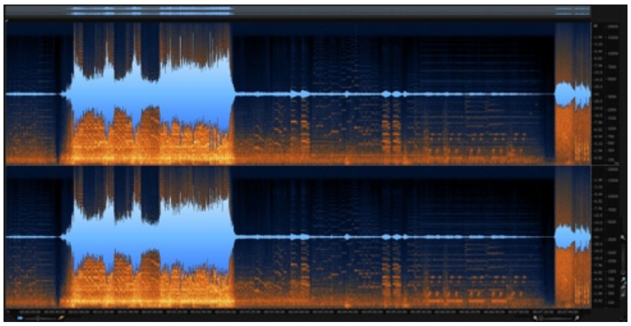


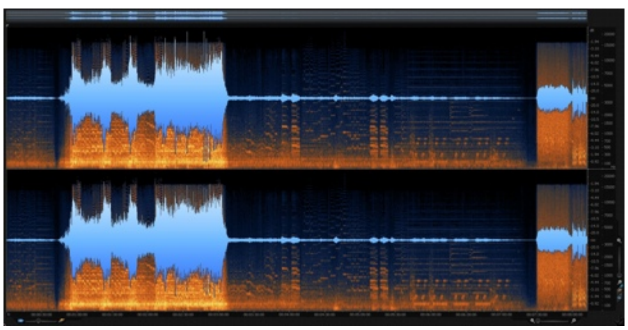
Figure 0: The iZotope RX2 legend, normally at the right of each of the following graphs.



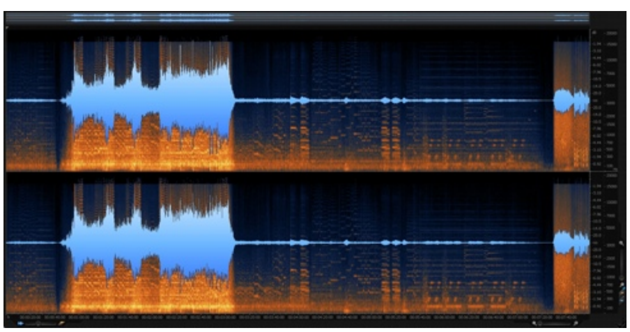
Slide 1. Mendelssohn broadcast master.



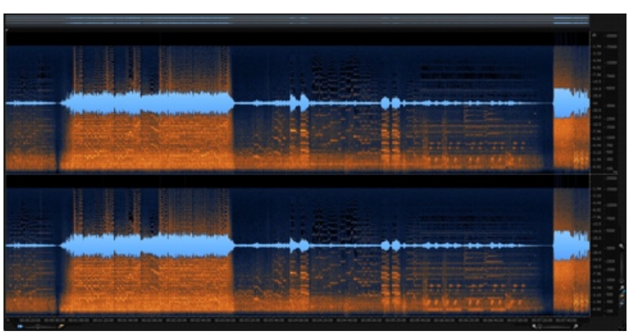
Slide 2. Shostakovich broadcast master.



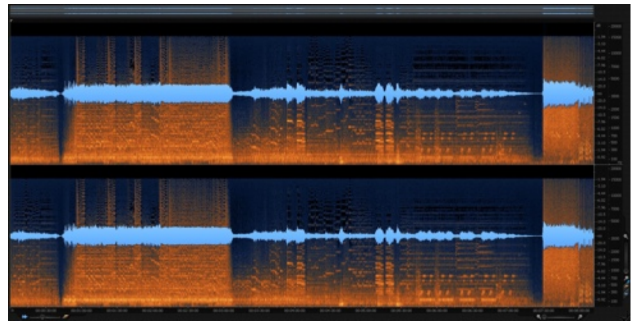
Slide 3. Shostakovich WGBH on-demand stream (128kbps).



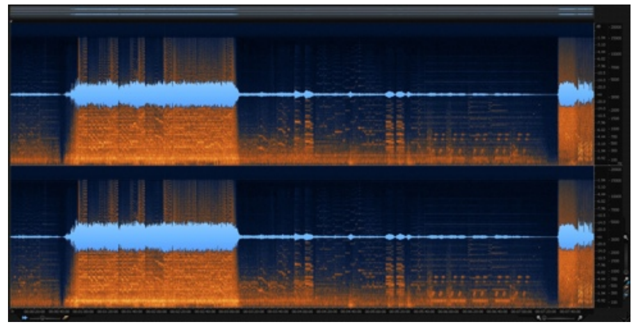
Slide 4. Shostakovich BSO Concert Channel stream (192kbps).



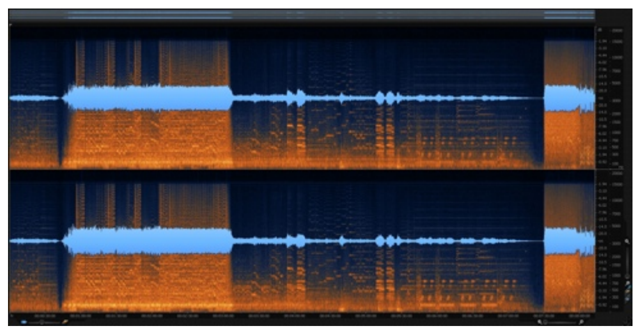
Slide 5. Shostakovich WGBH live web stream (128kbps).



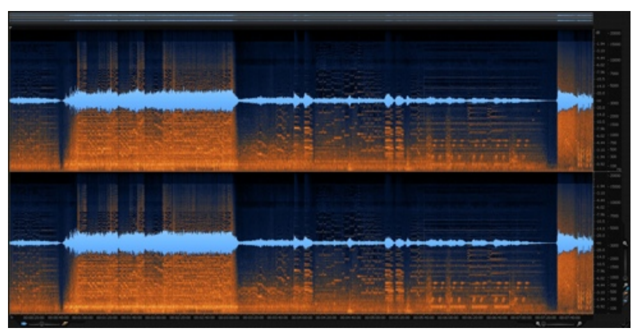
Slide 6. Shostakovich WGBH replay web stream (128kbps).



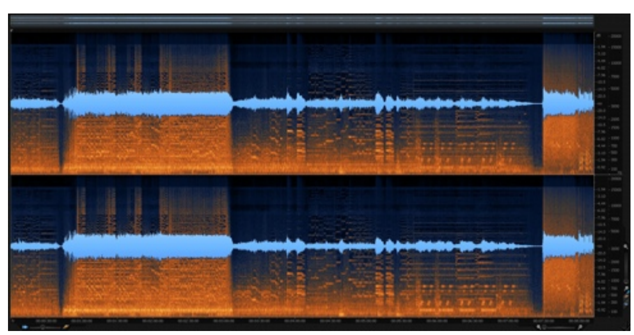
Slide 7. Shostakovich WCRB analog-FM live.



Slide 8. Shostakovich WCRB analog-FM replay.



Slide 9. Shostakovich WGBH HD2 live (48kbps).



Slide 10. Shostakovich WGBH HD2 replay (48kbps).

Classifieds

For sale: Heavy velour stage curtains. Great for sound deadening. David J. Weinberg (301.593.3230; WeinbergDJ@BostonAudioSociety.org).

Free: Used Antennacraft VHF antenna: five-element yagi, channels -7-13. I replaced it with one twice the size. David B. Hadaway dbsys2@ATT.biz.

The Boston Audio Society needs individuals who will write summaries of BAS meetings into *.pages, *.doc, or *.txt files. We usually have audio recordings of the meetings. We pay for your work. **Ask about our bonus program for meeting-writers!** Contact David B. Hadaway (603.899.5121; dbsys2@att.biz).

Discography of Test Records. I am compiling a list of technical test records from all recording eras and would like to exchange information in order to confirm, expand, correct and add entries. Send me your list and/or other information about test records and I will send you my discography, which has nearly 200 entries to date. I am interested in commercially released records, as well as discs intended primarily for in-house use by audio engineers, broadcasters and for other special applications. I am generally excluding demonstration discs of the "Super Stereo bouncing ping-pong ball" type unless they include at least some tracks of a technical nature, such as frequency response and tracking tests, etc. Nicholas Starin, Portland Oregon, NTSTestDisc@Gmail.com.

Nashville AES Test CD. (www.AESNashville.org; click the small round symbol where it says "Click here for more info:" to download the four-page description and track list. Use the link on the website under <Products> to send an email for pricing and availability.) From the website: "Assembled 'by engineers, for engineers' after a survey of the needs of a broad cross-section of working audio pros, the disc features 71 tracks of test signals ranging from level calibration tones, noise signals, third-octave tones, polarity and digital meter calibration pulses, band-passed noise and swept sinewaves. Additional tracks include dry-recorded kick and snare drums, acoustic guitar and piano along with spoken word and a cappella male and female vocals. A data area on the disc features BPM charts, Excel-based calculators and other bonus material. The CD booklet includes tutorial information on using the test disc."

For Sale: Insignia model NS-HDTUNE standalone digital HD/FM/AM tuner. Works well. No remote. Has 16 FM presets. Offers very clear HD Radio reception. The LCD display shows the frequency, signal strength, station call letters, artist name, and song title. Has digital coax and Toslink outputs as well as analog stereo output. Modified with larger heatsinks on the voltage regulators. $35. Nicolas Noiseux (NNoiseux@Tiac.net).

Seeking cartridge shell (black plastic) for AR Turntable. Does not have to be an original (in case any other company made a replacement). Will pay generously for one in excellent condition. Doug Pomeroy, Audio Restoration and Mastering Services. 718.855.2650; AudioFixer@Verizon.net.

Seeking a two-channel preamplifier such as an Apt Holman, an early Hafler, or an early NAD. Must work. Fred Ampel (FAmpel@KC.RR.com).

For sale by Randy Brown (RBGrid2@Hotmail.com; Surprise, NY — ~30 miles south of Albany):

- **Two McIntosh MC-30 amplifiers** that I modified for more output by replacing the 5U4 rectifier tube with two solid state diodes, plus adding a negative-temperature-coefficient thermistor for soft turnon (I understand the same changes were made by McIntosh Labs when switching from a tube to a solid state power supply). The amps clip symmetrically at ~45W. Discuss delivery/pickup. Make offer.
- **Two Altec Lansing 604D coaxial 15" speakers, drivers-only**, with 604-8G LF and HF diaphragms, reconed ~30 years ago, but not used and look new. Included is a pair of 604-8G passive crossovers. Discuss delivery/pickup. Make offer.

Still seeking prerecorded Advent cassettes, especially Russell Sherman Beethoven and Liszt recordings. David R. Moran (DRMoran@aol.com). (Looking for less printthrough, if possible, than the sets I very generously received from a BAS oldtimer.)

For sale by Michael Riggs (Michael@RiggsNet.com; buyer pays UPS shipping (US only) or may pick up in central New Jersey):

- **Adcom GFA-2535L** four-channel power amp in original box with manual. Rated at 60Wpc into 8 ohms, 90Wpc into 4 ohms; one pair of channels can be bridged for 200 watts into 8 ohms. Top panel is scratched, but the amp works fine. One channel hums faintly, just audible to me in a silent room, but that was present from the beginning. Price: $85.
- **NAD C740** stereo receiver in original box with remote and manual. Excellent condition. Price: $125.
- **Yamaha CDC-765** five-disc CD changer in original box with remote and manual. Excellent condition. Price: $50.

For sale: Several CD storage racks, some made by Bostonwood (http://BostonWood.com/cd.html): Two four-shelf units (25.5"Hx17"Wx5.5"D); One four-shelf unit (25.5"Hx24"Wx5.5"D); Two three-shelf units (19.5"Hx17"Wx6.5"D); one homemade three-shelf unit (18"Hx14"Wx6"D). I have attached a handle and chains to several units, making them convenient open-top carriers. David J. Weinberg (Maryland; 301.593.3230).

For sale: One Denon DR-M34HR cassette deck ($500srp) with three heads and live monitoring of the recording, dual Dolby B/C and Dolby HX Pro circuitry. Handles normal, chrome and metal-tape cassettes. Highly regarded in 1988. Gasparo Records used them as duplicators. In good shape. Data at www.VintageCassette.com/denon/dr-m34hr. $125ea+shipping. Contact David B. Hadaway. dbsys2@att.biz.

For sale: Gently used Arcam DV29 DVD/CD player (www.Arcam.co.uk/advice-and-support/discontinued-products/DV29DVD Player; and www.UpscaleAudio.com/products/Arcam-FMJ-DV%252d29.html; $3000srp; sell for $800+s/h). Wire transfer or Western Union only; no checks, no PayPal. Fred Ampel (FAmpel@KC.RR.com).

For sale: Gently used Arcam AVR 360 receiver (www.Arcam.co.uk/products,fmj,av-amplifiers,AVR360.htm; $1800srp; sell for $600+s/h). Wire transfer or Western Union only; no checks, no PayPal. Fred Ampel (FAmpel@KC.RR.com).

For Sale: Pair of mint condition Verity Audio Parsifal Ovation monitors (www.VerityAudio.com/en/index.php/products/parsifal, but without the bass module; $8900srp). Includes pair of custom color-matched maple speaker stands (cost me $1500). I have original ATA cases and documentation. Will sell complete set for $5000, plus shipping or local Boston-area pickup. Edward Gonzalez (617.742.0146; Edward.Gonzalez@RCN.com).

For sale: Two Tannoy HPD 315 speakers, made in the mid-1970s, woofers might need refoaming. Steven Weiner (SDWeiner@ATT.net; 818.865.0601; Thousand Oaks, CA) is asking $500+s/h. [I have seen a photo; the cabinets and grill look to be close to mint condition. DJW]

Own a movie theater sound system! Top-of-the-line High Performance Stereo HPS-4000-XL screen speaker systems. Three (left, center, right) HPS model 545-4 speaker systems (www.HPS4000.com/pages/fivefortyfive_.html) and two HPS model 545-W subwoofer speaker systems (www.HPS4000.com/pages/545_w_.html). All in perfect working order. The speakers are not light, but consist of modules that can be lifted by two moderately strong men. Truck delivery only, or come pick up. Make me an offer. David J. Weinberg (301.593.3230; WeinbergDJ@BostonAudioSociety.org).

Laserdiscs for sale by David J. Weinberg (301.593.3230; WeinbergDJ@BostonAudioSociety.org) Four have Dolby Digital 5.1 soundtrack. Nine have DTS5.1 soundtrack. Make offer for all or selected items.

Title	CLV/CAV?	Aspect Ratio
Video Essentials		
Apocalypse Now	CLV	2
Apollo 13	CLV	2.35
Batman (Michael Keaton)	CLV	1.78
Blade Runner (Director's Cut)	CAV	2.48
Blown Away	CLV/CAV	2.35
Crimson Tide	CLV	2.35
Daylight	CLV	1.85
Evita	CLV	2.35
Forbidden Planet	CAV	2.35
Goldeneye	CLV/CAV	2.35
The Grifters		
Highlander (10th Anniversary Director's Cut)	CLV/CAV	1.82
The Hunt for Red October	CLV	2.35
In the Mouth of Madness	CLV	2.35
Jurassic Park	CLV	1.85
Jurassic Park	CAV	1.85
Phenomenon	CLV	2.35
Pocahontas	CLV	1.66
Pretty Woman	CLV	1.85
Raiders of the Lost Ark	CLV	2.35
Raiders of the Lost Ark	CAV	1.33
Star Wars: Episode 1 - A Phantom Menace		
Toy Story	CLV	1.78
Who Framed Roger Rabbit	CLV	1.75

For sale by David J. Weinberg (301.593.3230; WeinbergDJ@BostonAudioSociety.org) — most units listed come with user manuals (some also with service manuals), remotes and cartons, as applicable. Make offer:

- **ReplayTV 2020** with aftermarket larger disk drive. Includes lifetime electronic program guide service.
- **Faroudja VP400A video quadrupler.** Upscales NTSC video to 960 progressively scanned lines for a front or rear CRT projector. Generates great image on my big screen. I don't need it because I'm modifying my video system.
- **Runco 980Ultra** CRT projector (original packing, user and service manuals).
- **Vutec concave high-gain silvered screen.** 8' wide, 16:9 aspect ratio, with wallmount brackets. Silvered surface makes it appropriate for 3D projection, as the high gain overcomes the substantial light loss caused by the various consumer 3D projection technologies, and the curved shape can deliver a wide viewing angle with little hotspotting.
- **Lexicon DC1** surround processor, with all options.
- **Panasonic RP-91D** DVD player with JVB Digital regionfree mod.
- **Sony DVP-NS755V** SACD (multichannel and stereo), DVD, etc., progressive-scan output.
- **Tektronix D11** rackmount single-beam storage oscilloscope with 5A15N vertical amp and 5B10N time-base/amplifier.
- **Tektronix 5111** rackmount single-beam storage oscilloscope with two 5A18 vertical amps and 5B10N time-base/amplifier.
- **Teac AN-180** external dual Dolby B processor. Simultaneous stereo record and playback.
- **Pioneer CT-F4242 cassette deck**, in excellent condition.
- **Garrard RC88 turntable with Shure M3D cartridge**, in excellent condition.
- **Sony XM-5046 car audio amp**, 4x50W into 4Ω.

•

Your Thoughts

Could Have Been Here !

Send them to me at

WeinbergDJ@BostonAudioSociety.org

The Audiophile Voice

POBox 43537, Upper Montclair, New Jersey 07043

Eugene Pitts, III — Editor/Publisher
AudiophileVoice@Verizon.net
973.778.4882

Special Offer for Boston Audio Society Members

Dear Audiophile:

Before I set off my own built-in B.S. detector, let me cut straight to the chase. I'm offering you a good deal on a subscription to ***The Audiophile Voice*** because you are a strong hi-fi hobbyist. And why shouldn't audio club members get a good deal on subscriptions to all the hi-fi magazines, the publications you most want to read? After all, you are the guys we publishers most want as readers. While I can't offer you a deal on *all* the magazines, I can offer you one on mine. But please understand, I'm too cheap to send you a long series of expensive direct-mail packages, and this offer won't get any better. So act now.

"*So, why is this a good deal, Pitts?"* you ask. It's this way: With printing and postage, each issue of ***TAV*** costs me about $1.75. But I will still send you six issues for $12.00, that's $2.00 per copy. And you can order two or even three years. But act now, because there won't be a cheaper offer later on.

I believe you will find ***The Audiophile Voice*** to be the most entertaining, readable, and certainly the most attractive magazine in this field. Better still, it's written and edited by long-time lovers of audio gear, like you . . . for audiophiles, like you. There's little or no jargon, no cult of personality, you won't need an engineering degree, and you don't have to put up with arrogant insiders. No nonsense, just good reading.

Don't believe me? Well, just email or snailmail your name and address to me and ask for my "no risk" free sample issue.

"Why don't you use a long series of direct-mail letters like other publishers?" Because the few additional subscriptions those publishers get don't offset the large additional cost of the mailings. Besides, I've never believed in pushing subs just to blow out ad rates. That's anti-high-end. And, yeah, there is a high-end in publishing, too, and the Big **S**_________ magazine is no longer part of it. And Harry's doing his Rapunzel Act. **A**_____ committed suicide in public. So if you want to get the hottest magazine in this field—the only indie magazine that is written "by audiophiles, for audiophiles"—then subscribe now. I might even let you use my B.S. detector.

Most Cordially,

Gene Pitts

Please send your check or money order in U.S. funds only; sorry, we don't take credit cards.
These policies help keep our rates fair to both of us.

Mail to: *The Audiophile Voice*, POBox 43537, Upper Montclair, NJ 07043

How Long?

( ) One year, 6 issues, $12.00 **( ) Three years, 18 issues, $36.00**

( ) Two years, 12 issues, $24.00 **( ) Four years, 24 issues, $48.00**

Name ____

Company ____

Street Address ______________________________ **Apt. No.** ____________

City ______________ **State** ____ **Country** _________ **ZIP or Postal Code** ____________

The Boston Audio Society

Membership Application 2014

***** Please fill in all fields *****

1. Name (please print all fields) ____

Address ____

City, state, zip, country if not USA ____

Optional (BAS use only!):

Occupation ________________________ Phone no(s) ________________________

Email ____

2. Reason for application:

Renewal _____ New member _____ Address change _____

3. BAS members receive a volume of the *Speaker* [check your preference(s)]:

____ electronically (as a *.pdf file) or the ____ printed edition.

Plus, for Boston-local zip codes, monthly meeting notices (check one: ____ emailed or ____ printed).

*** Please realize that printing and mailing are significant expenses for the B A S). ***

4. Dues

_____ $40 or more **Membership** (overseas** $70; includes *The B A S Speaker* by airmail);
Note: Basic dues covers only a portion of publication costs.

_____ $65 and above **Supporting** membership
(includes the BAS Test/Demo CD#1 or a *BASS* CD*, if you want it).

Please choose one:
___ Don't send the CD *or* ___ Send the BAS Test/Demo CD-1 *or* ___ Send a *BASS* CD*.

_____ $30 each **Also, Order** a *BASS* CD* (coincident with new/renewal membership).

* There are three *BASS* CDs to chose among. Select the one(s) you are ordering:
____ *BASS*v1-10 CD-R: Complete contents of *The B A S Speaker* volumes 1-10 in *.pdf format.
____ *BASS*v11-20 CD-R: Complete contents of *The B A S Speaker* volumes 11-20 in *.pdf format.
____ *BASS*v21-30 CD-R: Complete contents of *The B A S Speaker* volumes 21-30 in *.pdf format.

All checks must be in $US and made payable to the **Boston Audio Society**.
foreign checks ***must be from banks with US affiliates*.

Dues and/or contributions may also be paid via PayPal (online) or with MC, Visa, AMEX, or Discover by:
sending email to: dbSys2@att.biz *or* writing to: DB Systems; POBox 460; Rindge, NH 03461 USA.

Samples and back issues

Published issues of *The B A S Speaker* contain a trove of audio and other information. While there have been some 35 volumes since 1972, we suggest that new members consider acquiring volumes 17 on. For contents, refer to www.BostonAudioSociety.org.

Originally printed volumes 6-16 and 18 are complete, and we have some individual issues.
Copies of the entire 40-year opus are available, and will cost $0.20/page, plus $3s/h.
Volumes 1-30 are available on CD-R (see * above).

5. Send this form with your check to:

The B A S POBox 260211 Boston, Massachusetts 02126 USA

or go to www.BostonAudioSociety.org/membership.htm

Thank you.

BAS

THE BOSTON AUDIO SOCIETY
P O Box 260211
Boston, MA 02126

Address service requested

FIRST CLASS