Please also see references [49] and [50] on pg. 409 for more information and extensions of this work including time-domain measurements.

Cable pathways between audio components can affect perceived sound quality

Journal of the Audio Engineering Society, Volume 69 Issue 6 pp. 398-409, June 1st 2021

Milind N. Kunchur *(kunchur@gmail.com, kunchur@sc.edu)*

University of South Carolina, Columbia, SC 29208, U.S.A.

The arena of highest fidelity in music reproduction, sometimes referred to as high-end audio, has many controversial claims and contentious issues. One such controversy is whether the cables and topology used to interlink components together make an audible difference. There seems to be a disparity between anecdotal experiences reported by audiophiles and published formal scientific research, as to what are the minimal changes in system configuration that can be audibly distinguished. With the motivation of bridging this divide—which may originate from differences in instrumentation and subject-listening conditions used by the two groups—this work utilized a high-performance audio system and an extended-duration listening protocol that more closely resembles audiophile auditioning conditions. With these measures, the present work was able to prove through direct psychoacoustic testing that two different analog-interconnect pathways can be audibly distinguished.

0 INTRODUCTION

Audio recordings and their reproduction serve a wide range of needs and purposes ranging from monophonic voice recorders to multi-channel surround-sound systems to various specialized professional audio applications. Within this audio universe, two-channel stereophonic (stereo) systems tend to dominate sound reproduction for the purpose of music. The uppermost strata in stereo-system fidelity[1] are sometimes referred to as "high-end audio" (which I will abbreviate as HEA).

A digital HEA system typically consists of the following separate components interlinked together by cables: (1) A source such as a CD (compact disk) player or separate transport/server and DAC (digital-to-analog converter), (2) an amplifier, and (3) speakers (loudspeakers) optimally placed in a suitable listening room[2]. Some of these components involve complex designs and materials (e.g., atomic clocks or diamond diaphragms) and are offered at staggering prices. Furthermore, HEA systems require meticulous setup and attention to detail (e.g., precise speaker positioning for best acoustic coupling with the room, use of spikes to prevent speaker recoil, etc.). To those without experience in HEA, these measures can easily seem extreme and superfluous, especially since there is very little psychoacoustic research proving that any of these steps or components (other than speakers) makes an audible difference [3]–[5].

HEA level components other than speakers are already close to perfection in certain specifications, such as frequency response and harmonic and intermodulation distortions, that are commonly used to evaluate consumer audio performance[3]. For example, the amplifier used in the present work has a frequency response of DC to 1 MHz (±1 dB) and distortion of ~0.01% (total harmonic plus

[1] This work is mainly concerned with the chain of electronics between the source and speakers—the amplifier/s and various cables—for which perfect fidelity implies identical output and input waveforms (aside from gain). The term *distortion* is used in this work in the general sense (except where specified) to mean any alteration in the waveform. Subjectively, these alterations will be perceived through changes in spatialization (imaging) and in timbre (tonal quality). Timbre along with pitch, duration, and loudness is one of the four principal attributes that differentiate musical sounds [1]–[2]. Digital audio has no difficulty preserving pitch and duration (no tape-speed variations to worry about). Thus for two systems playing at equal sound pressure levels, subjective differences in sound quality correspond to differences in timbre.

[2] Additionally, there may be a preamplifier that provides input selection and volume control, a division of the amplifier into left and right monoblocks, external word clocks, etc.

[3] As discussed in other works, subtler forms of signal alterations, especially those associated with the time domain, may play a role at higher fidelity [6]–[14].

intermodulation). This makes it difficult to objectively predict what or if sonic differences can be expected, and understandably further propagates the skepticism that surrounds HEA.

This skepticism can hamper psychoacoustic research into HEA issues through a circular reasoning fallacy. An audio system is a chain whose strength is limited by its weakest link: *every* component and interconnection must have sufficient fidelity for *any* component to make a difference. If an experimenter testing a HEA claim is dismissive of any of the required measures, he/she could end up assembling a compromised system that is unable to resolve the difference being tested, which in turn confirms the skeptic's initial bias that the claimed difference is inaudible. For example, if a noisier amplifier had been used in the present work, its own noise might have masked the cable-noise difference. Similarly, in the system used here, the speakers were placed a considerable distance from walls in an acoustically well damped room; in a typical stereo setup, where speakers are placed <1 m from hard surfaces, the sound would have been adulterated by reflections making subtle differences difficult to detect.

A second potential pitfall of listening experiments probing into HEA claims, is that different conditions and protocols may be used by audiophiles, compared to researchers in psychoacoustics. The latter group often uses a method that I will refer to as *short-segment comparison* (SSC) in which short stimuli, A and B, are compared one after the other in various combinations. The rationale for SSC is that the stimuli will coexist simultaneously in short-term (working) memory (STM) so that the first stimulus does not have to be recalled from longer-term memory for comparing with the second one. On the other hand, in HEA, comparisons between two configurations of a system (e.g., replacement of a cable or component) utilize what I will refer to as *extended multiple-pass* (EMP) listening that may take place over weeks.

Two common stimuli-judging sequences are AB/BA (the stimulus is always actually changed and subjects are tasked with judging the randomized order) and AA/AB (the first stimulus is fixed and the second is randomly chosen to be repeated or changed, which the subjects judge). The risk of the AA/AB sequence is that the second A in an AA combination can sound different from the first because of variations in hearing, leading to a false positive. By comparing AB with BA, a listener knows the two are definitely different and needs to simply make a relative comparison to judge the order. HEA auditioning relies mainly on the AB/BA type of sequence. In the present work, a rigorous and effective EMP blind procedure was developed for comparing slightly different audio-system configurations. Furthermore, some insight is offered, based on psychology and other published results, to explain why EMP might be expected to be more effective than SSC in some situations[4].

So where does the science stand in the matter of cables and their audibility? There have been various investigations into the possibility of audibility of cable related effects using digital signal processing to simulate the sounds or theoretical modelling [16]–[18]. The complexities and subtleties of grounding configurations and their effects on cable performance and the overall system have been discussed in [19]–[21]. And the relative merits of balanced versus single-ended cabling schemes are reviewed in [22]. Nevertheless, to the author's knowledge, proofs of audibility through direct listening comparisons where the cable pathway was actually changed (i.e., not simulations or theoretical estimations) have not been previously published.

This paper is organized into these main sections: (1) 'Equipment and setup' (audio system and acoustical environment), (2) 'Psychoacoustics' (blind listening tests and their analyses), and (3) 'Electrical characterization' of the cables.

I EQUIPMENT AND SETUP

1.1 Audio System Description

Fig. 1 shows a block diagram of the audio system used in this work. The main components (text boxes) are interlinked by various types of cables[5] (arrows). The amplifier was plugged directly into a grounded wall outlet. The server and DAC were plugged into a Tice Audio[6] Microblock power conditioner (which provides an isolation transformer to prevent ground loops, plus surge suppression to protect the components).

[4] There are various listening protocol types for evaluating discriminability between distinct stimuli A and B, such as the forced-choice paired-comparison used here or ABX (with the task of judging whether X = A or X = B). Regardless of the protocol type, the label SSC is used here when stimuli durations are of the order of seconds or less (e.g., [15]) so that there is essentially only one sonic feature the subject is listening for (see detailed discussion of EMP below). Also there are testing protocols where the subject can control the volume level and duration, and go back and forth between the choices. This flexibility is not compatible with the present experimental setup's minimalist signal chain, which avoids intervening switch boxes and cables (due to their unknown additional signal alterations). Also subjects not having an understanding of SSC versus EMP, and of auditory fatigue, may pick durations that are too short and volumes that are too high.

[5] In audiophile jargon, the term "cables" is sometimes reserved for just the "speaker cables", with "wires" used as the collective term. However, in a general electrical-engineering sense, the interconnects and power cords are also "cables" and not "wires", since they contain multiple mutually insulated conductors with different functions/signals.

[6] Tice Audio Products Inc., Jupiter, Florida, U.S.A.

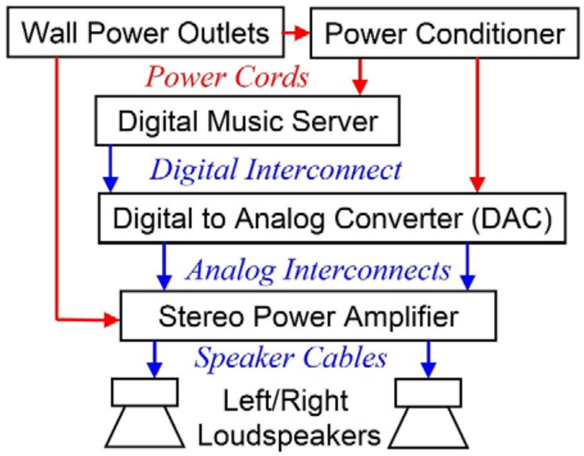

Fig. 1. Block diagram of the audio system used for the experiments. The two configurations (A and B) compared in the listening trials differed only in the choice of the analog interconnects.

The music (media) server was a Bryston BDP-1 Digital Player[7]. This fed a 1.5 m Straight-Wire[8] Infolink digital interconnect through an AES/EBU XLR connector. The digital feed went to a Berkeley Audio Design[9] Alpha DAC Series 2. This DAC has two isolated buffered pairs of analog outputs[10]—single ended RCA and balanced XLR—which were always active and fed the two interconnects that were compared in the listening trials: (A) a Straight-Wire Virtuoso higher-end (retail price ~\$500 for 0.5 m) 0.5 m long balanced XLR-to-XLR cable with polytetrafluoroethylene insulation and (B) a Monster-Cable[11] Interlink 400 entry-level (retail price ~\$50 for 2 m) 2 m long RCA-to-RCA cable with polyethylene insulation. They were both continuously connected to a class-A solid-state Spectral DMA-250S Studio Universal Amplifier[12], which has balanced and single-ended switch-selectable inputs. This arrangement avoids the need to disconnect and reconnect interconnects during trials, and avoids intervening external switch boxes and additional cables and circuit paths, which might compromise fidelity[13]. The signal levels for the two configurations were exactly matched within ± 0.0045 dB (using an AC-voltage measurement at the speaker terminals). The amplifier's own noise levels were equal within ±0.004 mV (measured using the same instrumentation as for the cable noise as described below); by comparison, the noise difference between cables was a hundred times greater (~1 mV).

The amplifier's output was bi-wired to a pair of 2-way-monitor ProAc Response D2 loudspeakers[14] through 3 m long Straight-Wire Maestro II speaker cables. These cables were terminated with optimized RC networks (R=18 Ω in series with C=3.3 nF at the tweeter terminals and R=18 Ω in series with C=5 nF at the woofer terminals) to suppress reflections back toward the amplifier. The speakers were mounted on Target[15] HJ20/T stands with lead filled columns, and with spikes on the top and bottom plates to suppress recoil.

Speaker distances (from front centers) were: 1.88 m to the back wall, 1.69 m to side walls, and spaced 2.08 m left to right. The room has a 56 m^2 area with 1.37 aspect ratio, and 2.7 m ceiling height. The back wall was lined with 1.22 m tall air-spaced rock-wool panels (average thickness of 245 mm) fronted by Roomtune[16] reflective/absorptive aluminum-sheet/glass-wool panels and louvres of wooden strips. The floor is covered with very dense and thick nylon carpeting atop thick padding, with a double layer of carpeting (total thickness of ~40 mm) over 2 m^2 areas where the first reflections from the floor occur. As a result, the room has well controlled acoustical characteristics: At the listening position the 60 dB reverberation decay time is RT60 = 0.30 s with a negligible reverberant intensity I_R—

[7] Bryston Ltd., Peterborough, Ontario, Canada. This server's master-clock jitter is specified at <20 ps (standard deviation) over 10 Hz–10 MHz. The jitter level at the server's AES/EBU output (which includes the effect of transmitter impedance and datalink bandwidth) was measured in [23] to be 637 ps (peak) over 50 Hz–100 kHz.
[8] Straight Wire Inc., Hollywood, Florida, U.S.A.
[9] Berkeley Audio Design LLC, El Cerrito, California, U.S.A. This DAC's total distortion (all products) at the present output level is specified at -120 dB of full scale.
[10] This DAC's absolute voltages at the left and right RCA outputs and the XLR inverted and non-inverted outputs (pins 3 and 2 relative to pin 1) are equal within a standard deviation of 0.13% (i.e., 0.011 dB). The six voltages were measured at the pins while playing a -12 dB (25.12% of full scale) 1 kHz sinusoidal tone from a 16 bit, 44.1 kHz wave file.
[11] Monster Products Inc., Brisbane, California, U.S.A.
[12] Spectral Audio Inc., Mountain View, California, U.S.A. This amplifier has a dual-monaural design whose specifications are (per channel into 8Ω): 200W continuous rms power; 60 A peak output current; DC to 1.8 MHz ±3dB, DC to 150 kHz ±0.1dB frequency response; 0.009% static distortion at 150W rms (DC–100 kHz); 0.01% dynamic distortion (8-tone cluster test at 20 kHz); 400 ns rise time; 1.5 μs (–40 dB) settling time; 600 V/μs slew rate; 100 kΩ//100 pF input impedance; 97 dB unweighted and 107 dB ASA-A signal-to-noise ratio.
[13] HEA systems tend to be minimalist, as every additional link in the chain potentially adds unintended noise and distortions, weakening the overall performance. For example, the DAC used here doesn't have a power-on/off switch and is connected directly to the amplifier without an intervening preamplifier. (A phono preamplifier following a turntable cartridge is an exception to this rule, because it serves the essential purpose of equalizing the frequency response and boosting the feeble level.)
[14] ProAc Limited, Brackley, Northamptonshire, U.K. This speaker's specifications are: 8Ω impedance; 25mm air-cooled silk-dome tweeter; 165mm glass-fiber-weave woofer with Excel magnet system and copper phase plug; 30 Hz to 30 kHz frequency response; 88.5 dB for 1W at 1m sensitivity; 11 kg mass; and dimensions of 430mm height, 203mm width, and 260mm depth.
[15] Target Audio Products, Aurora, Ontario, Canada.
[16] https://www.michaelgreenaudio.net/

providing for a relatively unadulterated direct sound (acoustical measurements are described below). A photograph of the audio system is provided in Fig. 1 of [24].

In the hope that some audible difference will be detected, the cables in the two configurations were chosen to be as different as possible. With its much shorter length, balanced (versus single-ended) topology, and faster dielectric, the higher-end cable A can be expected to have a more detailed and accurate sound. The question was whether interchanging these interconnects would produce a recognizable and memorable timbral change that would be discernable in blind listening tests.

1.2 Acoustical Measurements of the Listening Room

The standard reference used for determining absolute sound pressure levels was a calibrated ACO Pacific[17] model 7016 measurement microphone (with a sensitivity of 3.27 mV/Pa) coupled with its accompanying model 4012 preamplifier (with a 40 dB gain stage) feeding a LeCroy[18] model LT342 500 MHz digital storage oscilloscope. The frequency response of this measurement chain was flat (±3 dB) from 4 Hz to 120 kHz.

To evaluate the listening room's reverberation decay time, white noise playing through the system's speakers was abruptly cut off while the sound intensity[19] I was measured using a Samson[20] Meteor microphone connected by USB (universal serial bus) to a laptop computer running the Audacity[21] recorder software application. Fig. 2(a) shows this I vs t. From the inverse semi-logarithmic slope $\tau = dt/d(\ln I)$ of the exponential decay (slanted line) one finds RT60 = 13.8 τ = 0.30 s.

To assess the relative proportions of the direct intensity I_D and the reverberant intensity I_R, the total $I = I_D + I_R$ was measured for different distances from the right speaker (continuously playing white noise) along a straight path crossing the listening location ("sweet spot"). The setup was the same as for RT60, except that I was measured using the SPL (sound pressure level) meter application within the REW (Room Equalization Wizard) software suite[22]. Reverberation and other classes of reflected sound modify the direct sound's inverse-square law ($I_d = \alpha/r^2$) to a total of $I = \alpha/r^2 + \beta$ (where r is the distance, and α and β are positive constants). Fig. 2(b) shows hardly any deviation from the inverse-square law: the measured I=4.90 μW/m^2 and linear fit (to I α 1/r^2) predicted I= 4.93 μW/m^2 at the sweet spot differ by < 1%, indicating negligible contamination by reflected sounds ($I_R \ll I_D$).

The procedures and analyses for these acoustical measurements follow [25]. Note that measuring I_R in addition to RT60 provides more information about the room's acoustics. Everything else being the same, a smaller room will have a shorter RT60 but may have high reflected intensity (say from a hard floor), which will interfere with the direct sound and stereo imaging [24].

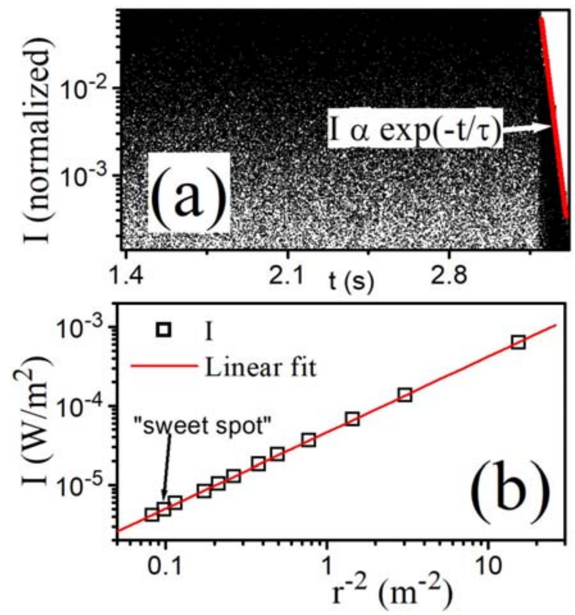

Fig.2 (a) The reverberant intensity decays exponentially after the white noise is turned off. The graph shows about 400000 samples and each spot represents one sample.
(b) The measured intensity closely tracks the inverse-square law indicating the low contamination of the direct sound by room reflections.

2 PSYCHOACOUSTICS

2.1 Listening for Subtle Differences

It takes a considerable amount of time to form a robust and detailed impression of an audio system's sound quality, because of the human *perceptual bandwidth* [26]–[27] that limits the rate at which information can be absorbed and processed mentally[23]. For this reason, audiophiles often audition a component using a variety of music recordings for numerous days, because of variations in a listener's mental and physical state and consequent dissimilitude in subjective assessment from day to day.

It is worth further exploring why the standard psychoacoustics SSC approach (which works well for discrimination of sound levels or pitches) might

[17] ACO Pacific, Inc., Belmont, California, U.S.A.
[18] Teledyne LeCroy, Chestnut Ridge, New York, U.S.A.
[19] $I \propto V^2$, where V is the microphone voltage.
[20] Samson Technologies, Hicksville, New York, U.S.A. This LDC (large diaphragm condenser) microphone has a 25 mm diameter diaphragm and 20 Hz to 20 kHz frequency response. It was calibrated against the ACO Pacific microphone, but for the present measurements the absolute level is immaterial.
[21] Audacity, https://www.audacityteam.org/
[22] REW, https://www.roomeqwizard.com/. I= $10^{[L/10]}$ in pW/m^2, where L is the sound level in dB.
[23] This concept has also been studied in the context of vision, where the idea of "ensembles and summary statistics" are used to explain the apparent contradiction between the richly detailed subjective impression and the limitations of cognitive mechanisms such as attention and working memory [28]–[30].

underperform in discriminating subtle audio distortions. Naïve intuition might lead one to expect SSC to be more productive because the first sound will still be maintained in STM through activity in the auditory cortex, hippocampus, and frontal cortex [31], when the second sound is played. Two speculations are offered here—one is related to overlap in auditory short-term working memory and the other is related to the length of the segment providing more examples for the distortion.

STM can maintain several different items at one time: If you are shown a blue circle, red triangle, and green square, after 30 seconds you can still recollect that the circle was blue and the triangle was red. Here the items are clearly very different from each other and get categorized into different "slots" in STM and each is kept alive. In the case of two slightly different audio configurations, the sounds are so similar that instead of being stored as different items in STM, they may go into the same slot. Then the second sound may overwrite the first item, thus vanquishing distinguishability. Hence, playing B too soon after A, may actually yield a worse discernment instead of better.

A second possibility is that listeners can better discern a certain distortion with longer segments because they have many more instances of this distortion occurring in different contexts, thereby forming a more robust category. Consider the following analogy: When you first glimpse a new face, you may forget its details almost immediately, perhaps remembering only the gender. With repeated exposures, you progressively improve your ability to recognize that person. At first you might mistake the face if you see it from a different angle or if the person changes their hairstyle or is seen in a different context (e.g., in a store instead of a classroom). However, the fallibility of recognition diminishes as you get to know the person better. Similarly, it takes time to form a firm and definitive opinion about the qualities of sound, and to develop a mental sense of the underlying distortions in an audio system.

Overall, the brain appears to cumulatively gather and retain fragments of information to progressively synthesize an increasingly detailed picture. This cumulative process might potentially lead to another complication in listening comparisons that may especially affect the very first (virgin) trial. If the lower resolution configuration B is followed by the higher resolution configuration A, listeners may be more likely to judge that A is more detailed by noticing the extra revealed information. However, when A is played first, the brain may remember its more detailed sound and fill in the details when B is played later making it harder to discern the cable interchange (this should be especially true for musicians). Such filling in of missing information based on memory/expectation also occurs in speech recognition as the well-known *phonemic restoration* process [32]–[33] and occurs in tone perception as the *illusory continuity* effect [34]. The measurements of the present work show a suggestive hint of this hypothesized effect.

The above observations explain why a blind testing procedure that quickly switches back and forth between A and B might be less effective than one that uses extended segments as in the present work. The latter allows the listener to "see the distortion from many angles" to notice an unmistakable (or less mistakable) pattern.

2.2 Blind Listening Trials

The author completed a Collaborative Institutional Training Initiative (Citi Program) course (Completion Date: 29-Apr-2018; Expiration Date: 28-Apr-2021; Curriculum Group: Human Research; Course Learner Group: Social & Behavioral Researchers). The research proposal for this study was reviewed on 03-May-2018 by the Office of Research Compliance of the University of South Carolina Institutional Review Board (IRB). All experiments were conducted in compliance with the reviewed proposal.

As explained in the "Introduction" section, HEA level components tend to already be extremely good in specifications and subjective (sighted) sonic performance. As a result, blind tests proving audible distinguishability between components (even major components like level-matched amplifiers and CD players, not just cables) tend to fail (these negative results don't get published in journals). It was therefore necessary to keep an open mind and think beyond standard psychoacoustic protocols to develop an effective method.

To this end, two concrete steps were taken. One was the research described in the previous section into factors that might affect blind testing effectiveness. This indicated the likelier success of EMP versus SSC protocols. Secondly, preliminary exploratory experiments sought feedback and suggestions from an "advisory panel" of 5 preliminary listeners[24] to assist establishing the design and protocol for the formal listening tests. During these informal listening sessions, which listeners participated in independently, the slow EMP process did appear to be more sensitive than the quick SSC process: Various durations of test sounds (ranging from a single note to 20 minutes) were tried, and it appeared nearly impossible to tell a difference when a single piano note was compared on different system configurations. Three listeners found even 20-second long music segments too short, and two musicians indicated needing at least 5-10 min to form a reliable impression of the sound's quality.

During this pilot testing, one listener also expressed that playing the second sound immediately (within 5 s) after the first "caused their memories to overlap" and suggested having a short break or diversion between the two sounds (like the palate cleansing step in food and wine tasting). All listeners felt that both

[24] These five individuals were University of South Carolina graduate students or undergraduate seniors with whom the author was acquainted, with ages ranging 22–28 years (mean was 25 and median was 26 years), 1 female and 4 males. All were musicians, with an average training of 6 years (standard deviation of 2.5 years) and having given public/stage performances (2 were members of the university symphony orchestra). The instruments they played were trombone, guitar, violin, viola, cello, and double bass. These individuals had no ownership or experience with HEA.

configurations start sounding the same after repeated trials because of acclimatization and loss of attention. Hence it seemed best to allow listeners to participate in at most 5 trials per session per day, to avoid fatigue and psychological pressure. Based on these observations, the final design for the formal experiment was to play 5:40 m:s segments for each configuration separated by 40 s for each trial, with a 4 min break between trials (details of the music are given below). This allowed comfortably completing the target of a 3-trial session per subject well within an hour. In addition, listeners must be instructed to not form a quick opinion as soon as the music began, but to listen to the entire segment to avoid forming an early bias that might skew expectation.

Three preliminary listeners independently suggested providing a set of descriptions/adjectives, so listeners know what to look for in their assessment of the sound. Since non-audiophiles are not familiar with audiophile vocabulary (e.g., liquid, reticent, imaging specificity, low-level detail, etc.), a pool of subjective descriptions was gathered from the preliminary listeners themselves (none of whom were conversant in HEA). From this pool, adjectives that seemed popular and subjectively meaningful were selected by the author to form the following standardized lists for the two configurations[25]:

Interconnect A: Subjective frequency extremes are emphasized. Piano is more noticeable and better matched/balanced with the violin. Violin is more detailed, allowing hearing the change in bow direction. Sound may be dry or edgy.

Interconnect B: Subjective midrange is emphasized. Violin may seem more dominant than the piano, and a touch softer and fuller.

Not all of these adjectives will mean something to everyone, but this approach gives the subjects more than just "clear" or "dull" to latch onto, and provides more examples representing the distortion in support of the cognitive processes discussed earlier. The two configurations were played in opposed pairs: either A/B (i.e., A followed by B) or B/A (B followed by A) chosen randomly. The experimenter had no advance knowledge of the random selections, which prevented unintentional communication with the subjects and provided a double blind condition. The subjects' task was to match the aforementioned standardized subjective adjectives to determine the order (A/B or B/A) of each random pair.

The formal experiment included 18 new listeners (who were not part of the group of 5 preliminary listeners who assisted with the exploratory testing and experimental design) in order to have fresh inexperienced ears. The group consisted of 11 females and 7 males, and 7 musicians[26] and 11 non-musicians, in the age range 20–23 years (median age 21 years, mean age 21.3 years) who were undergraduate students at the University of South Carolina. No academic credit or remuneration was provided. None of these subjects had any prior experience with HEA. Nor had they heard this particular musical piece before. Subjects received no training or practice. Subjects were not informed about the title or label of the recording, nor provided any information about the interconnect cables being tested (which were always invisible from their view).

The recording played was the first 5:40 m:s of the first track[27] of the CD "Delius: Three Sonatas for Violin & Piano" (CD 4012) by the label "Connoisseur society" recorded in 1974. At the listening position, the average A-weighted level[28] for the piece was 60.2 dB-A SPL, and the peak levels were 70.9 and 74.4 dB-A for slow and fast response settings respectively.

2.3 Results and Statistical Analyses

Table I shows the results. The goal was to have each subject undergo a 3-trial session which lasted less than an hour. The subjects were allowed to stop participation and leave at any time. Subjects N, O, and P volunteered to perform one or two additional trials in the same session. Subject P volunteered to participate in a second session on another day. Subjects Q and R completed only two trials. Thus a total of 59 blind trials were conducted of which 43 were judged correctly and 16 were incorrect, the statistical significance of which is calculated below.

In psychoacoustics, sufficiency of statistics is not a matter of subjective opinion and is also not represented by the intuitive "percentage of correct responses". Rather, in a distribution of possible outcomes, one looks at the probability p for occurrence by random chance, with a p value of 0.05 (for a right-tailed distribution) commonly taken as the standard threshold; thus a score of 3 out of 3 may equal 100% but fails because there is a >5% probability of this happening by random chance[29].

The *Wilcoxon signed-ranks* (WSR) test is a nonparametric test that does not assume any specific distribution of the data [36]. It can compare paired data sets

[25] In principle, one could keep researching this approach to further optimize the set of adjectives; but the sets used were effective enough for the subsequent fresh listeners to succeed in blind testing. Here are some other examples of spontaneous descriptions from listeners (unknowingly comparing cable A relative to B played in random order): "real life versus CD"; "felt performers were in the same room, piano was more prominent"; "scratchier, can hear more imperfections, more info"; "I felt I could hear each key pressed and each note played. I could distinguish between the 2 instrumental sounds more easily"; "really enjoyed the clearer sound"; "sharper violin at the end of the notes, almost like it hit a peak right before the next tone."; "sounded more like live sound".

[26] At least 2 years of musical training. All of them still regularly played musical instruments, which include: piano, guitar, violin, cello, and drums.

[27] Wave-file statistics for the (left, right) channels are: maximum level = (63.75, 59.21) % of full scale; rms power = (5.16, 5.17) % or (-25.74, -25.73) dB of full scale; zero crossings = (828.31, 827.92) Hz.

[28] This was measured by playing a white-noise file through the system with the same effective power as the rms power of the music file (see preceding footnote). The instrumentation for acoustical measurements is described in an earlier section.

[29] It has been argued that these standards, although well surpassed in the present work, are overly stringent [35].

and also compare a single set of observations with an expected median value M that satisfies a null hypothesis. Here we compare column F (fraction of correct responses) in Table I with M=0.5 (if subjects guessed randomly). For audibly indistinguishable configurations, the null hypothesis predicts that the subjects' F values will be symmetrically distributed above and below M, so that the differences (column D), will have equal negative and positive "ranks". The asymmetry's statistical significance is then obtained from a WSR table [36], [37] using the number of reduced (non-zero) observations N=16 and W_{min}=5.5, which is the smaller of W-=5.5 and W+= 130.5 (sums of negative and positive ranks respectively); this procedure gives $p < 0.0005$.

The above results include all trials conducted. If we exclude the extra trials by subjects N, O, and P, and P's second session, and exclude the 2-trial sessions by Q and R, then we uniformly have only 3-trial first sessions for each of 16 subjects. This gives N=16 and W_{min}=6.5, which again produces $p < 0.0005$ from the WSR table. Regardless of whether the data is selected or taken in its entirety, the experiment's statistics are over 100 times more stringent than the required $p < 0.05$.

S	M	T1	T2	T3	T4	T5	F	D	SR
A	1	1	1	1			1	0.5	14.5
B	0	1	0	1			0.67	0.17	5.5
C	0	1	0	1			0.67	0.17	5.5
D	0	1	1	1			1	0.5	14.5
E	1	1	1	0			0.67	0.17	5.5
F	1	1	0	1			0.67	0.17	5.5
G	0	1	0	1			0.67	0.17	5.5
H	0	1	0	1			0.67	0.17	5.5
I	0	1	1	1			1	0.5	14.5
J	0	0	0	1			0.33	-0.17	-5.5
K	0	0	1	1			0.67	0.17	5.5
L	0	1	0	1			0.67	0.17	5.5
M	1	0	1	1			0.67	0.17	5.5
N	0	1	0	1	1		0.75	0.25	11.5
O	0	1	1	1	1		1	0.5	14.5
P	1	1	1	0			0.75	0.25	11.5
P	1	0	1	1	1	1			
Q	1	0	1				0.5	0	
R	1	0	1				0.5	0	

Table I: Results of blind listening tests. Column headings refer to: S=subject label; M=1 for musicians; T1–T5 are trial numbers; F=fraction of correct responses; D=F-0.5; SR=signed rank for non-zero values. (F for subject P combines both sessions.)

We can check if the subjects improve with experience in subsequent trials by conducting Wilcoxon paired comparisons (on the same 3-trial first-session data set) between columns T1 and T2 and between T2 and T3, which give N=9 with W_{min}=10 and N=10 with W_{min}=11 respectively. The WSR table indicates that these W_{min} values exceed their respective thresholds of 8 and 10 for p=0.05 (two-tailed value). Thus there is no evidence of either systematic improvement nor worsening in performance with progression of trials.

Yet another way to analyze the results is to consider the question of whether anyone in any group can discern a difference in the audio configurations. Then the 18-subject group can be viewed as a "team" in a fixed-effects analysis. Here one is neither trying to prove that certain specific individuals are especially perceptive nor drawing a conclusion about the population. We are only testing the discrimination performance of this specific sample. The question is simply whether this team treated as a "single subject" can obtain a winning score. Then one can perform a *chi-squared* test on the total score of 43 correct responses out of 59, which gives a value of (for 1 degree of freedom): $\chi^2 = (C - T/2)2/(T/2) + (I - T/2)2/(T/2) = 12.4$, where T is the total number of trials, C is the number of correct judgments, and I is the number of incorrect judgments. This corresponds to p = 0.00044, which again is over 100 times more stringent than the p = 0.05 standard[30]. It is worth noting that just the non-musicians taken as a group, have a passing score of p=0.0041 for their 26 out of 35 correct responses.

The data provide an additional suggestive hint. Of the 18 virgin trials, 8 were B/A and 10 were A/B, which scored 8 out of 8, and 5 out of 10, respectively. Of the 5-out-of-10 score for the A/B sequences, non-musicians scored 4 out of 6, and musicians scored worse with 1 out of 4. This is consistent with the "phonemic-restoration" like process discussed earlier: musicians should be better able to remember and reconstruct missing details thus masking the deficits in the second interconnect and making it harder to distinguish. Although not statistically conclusive in the present work (with p = 0.32 and 0.41 for the two cases) these suggestive observations are noted here for the purpose of instigating future research.

3 ELECTRICAL CHARACTERIZATION

If two audio configurations are audibly distinguishable, then physical differences between their signals must necessarily exist. However, measuring them and interpreting their relevance for audio performance is challenging because of the difficulties in matching the extraordinary capabilities of the ear, and because of our incomplete understanding of auditory neurophysiology. For example, the ear has a 120 dB dynamic range and the sensitivity to detect a cochlear basilar-membrane amplitude of ~1 pm [39]–[41]. It was not the primary goal of the present work to pinpoint the exact physical reason/s for why the interconnects sound different, just that they do. Nevertheless, in the electrical measurements presented below, the sonically superior higher-end interconnect did perform better in all of these measurements.

[30] In other terminology, this corresponds to a certainty >99.95%; and in the language of post-hoc statistical power analysis [36], [38], the total data corresponds to a statistical power of 96.2%, i.e. β= 0.038, taking a nominal α= p= 0.05.

3.1 Frequency response

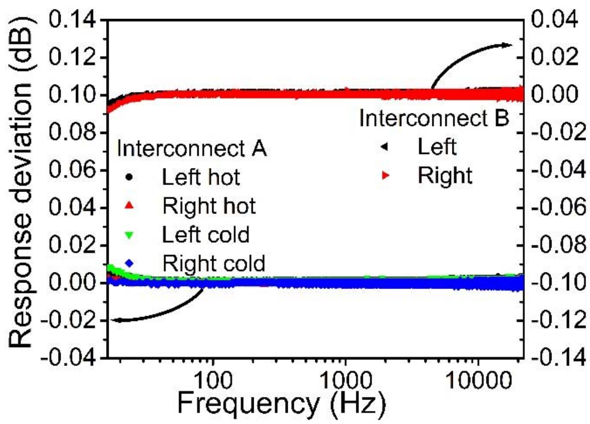

Fig. 3: Frequency responses of interconnect cables A and B shown as deviations from the 1 kHz value. Relative phase varies by less than ±0.12° for A and ±0.18° for B. Responses were measured using the Room-equalization-wizard system taking the average of 8 frequency sweeps each scanning the 2 Hz to 24 kHz range in 65530 steps.

Fig. 3 shows the measured frequency responses of the two interconnects. Although interconnect A's response is more nearly constant, both cables have deviations that are within ±0.005 dB over the 16 Hz–22 kHz range, which are expected to be subliminal [42]–[43]. Thus the difference in sonic performance does not seem to be related to their frequency responses. The balanced interconnect A has two conductors—non-inverted (hot) and inverted (cold)—within a grounded shield. Each was measured separately and is shown for both left and right cables in Fig. 3.

3.2 Electrical Parameters

Table II shows the measured electrical parameters averaged over left and right channels (and averaged over the hot and cold conductors in the case of the balanced interconnect A). All parameters are significantly superior for interconnect A, as are the characteristic decay times. It should be noted that resistive losses in these audiophile interconnects are too small to be relevant—with a worst case attenuation of only $1.48\text{x}10^{-4}$ dB (for interconnect B). Are the decay times—the longest of which is only ~30 ns—subliminal? Audiophiles sometimes view cables as "tone controls", thinking that their effect on timbre is the result of fine changes in frequency response. This notion is wrong. First of all, at the level of HEA, time-domain effects rather than spectral alterations are more influential on sound quality [6]–[14]. Secondly, the nominal reactances are too small to produce audible spectral changes. With the cable's low-pass filtering mainly controlled by the capacitive decay time τ_C=RC, in the worst case, this leads to a subliminal attenuation of only $\Delta L \approx 20 \log(1+2\pi fRC) = 0.033$ dB at f=20 kHz, in agreement with the flatness of the measured frequency response shown in Fig. 3.

Cable	R (mΩ)	L (nH)	C(pF)	Z (Ω)
A	32.2±1.3	112.5±2.5	84.6±1.4	36.5
B	171.5±5	1265±10	295±4	65.5
Cable	τ_L (ps)	τ_C (ns)	T_{LC} (ns)	P_L (μB)
A	11.3	8.5	4.9	2.8
B	127	29.5	30.4	14.8

Table II: Measured electrical parameters: resistance (R), inductance (L), and capacitance (C) and calculated quantities: characteristic impedance $Z= (L/C)^{1/2}$, inductive decay time $\tau_L=L/R_{load}$, capacitive decay time $\tau_C = CR_{source}$, self-oscillation period $T_{LC}=2\pi(LC)^{1/2}$, and resistive power loss $P_L = 2x10^6 \log[(R_{cable}+R_{source}+R_{load})/(R_{source}+R_{load})]$ in microbels (0.00001 dB) with R_{source}=100 Ω and R_{load}=10 kΩ. Measurements were made with an MCH® 2811c LCR meter using four-terminal (Kelvin) sensing and found to be independent of its excitation frequencies (100 Hz, 1 kHz, and 10 kHz) to within ~1%.

The temporal discriminability τ of the human auditory system is much finer than one might infer from the upper audibility limit of f_{max} <18 kHz [44], and not directly related to it. Previous experiments [45]–[46] set an upper bound of τ ~ 5 μs. But it should be noted that those listening trials used an SSC protocol with a very simple form of stimulus (7 kHz square wave tone) and therefore may have overestimated τ. The experience of the present work suggests repeating those experiments with music, rather than a tone, and following an EMP approach to determine a more accurate (and probably shorter) estimate of τ. Similarly, the theoretical value for τ, which was estimated to be as low as 2 μs from neurophysiological modeling [45], was also based on SSC and is potentially an overestimate. Furthermore, there may be possible exotic time-domain effects that prolong the decay beyond the nominal decay times calculated here, which are based on idealized reactive behavior. These questions are worth revisiting in future.

3.3 Noise Susceptibility

Fig. 4 shows the total full bandwidth noise picked up by each interconnect, with one end terminated into 100 Ω (corresponding to the source output impedance) and the other end connected to an oscilloscope. As can be seen, interconnect A has dramatically lower noise than B, almost down to the intrinsic noise floor. The absolute noise level of interconnect B, at 2.24 mV rms (average of both channels), is only 25.1 dB below the rms value of the music that was played in the listening tests. This RF noise will undergo rectification-demodulation upon entering the amplifier and can be expected to contribute an audible signature to the music playback.

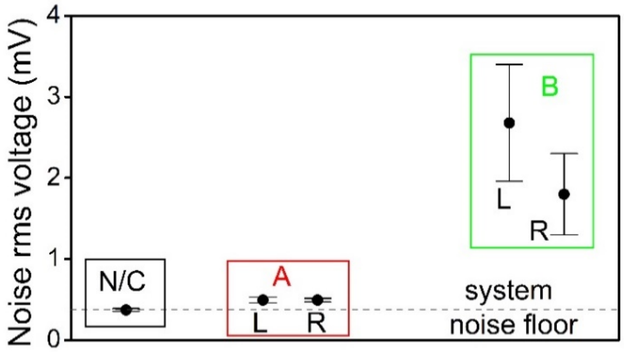

Fig. 4: Noise pickup in interconnect cables A and B ("N/C" is with no interconnect connected) showing mean and standard deviation of 11 measurements for left (L) and right (R) cables. Each rms value was calculated over 250000 digital samples of a trace obtained with a LeCroy (Teledyne LeCroy, Chestnut Ridge, New York, U.S.A.) LT342 oscilloscope (set at 5 ms/div horizontal time scale, 5 mV/div vertical voltage scale, 1 MΩ DC coupling with no probes used, sampling at 5 MHz with a 500 MHz analog bandwidth). For the balanced interconnect, the hot and cold conductors were connected to channels 1 and 2 respectively, and differenced internally.

Fig. 5 shows the spectra of the cable noise depicted in Fig. 4. As can be seen for cable B, the source of the noise is mostly radio frequency interference (RFI), which cable A seems to block more effectively. Although it is present in much smaller proportions than RFI, the audible-band (20 Hz – 22 kHz) noise levels for the two cables are somewhat closer but cable A is still better: with rms noise-power ratios of 1:1.42 (for unweighted band-pass) and 1:2.67 (for ITU-R 468 weighting [47]–[48]). This is expected because common electric shielding (braid, foil, metalized plastic, etc.) is mainly effective against RFI and will not screen out lower-frequency magnetic fields (from transformers, power cords, etc.). This type of screening would require magnetic shielding (e.g., mu-metal) and/or a conducting shield that is thicker (> 9 mm for copper) than the skin depth; such measures are usually not employed in commercially available interconnects (at least not in the present cables A and B).

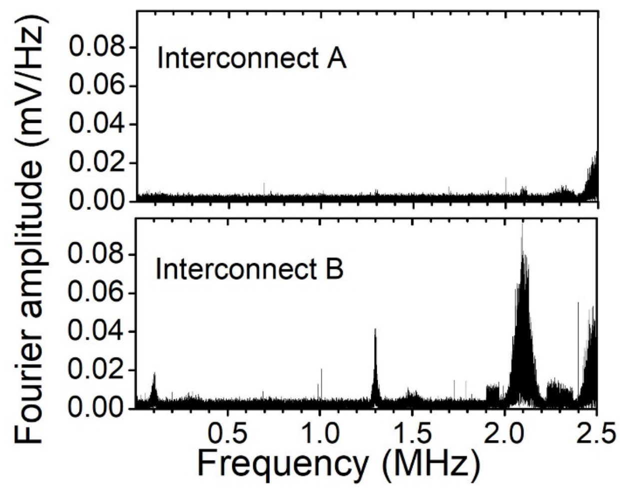

Fig. 5: Noise spectra for interconnect cables A and B. Interconnect B shows significantly more RFI pickup.

To ascertain that differences in the amplifier's own noise levels for the RCA and XLR input paths was negligible compared to the cable-noise differences, the amplifier's output voltage (across an 8 Ω non-inductive resistor) was measured using exactly the same instrumentation and procedure as was used to measure the cable noise; here both cables were disconnected from the inputs and replaced by "shorting plugs"—connectors with 100 Ω resistors between the hot and signal ground pins (i.e., center and shield for RCA and pins 2 to 1 and 3 to 1 for XLR). Referred to the input (the amplifier has a voltage gain of 20 times), the noise voltages were 216 ± 1.4 μV for the XLR and 208 ± 4.2 μV for the RCA inputs[31]. The difference between these is less than a hundredth of the cable-noise difference. Moreover, the absolute amplifier noise level is ten times smaller than the noisier cable B. The situation is probably reversed in budget consumer electronics, with the amplifier likely noisier than the cables. This may provide one explanation for why cables are not commonly noticed to make a difference.

Fig. 6 shows one additional noise assessment, where the amplifier's output was measured with the interconnects connected to its input, with the shorting plugs on their source ends. The amplifier's bandwidth blocks some of the RF noise power from passing through, resulting in smaller differences (113 μV and 127 μV rms, referred to the input, for configurations A and B respectively[32]). Nevertheless, cable A still has the advantage. Also these

[31] In principle, these measurements include the noise of the connectors used as "shorting plugs" and the thermal Johnson-Nyquist noise of their 100 Ω resistors (of which the XLR has two) at the amplifier inputs. But the connectors are impervious shells of thick metal without active electronics and having much smaller dimensions than the cable wires and the amplifier. Also the squared thermal noise voltage given by $V^2 = 4k_BTR\Delta f$ (where k_B=1.38 x 10^{-23} J/K is the Boltzmann constant, T =297 K is the absolute temperature, Δf = 2.5 MHz is the frequency bandwidth, and R=100 Ω) is only V_{rms} = 2.02 μV (which is much less than the ~200 μV measured noise). In any case, the combined noise of connectors plus amplifier (~200 μV) is much smaller than the entry-level cable's 2 mV noise.

[32] These spectra contain smaller total powers, reflective of their narrower frequency range. They were measured with the oscilloscope set to a 25 MHz analog bandwidth and 50 Ω input impedance (instead of 500 MHz and 1 MΩ).

noise measurements were in the absence of a signal, which when present may intermodulate with the noise (i.e., the effect of noise may be more than that of simple masking).

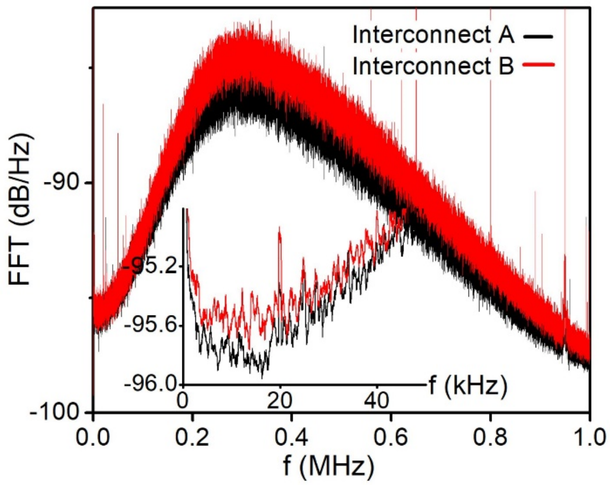

Fig. 6: Noise spectra of the amplifier's output voltage with the cables connected. The FFT (Fast Fourier Transform) was internally calculated and averaged for 100 sweeps within the oscilloscope. For the vertical axes, 0 dB corresponds to 0.315 V peak. The Inset shows a magnified view of the audible-frequency range; inset curves are 25-point moving averages of the main curves.

4 CONCLUSIONS

High-end audio is a subject that is shrouded in controversy. Aside from loudspeakers, consumers exhibit varying degrees of skepticism as to what affects sonic performance. The most contentious ingredient in the chain is the interconnection between components, which concerns both the topology (balanced versus single-ended) and the characteristics of the cable itself. This work shows that two system configurations differing only by the interconnect pathway are audibly discernable, even by average listeners with no special experience in music or audio. To the author's knowledge, this may represent the smallest change in an audio system proven to be discernable through IRB approved blind listening tests.

The success of these experiments depended first on assembling an audio system with sufficient fidelity to avoid masking the minute differences being auditioned. Secondly, the approach to designing blind listening tests was scrutinized to see what might improve sensitivity. An extended multiple pass (EMP) listening protocol was developed, because preliminary experimentation along with other published observations [22]–[23] indicated that it would be more likely to form a robust and detailed impression of a HEA system's sound quality compared to a short-segment comparison (SSC) method.

This work did not conduct an exhaustive determination of all possible physical causes of sonic differences in interconnects. For example, time-domain effects such as reflections were not studied because a balanced cable requires a differential amplifier and extra cable (both adding their own noise and distortions) before an oscilloscope. However, the electrical measurements conducted here indicate that noise levels may be one determining factor of sonic performance. The measurements also show that characteristics such as resistance and frequency response, that naïve consumers may focus on, are irrelevant for distinguishing HEA interconnect cables.

A worthwhile future extension of this work, would be to develop high-performance instrumentation that can cleanly switch between two single-ended interconnects. This will allow assessing sonic differences arising from cables' transmission characteristics that are unrelated to topology, and also facilitate the study of time-domain effects.

Post-publication notes: A follow-up in-depth electrical study of interconnects, which includes time-domain behavior, is presented in [49]. The AES Journal Forum for the present article [50] provides some additional details and clarifications.

5 ACKNOWLEDGMENT

The author gives a special thanks to Prof. Rutvik H. Desai for providing invaluable help and clarity on various issues related to memory, perception, and psychoacoustics, and to Prof. David Mott for discussions related to the neuroscience aspects of auditory memory. The author acknowledges other useful discussions with Prof. James M. Knight, Dr. Charles L. Dean, Prof. Grigory Simin, Prof. Reginald Bain, and Mr. Stacy D. Varner, and also thanks all four reviewers for their valuable input and the subjects who kindly volunteered for the listening tests.

6 REFERENCES

[1] T. D. Rossing, F. R. Moore, and P. A. Wheeler, *The Science of Sound*, 3rd ed. (Addison Wesley, Boston, MA, 2002). ISBN 0-8053-8565-7.
[2] R. L. Pratt and P. E. Doak, "A SubjectiveRating Scale for Timbre," *J. Sound Vib.*, vol. 45, no. 3, pp. 317–328 (1976 Apr.). http://dx.doi.org/10.1016/0022-460X(76)90391-6.
[3] S. P. Lipshitz and J. Vanderkooy, "The Great Debate: Subjective Evaluation," *J. Audio Eng. Soc.*, vol. 29, no. 7/8, pp. 482–491 (1981 Aug.). http://www.aes.org/e-lib/browse.cfm?elib=3899.
[4] S. P. Lipshitz, "The Great Debate: Some Reflections Ten Years Later," in *Proceedings of the 8th International Conference: The Sound of Audio: The Sound of Audio* (1990 May), paper 8-016.
[5] F. E. Toole, "Listening Tests—Turning Opinion Into Fact," *J. Audio Eng. Soc.*, vol. 30, no. 6, pp. 431–445 (1982 Jun.).

[6] H. R. E. van Maanen, "Temporal Decay: A Useful Tool for the Characterization of Resolution of Audio Systems?" presented at the *94th Convention of the Audio Engineering Society* (1993 Mar.), paper 3480.
[7] H. R. E. van Maanen, "Requirements for Loudspeakers and Headphones in the 'High Resolution Audio' Era," in *Proceedings of the 51stInternational Conference: Loudspeakers and Headphones* (2013 Aug.), paper 1-3.
[8] H. R. E. van Maanen, "Is Feedback the Miracle Cure for High-End Audio?" (2017 May). https://www.temporalcoherence.nl/cms/images/docs/FeedbackHvM.pdf
[9] W. Woszczyk, "Physical and Perceptual Considerations for High-Resolution Audio," presented at the *115th Convention of the Audio Engineering Society* (2003 Oct.), paper 5931.
[10] N. Thiele, "Phase Considerations in Loudspeaker Systems," presented at the *110th Convention of the Audio Engineering Society* (2001 May), paper 5307.
[11] J. R. Stuart, "Coding for High-Resolution Audio Systems," *J. Audio Eng. Soc.*, vol. 52, no. 3, pp. 117–144 (2004 Mar.).
[12] J. R. Stuart and P. Craven, "A Hierarchical Approach to Archiving and Distribution," presented at the *137th Convention of the Audio Engineering Society* (2014 Oct.), paper 9178.
[13] H. M. Jackson, M. D. Capp, and J. R. Stuart, "The Audibility of Typical Digital Audio Filters in a High-Fidelity Playback System," presented at the *137th Convention of the Audio Engineering Society* (2014 Oct.), paper 9174.
[14] J. D. Reiss, "A Meta-Analysis of High Resolution Audio Perceptual Evaluation," *J. Audio Eng. Soc.*, vol. 64, no. 6, pp. 364–379 (2016 Jun.). http://dx.doi.org/10.17743/jaes.2016.0015.
[15] A. Pras and C. Guastavino, "Sampling Rate Discrimination: 44.1 kHz vs. 88.2 kHz," presented at the *128th Convention of the Audio Engineering Society* (2010 May), paper 8101.
[16] A. Yoneya, "Perceptually Affecting Electrical Properties of Headphone Cable – Factor Hunting Approach," presented at the *147th Convention of the Audio Engineering Society* (2019 Oct.), paper 532. http://www.aes.org/e-lib/browse.cfm?elib=20555.
[17] A. Yoneya, "Physical Characteristics of Analog Audio Cables and Their Effect on Sound Quality," presented at the *148th Convention of the Audio Engineering Society* (2020 May), paper 10338. http://www.aes.org/e-lib/browse.cfm?elib=20755.
[18] R. Black, "Audio Cable Distortion Is Not a Myth!" presented at the *120th Convention of the Audio Engineering Society* (2006 May), paper 6858. http://www.aes.org/e-lib/browse.cfm?elib=13662.
[19] K. R. Fause, "Fundamentals of Grounding, Shielding, and Interconnection," *J. Audio Eng. Soc.*, vol. 43, no. 6, pp. 498–516 (1995 Jun.).
[20] S. R. Macatee, "Considerations in Grounding and Shielding Audio Devices," *J. Audio Eng. Soc.*, vol. 43, no. 6, pp. 472–483 (1995 Jun.).
[21] N. Muncy, "Noise Susceptibility in Analog and Digital Signal Processing Systems," *J. Audio Eng. Soc.*, vol. 43, no. 6, pp. 435–453 (1995 Jun.).
[22] B. Whitlock, "Balanced Lines in Audio Systems: Fact, Fiction, and Transformers," *J. Audio Eng. Soc.*, vol. 43, no. 6, pp. 454–464 (1995 Jun.).
[23] J. Atkinson, "Bryston BDP-1 Digital Audio Player Measurements," https://www.stereophile.com/content/bryston-bdp-1-digital-audio-player-measurements. (accessed May 4, 2021).
[24] M. N. Kunchur, "3D Imaging in Two-Channel Stereo Sound: Portrayal of Elevation," *Appl. Acoust.*, vol. 175, 107811 (2021 Apr.). https://doi.org/10.1016/j.apacoust.2020.107811.
[25] C. R. Kunchur, "Evaluating Room Acoustics for Speech Intelligibility," *Open J. Appl. Sci.*, vol. 9, no. 7, pp. 601–612 (2019 Jul.). http://dx.doi.org/10.4236/ojapps.2019.97048.
[26] T. Lund, A. M¨akivirta, and S. Naghian, "Time for Slow Listening," *J. Audio Eng. Soc.*, vol. 67, no. 9, pp. 636–640 (2019 Sep.). http://dx.doi.org/10.17743/jaes.2019.0023.
[27] T. Lund and A. M¨akivirta, "On Human Perceptual Bandwidth and Slow Listening," in *Proceedings of the AES International Conference on Spatial Reproduction - Aesthetics and Science* (2018 Jul.), paper P6-2. http://www.aes.org/e-lib/browse.cfm?elib=19621.
[28] M. A. Cohen, P. Cavanagh, M. M. Chun, and K. Nakayama, "The Attentional Requirements of Consciousness," *Trends Cogn. Sci.*, vol. 16, no. 8, pp. 411–417 (2012 Aug.). http://dx.doi.org/10.1016/j.tics.2012.06.013.
[29] M. A. Cohen, D. C. Dennett, and N. Kanwisher, "What is the Bandwidth of Perceptual Experience?" *Trends Cogn. Sci.*, vol. 20, no. 5, pp. 324–335 (2016 May). http://dx.doi.org/10.1016/j.tics.2016.03.006.
[30] D. Whitney, J. Haberman, and T. D. Sweeny, "From Textures to Crowds:Multiple Levels of Summary Statistical Perception," in J. S. Werner and L. M. Chalupa (Eds.), *The New Visual Neurosciences*, pp. 695–710 (MIT Press, Cambridge, MA, 2014).
[31] S. Kumar, S. Joseph, P. E. Gander, N. Barascud, A. R. Halpern, and T. D. Griffiths, "A Brain System for Auditory Working Memory," *J. Neurosci.*, vol. 36, no. 16, pp. 4492–4505 (2016 Apr.). http://dx.doi.org/10.1523/JNEUROSCI.4341-14.2016.
[32] R. M. Warren, "Perceptual Restoration of Missing Speech Sounds," *Sci.*, vol. 167, no. 3917, pp. 392–393 (1970 Jan.).
[33] E. Myers, "From Sound to Meaning," *Phys. Today*, vol. 70, no. 4, pp. 34–39 (2017). http://dx.doi.org/10.1063/PT.3.3523.
[34] R. M.Warren, J. M. Wrightson, and J. Puretz, "Illusory Continuity of Tonal and Infratonal Periodic Sounds," *J. Acoust. Soc. Am.*, vol. 84, no. 4, pp. 1338–1342 (1988 Oct.).
[35] L. Leventhal, "How Conventional Statistical Analyses Can Prevent Finding Audible Differences in Listening Tests," presented at the *79th Convention of the Audio Engineering Society* (1985 Oct.), paper 2275.
[36] B. Rosner, *Fundamentals of Biostatistics*, 7th ed. (Brooks/Cole, Boston, MA, 2011).

[37] "Wilcoxon Signed-Ranks Table," https://www.real-statistics.com/statistics-tables/wilcoxon-signed-ranks-table/. (accessed May 4, 2021).
[38] M. Levine and M. H. Ensom, "Post Hoc Power Analysis: An Idea Whose Time Has Passed?" *Pharmacother.*, vol. 21, no. 4, pp. 405–409 (2001 Apr.). https://pubmed.ncbi.nlm.nih.gov/11310512/.
[39] M. Lawrence, "Dynamic Range of the Cochlear Transducer," *Cold Spring Harb. Symp. Quant. Biol.*, vol. 30, pp. 159–167 (1965).
[40] B. M. Johnstone, K. J. Taylor, and A. J. Boyle, "Mechanics of the Guinea Pig Cochlea," *J. Acoust. Soc. Am.*, vol. 47, no. 2B, pp. 504–509 (1970).
[41] W. S. Rhode, "Observations of the Vibration of the Basilar Membrane in Squirrel Monkeys Using the Mossbauer Technique," *J. Acoust. Soc. Am.*, vol. 49, no. 4B, pp. 1218–1231 (1971).
[42] W. Jesteadt, C. C.Wier, and D. M. Green, "Intensity Discrimination as a Function of Frequency and Sensation Level," *J. Acoust. Soc. Am.*, vol. 61, no. 1, pp. 169–177 (1977 Jan.).
[43] E. Zwicker and H. Fastl, *Psychoacoustics: Facts and Models*, *Springer Series in Information Sciences*, vol. 22 (Springer, New York, NY, 1999). ISBN-10: 9783540650638.
[44] D. E. Hall, "Chapter 6: The Human Ear and Its Response," in *Musical Acoustics*, 3rd ed., pp. 94 (Brooks/Cole Thomson Learning Publishing, Boston, MA, 2002).
[45] M. N. Kunchur, "Audibility of Temporal Smearing and Time Misalignment of Acoustic Signals," *Electr. J. Tech. Acoust.*, vol. 17, pp. 1–18 (2007 Aug.). http://www.ejta.org/en/kunchur1.
[46] M. N. Kunchur, "Temporal Resolution of Hearing Probed by Bandwidth Restriction," *Acta Acust. United Acust.*, vol. 94, no. 4, pp. 594–603 (2008 Jul./Aug.). http://dx.doi.org/10.3813/AAA.918069.
[47] Audio Engineering Society, "Pro Audio Reference (W)," https://www.aes.org/par/w/#ITU_R_468.
[48] ITU-R, "Measurement of Audio-Frequency Noise Voltage Level in Sound Broadcasting," *Recommendation ITU-R BS.468-4*, https://www.itu.int/dms_pubrec/itu-r/rec/bs/R-REC-BS.468-4-198607-I!!PDF-E.pdf.
[49] M. N. Kunchur, "An electrical study of single-ended analog interconnect cables", *IOSR Journal of Electronics and Communication Engineering*, vol. 16, no. 6, pp. 40-53 (2021 Dec.). DOI: 10.9790/2834-1606014053 http://boson.physics.sc.edu/~kunchur/papers/Interconnect-cable-measurements--Kunchur.pdf
[50] AES Journal Forum for the present article: https://secure.aes.org/forum/pubs/journal/?ID=979 (does not require AES membership for access).

* Preprints of this and other papers by the author can be downloaded from the web site:
http://boson.physics.sc.edu/~kunchur/
Click on the "High-fidelity sound reproduction…" link under "Research areas"

M. N. Kunchur

Milind N. Kunchur is a Governor's Distinguished Professor and Michael J. Mungo Distinguished Professor at the University of South Carolina in Columbia, U.S.A. He received his Ph.D. in Physics from Rutgers University, U.S.A. in 1988. He is a Fellow of the American Physical Society and the South Carolina winner of the CASE and Carnegie Foundation U.S. Professors of the Year award. He has also received the George B. Pegram Medal, Ralph E. Powe Award, Donald S. Russell Award, Martin-Marietta Award, Michael A. Hill Award, Michael J. Mungo Award, and held a National Research Council Senior Fellowship. He served as a panelist on a High-Resolution Audio workshop at an Audio Engineering Society convention.