Music, Sound Quality, and Hearing Aids: An Interview with Brian Moore and Richard Einhorn

By DOUGLAS L. BECK, AuD

A few weeks ago, I had a conversation with Brian Moore, PhD, and Richard Einhorn regarding multiple factors which influence the perceived quality of live versus recorded music. Dr Moore is the Emeritus Professor of Auditory Perception at the University of Cambridge in the United Kingdom. He has written or edited 17 books and over 590 scientific papers and book chapters and also serves as an associate editor of the journal *Hearing Research*. His research is vast and admittedly complicated, yet always involves practical applications such as fitting hearing aids, managing cochlear dead zones, high fidelity sound reproduction, and the perception of sound.

Richard Einhorn is a composer who has received critical acclaim for *The Origin*, an opera/oratorio based on the work and life of Charles Darwin, and *Voices of Light*, an opera with silent film which has enjoyed over 100 performances throughout the world, including sold-out events at the Kennedy Center and Wolf Trap with the National Symphony. In fact, *Voices of Light* became a Billboard classical bestseller—earning Einhorn the unique distinction of being one of only a few composers to have made "the charts." Particularly valuable to our field, he has a significant hearing loss, writes regularly about hearing loss and music, and serves as a hearing loss consultant at his firm, Einhorn Consulting LLC in New York City.

Douglas L. Beck, AuD, is the Director of Academic Sciences at Oticon Inc, Somerset, NJ. He has served as Editor In Chief at AudiologyOnline, and Web Content Editor for the American Academy of Audiology (AAA). Dr Beck is an adjunct professor of audiology at Lamar University, and an adjunct associate professor of audiology at the University of Hawaii. He also serves as Senior Editor of Clinical Research for the *Hearing Review's* Inside the Research.

Beck: Good morning Brian and Richard. Thanks for your time and thanks for participating in this discussion about sound quality, amplification, and music.

Brian, let's start with your recent article, "Effects of Sound Induced Hearing Loss and Hearing Aids on the Perception of Music"[1] which has published in the *Journal of the Audio Engineering Society*. You mentioned that, given typical mild-to-moderate sensorineural hearing loss (SNHL), the listener loses many of the perceptual characteristics of music, such as fine temporal structure, pitch perception, loudness based dynamics, and more. These changes impact the quality of speech and music perceived by the listener with hearing loss, and so...how does one quantify or qualify the impact of hearing loss on speech and music?

Moore: It is hard to quantify the effects of hearing loss, but it *is* possible to simulate some effects of hearing loss, via suitable signal processing, to illustrate the effects of hearing loss to a person with normal hearing. For example, the effects of loudness recruitment combined with threshold elevation can be simulated using multi-channel dynamic range expansion,[2] and the effects of reduced frequency selectivity can be simulated using spectral smearing.[3] We have assessed the validity of these simulations using participants with one normal ear and one hearing-impaired ear.[4] Participants were asked to compare the speech or music heard in their impaired ear with the same sound processed to simulate their hearing loss and presented to their normal ear. Most participants thought the processed sound in the normal ear matched what they heard in their impaired ear fairly well, although what they heard in their impaired ear was a bit worse. I produced a compact disk of these simulations called "Perceptual consequences of cochlear damage." [For details, go to http://hearing.psychol.cam.ac.uk/Biogs/cd.html.]

Beck: What typically happens to speech and music quality when speech and music are amplified through modern hearing aids? What are the major factors?

Moore: One factor that needs to be considered is the dynamic range of the analog-to-digital converter (ADC). Peak sound levels can be very high for live music, and the upper limit of the ADC can easily be exceeded. This can result in nasty-sounding distortion. Hence, a wide dynamic range is important, especially for live music.

The frequency range of the hearing aid is also important. Many hearing aids provide little or no amplification for frequencies below 200 Hz or above 5,000 Hz, which can lead to poor sound quality.[5] Irregularities in the frequency response can degrade sound quality. Hence, a smooth wideband frequency response is desirable.

Another issue is the speed of the multichannel compression system. The preferred compression speed varies across individuals. For speech, some people prefer fast-acting compression and some prefer slow-acting compression, while for music, most people prefer slow compression or they have no preference.[6]

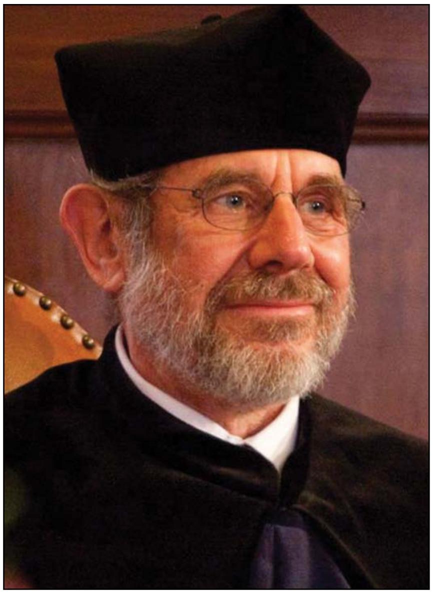
Finally, different types of adaptive processing in hearing aids can also reduce sound quality, especially for music. Hence, for music listening it may be desirable to have a special program (or to have the hearing aid automatically select a program) for listening to music in which all non-essential forms of adaptive processing are turned off.

Beck: Thanks Brian. Richard, in your recent Letter To The Editor in *Hearing Review*,[7] you said "electronic sound reproduction should be as flat and distortion free as possible…" Would you please elaborate on that?

Einhorn: Sure! My background is in music composition, audio engineering, and record production. One of the first things my mentor (the great classical producer Andrew Kazdin) told me was the ideal audio reproduction system was "a copper wire with gain." What he meant was that sound which comes through loudspeakers or headphones should sound exactly like the original sound (unless you proactively want to alter the sound for artistic reasons).

Of course, perfect sound reproduction is never entirely possible, although modern pro audio gets awfully close. Recently, I was looking at specs for a microphone that claimed a frequency response of 6 Hz to 30 kHz +1/-3dB (Earthworks). Now that's flat! Professional digital audio, recorded at 24 bits per sample, has a maximum dynamic range of 144 dB. Although there's some controversy regarding sound quality at very high sample rates, typical professional recordings are at either 96 kHz or 192 kHz, producing a theoretical frequency response way beyond the probable human limit of 20 kHz.[8]

So, for all practical purposes, pro audio tech essentially takes equipment sound quality off the table. Even at reasonable price points, like under $1,000 for an ADC/DAC, the sound quality of modern audio is remarkable.[9]



Dr Brian C.J. Moore

Richard Einhorn

Beck: I agree, that is remarkable. Do you think hearing aids should be as flat and distortion free as possible? Or should they perhaps "shape the sound" for the listener of speech or music?

Einhorn: The answer is yes and yes, and in that order! Although hearing aids have design constraints that professional audio doesn't: small size, limited battery power and feedback control, for example. It strikes me as more than likely that, if fundamental hearing aid design aspired to the "copper wire with gain" ideal, rather than the "do what it takes to make speech audible" approach, live music could be reproduced more accurately for people with hearing loss.

I suspect, in many situations, very high sound quality will improve speech comprehension. With friends who have extremely serious sensorineural hearing losses, I've done some informal tests with great sound equipment and they have benefited greatly. Chasin[10] found similar results with a new hearing aid input gain structure that delivers less clipping at high sound levels. This is fully congruent with the thrust of Brian's latest paper and my experience. It also reinforces Mead Killion's insights that, in hearing aid design, sound quality matters—and it matters a lot.[11,12]

Of course, people with hearing loss need help to hear music, and each person's needs are different. So, given truly accurate audio equipment, I believe compensatory equalization (EQ) and compression should be added, but as little as possible.[13] It seems to me likely that most music lovers with hearing loss could—if given a simple app-based user interface—adjust the frequency balance themselves for enjoyable music listening.[14]

Regarding compression for music (which is much harder to self-adjust), I'd suggest only enough to create a maximum dynamic range of about 15 dB (or less) for most pop music and most people with hearing loss. Only a few compression bands—possibly only one—and slow release times would be helpful for music.

Additionally, live music lovers need to be trained in hearing aid "mic techniques." For example, they should sit as close as possible at classical music concerts and turn down the gain at loud amplified events.

Beck: And what about people who listen to music through frequency lowering hearing aid circuits?

Einhorn: Brian's paper[1] implies that, at best, small amounts of frequency lowering might not do much harm, but they don't improve music perception. I agree, and my experience with similar techniques suggests why they can't really help. Totally missing pitches, such as high flute notes, or misheard pitches, such as hearing an E-flat as G-sharp (a problem I have in my right ear), can't be adequately corrected in the real world, and indeed, significant frequency lowering has the potential to distort the melodic and harmonic character of music.

This brings up a larger issue. Although most of us would love to hear well again, hearing loss sometimes creates problems which are simply intractable. For example, due to a combination of sudden sensorineural hearing loss in my right ear and a moderate/severe mixed hearing loss in my left, my right ear has severe loudness recruitment/hyperacusis, as well an unbelievable amount of pitch distortion. In my left ear, which has a moderate/severe mixed loss, I can still hear pitch accurately, but I have a limited dynamic range, some loudness recruitment, and high frequency loss. So my left ear requires significant amplification, multi-band compression, and a very high SNR for speech comprehension.

Beck: One thing I've often considered, which may seem a bit controversial,[15] is that musicians should not wear hearing aid amplification while practicing or performing. Specifically, if I strum an acoustic guitar lightly, and if I sing along with that same sound in my Dylan-esque singing voice, the level can easily reach 90 dB SPL, and quite simply, unless one has a profound hearing loss, the need for amplification while practicing or performing is usually about zero. Your thoughts, please?

Einhorn: I'm not sure it's that clear cut. Some musicians may be fine using hearing aids when playing while others may find music doesn't have enough bass, or treble, or their instrument sounds distorted. One alternative may be to use an iPhone with a good set of earphones, a recording app like Rode Rec, and a great microphone. This is a nice portable amplification system with amazing sound quality.

However, there obviously are some situations—such as playing in a symphony orchestra—where an elaborate music tech monitoring setup simply isn't feasible. So I think it has to be managed case by case. The more flexible the musician with hearing loss is, and the more open to experiment, the better. If playing with hearing aids sounds good, terrific, go with it.

Some musicians with profound hearing loss find unique individual ways to perform beautifully. Percussionist Evelyn Glennie takes her shoes off and feels the stage vibrations to help her perform. Composer/pianist Jay Alan Zimmerman relies on clear visual sightlines with his collaborators to keep his place—and his mind-blowing musical imagination.

Beck: I've played with musicians all over the world, and the thought of some of those guys taking their shoes off is kinda scary.... but OK, I know what you mean. Brian, what are your thoughts on musicians wearing amplification to practice or perform?

Moore: I understand your argument, but there are some drawbacks to what you propose. Even though the overall sound level may reach 90 dB SPL, the level at some frequencies will be much lower than this. Amplification at frequencies where the level is lowest and/or the hearing loss is greatest may be required to restore a natural sound quality.

For example, I have a mild-to-moderate high-frequency hearing loss. When I play an acoustic guitar without my hearing aid turned on, the sound quality is rather dull; the high harmonics are softer than they should be. I might try to compensate for this by plucking the strings closer to the bridge, but this would lead to a somewhat harsh sound quality for anyone with normal hearing who was listening. When I play the guitar with my hearing aid turned on, the sound is much brighter and more natural (although I have found that, with some hearing aids, the high frequencies are over-emphasized and the sound becomes too bright), and I can judge better what a normal-hearing audience might be hearing.

Of course, this all depends on having an appropriately fitted hearing aid with a good amplitude-compression system. The aid should not amplify sound components whose level is already high.

Beck: Fair enough, good point Brian. So then, with regard to the patient with a symmetric mild-moderate SNHL, what do you suggest with specific regard to compensating for threshold elevation and loudness recruitment, when this individual chooses to listen to music at home on her stereo, and at a live concert?

Moore: For listening at home, the main requirements are a smooth wideband frequency response, an appropriately fitted hearing aid with multi-channel compression (or "channel-free" compression processing), and a "Music" program for which all fancy signal processing is turned off (no noise reduction, no adaptive directionality, minimal feedback cancellation, etc).

For listening to live music, a hearing aid with a wide input dynamic range is needed, as mentioned earlier. If a "closed" fitting is used [where the earmold or receiver is sealed into the ear canal] it may be preferable for the hearing aid to attenuate high-level sounds, especially intense low-frequency sounds. This can help protect the ear from damage and reduce the masking of medium and high-frequency sounds by intense low-frequency sounds. Many hearing aid manufacturers avoid negative gains, but I think there are good reasons why the gain should become negative when the input level is sufficiently high.

Einhorn: Frankly, existing "music programs" in hearing aids have not helped me hear music better. At home, when listening casually, I use my standard program. For serious listening, I take my hearing aids out and use good headphones like the Sennheiser 650 (around $500). For live music, I often use the house assistive listening system. It's usually FM, but I substitute my own in-ear earphones for the invariably awful house headphones. Hamilton on Broadway has a loop system installed and the sound quality is excellent via t-coil. When there isn't a good house ALD, I simply sit close, take out my iphone, plug in my Blue Mikey, my in-ears, and use Jacoti ListenApp (disclosure: I consult for Jacoti).

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Beck: Brian, I recall that, in your 2016 paper,[1] you address slow- and fast-acting AGC circuits. If you can take a moment to re-define those here, as well as the advantages and disadvantages of each, that would be great.

Moore: Almost all hearing aids incorporate some form of automatic gain control (AGC, also called multi-channel compression) to compensate for the reduced dynamic range and loudness recruitment associated with cochlear (SNHL) hearing loss. The speed of response of the AGC is usually measured by using an input sound whose level changes abruptly between two values. When the sound level abruptly increases, the gain decreases, but this takes time to occur. The time taken for the output to get within 3 dB of its steady value is called the "attack time." When the sound level abruptly decreases, the gain increases, but again, this takes time to occur. The time taken for the output to increase to within 4 dB of its steady value is called the "recovery time" or "release time."

Some AGC systems are intended to adjust the gain automatically for different listening situations. The gain is changed slowly with changes in input sound level. This is achieved by making the recovery time, or both the recovery time and the attack time, relatively long—usually between 0.5 and 20 seconds. These systems are often referred to as slow-acting AGC. Slow-acting AGC systems do not restore loudness perception to "normal" but they can make soft sounds audible without making intense sounds uncomfortably loud. They also produce very little distortion of the temporal envelopes of sound.

The second class of AGC system is intended to make the hearing-impaired person's perception of loudness more like that of a normal-hearing listener. Loudness recruitment behaves like fast-acting multichannel expansion, so restoration of loudness perception to normal requires fast-acting multi-channel compression.[16] Systems with this goal have relatively short attack times (0.5 to 20 ms) and short recovery times (5-200 ms) and are referred to as "fast-acting compressors" or "syllabic compressors," since the gain changes over time, comparable to the durations of individual syllables in speech. While such systems can come close to restoring loudness perception to normal, they distort the temporal envelope of sounds, and they can introduce an effect called "cross-modulation" where the temporal envelopes of independent sounds (eg, two people talking) become partly correlated at the output of the hearing aid.[17] This can make it harder to perceptually separate the two sounds.

Beck: And what about the use of adaptive compression circuits, which automatically change their release times based on the acoustic environment?

Moore: In principle, adaptive compression, and variants of it, can work well if the parameters are chosen appropriately. In my laboratory we developed a dual time-constant system which acted as a slow-acting compressor most of the time, but which briefly changed to a fast compressor when sudden changes in sound level occurred.[18,19] This system has been implemented in a cochlear implant system,[20] and variants of it are also used in hearing aids.

Beck: In hearing aid technology, most of the major manufacturers offer bandwidths that approximate 10 kHz. Is it necessary to go beyond 10,000 Hz for speech or music?

Moore: Although many manufacturers claim a 10-kHz bandwidth, in my experience few hearing aids provide useful gain for frequencies above about 6 kHz. However, to answer your question, there is clearly a benefit in increasing the bandwidth from 5 or 6 to 10 kHz,[21] but I think that a 10 kHz bandwidth is sufficient for both speech and music.

Einhorn: I agree with Brian that 10 kHz is probably sufficient. However, in keeping with pro-audio's concept of taking equipment coloration completely off the table, I'd like to hear full bandwidth hearing assistance sometime in the future—or about 20 Hz to 20 kHz. With something of that quality, any issues regarding sound quality become moot and hearing care professionals could feel comfortable that they are compensating only for the person's hearing loss, not the equipment's frequency response.

I'd love to see formal research on whether such a different approach to hearing assistance—one that optimized equipment sound quality to the standards of contemporary audio and used common professional techniques—could significantly improve music, and possibly speech, comprehension for people with sensorineural hearing loss. I suspect the answer is yes for music and quite possibly yes for speech, in most cases. If so, then the far-from-trivial challenges become how to create practical, attractive, and affordable hearing technologies that incorporate such an approach.

Beck: That's an interesting point Richard. Some of us have been advocating high frequency testing protocols as a standard protocol for decades, but to be fair, very few professionals have equipment to test above 8 kHz. Of course the equipment (audiometers and head phones) do exist, but calibration and interpretation of results remain an issue. One final issue, what about the effect of dynamic range (ie, amplitude) compression, in general, on music?

Einhorn: From my perspective, with few exceptions, compression is more a problem with live sound than it is with recorded. Pop music's dynamic range is often less than 10 dB.[22] Non-amplified live music is an entirely different story. A classical concert can have such a vast dynamic range that it creates essentially insurmountable compression obstacles for a live mic hearing aid system—when it gets quiet the compression amplifies extreme amounts of background noise which is very unpleasant.

By far the best solution is to go direct from on-stage mics into hearing aids, which minimizes excessive ambience. The only way to do that with good sound is via induction loops, but sadly, loops are still very rare in the United States (neck loops don't provide adequate sound quality for music listening). So the best thing to do is, again, to remove your aids, place them somewhere safe, and use the house ALD with your own in-ear earphones if you can.

Moore: No matter how carefully they are set up, compression systems never compensate fully for the loudness recruitment and reduced dynamic range of the hearing-impaired listener. This partly reflects limitations in signal processing and partly reflects limitations in the way hearing aids are fitted. Many hearing aid dispensers perform real-ear measurement and make adjustments to try to match "target" gains at different frequencies, but they rarely check the frequency-gain characteristic for more than one input level. In my experience, the amount of amplitude compression that is actually achieved is often less than the programmed amount of amplitude compression. As a result, some soft sounds remain inaudible while some intense sounds are too loud.

Einhorn: Agreed. There are situations with hearing loss in which no technology can help. And for passionate music lovers with hearing loss, it points to the all-important counseling function of hearing care professionals. We need to accept that our hearing is not what it could be—which can be very hard to accept—and the professionals can help us understand that.

Beck: Thanks Richard and Brian. It's been a pleasure working with each of you. Your expert opinions and guidance are appreciated and I suspect you've provided our colleagues many thoughts for their consideration and reflection. I'll look forward to the next time our paths cross. ◗

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REFERENCES can be found in the online version of this article at www.hearingreview.com.