Music and the Problem With Hearing Aids

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This article is a review of the strengths and weaknesses of modern digital hearing aids to be able to transduce music effectively. One major area that is problematic is the analog to digital (A/D) converter that is located just after the input to the hearing aid. A theoretical limit of 96 dB is given for the dynamic range of any 16-bit system, and because many components of music are in excess of 96 dBA, the A/D converter element is the weak point in modern hearing aids. In contrast, because the highest sound levels of speech as an input to a hearing aid are on the order of 85 dBA, this limitation is not an issue for speech. The author makes a survey of the technical innovations that have reached the market place, and provides a series of clinical strategies and overall recommendations.

Modern day analog-to-digital (A/D) converters are the weak point with digital hearing aids. Many of the innovations and technologies have been developed in order to avoid some of the inherent problems. In a nutshell, given the 16-bit architecture in the hearing aid industry, while a dynamic range of 96 dB or less is quite capable of transducing the various elements of speech, it falls short when it comes to be able to handle the more intense elements of music. Technical strategies and clinical recommendations are provided below that will improve the ability of modern digital hearing aids to be high-fidelity transducers of music.

Acoustic Properties of Speech

Modern digital hearing aids are capable of transducing speech of all languages. While there are many phonemic differences among the various languages of the world (Chasin, 2011) there are minimal phonetic differences (Byrne et al. 1994). The long-term average speech spectrum is sufficiently similar to be used for a number of languages. This phonetic similarity is quite understandable, given that speakers from around the world have a vocal tract that is roughly 17 cm long, a nasal cavity that is in parallel with the oral cavity, and tongues and lips whose mechanical and neurological characteristics do not vary from country to country and from language to language.

All languages consist of low-frequency sonorants (vowels, nasals, and liquids) and high-frequency obstruents (stops, affricates, and fricatives). Modern hearing aids have microphones, receivers, and amplifiers that can handle all phonetic elements of speech within their operating characteristic.

Because of the nature of the mechanics of the human vocal tract and air supply from the lungs, lower frequency sonorants are typically more intense than higher frequency obstruents. While the long-term average speech spectrum (LTASS) may vary slightly from language to language, the differences are minimal.

The highest sound level of speech is typically in the 80–85 dB SPL (sound pressure level) range. People can shout at levels of 90 dB SPL, but this is not a routine part of normal conversational behavior, nor is it part of the calculation toward the LTASS.

The *fundamental frequency* of speech (also known colloquially as the *pitch*) is correlated with the vibration of the vocal chords (in the larynx) and can be as low as 100 Hz for large men and as high as 400 Hz for small children. The fundamental frequency is the lowest possible frequency in the spectrum of speech. And like violins and guitars, the human vocal cords function as a one half-wavelength generator with higher frequency harmonics spaced in integer intervals of the fundamental frequency. For a male with a fundamental frequency of 125 Hz, the first harmonic is at 250 Hz, then at 375 Hz, and so on, spaced out at 125 Hz intervals. There is not speech energy below the frequency of the fundamental.

Acoustic Properties of Music

Speech possesses a well-defined spectrum. This is in contrast to music, which may possess highly variable spectra. Music can be speechlike, with significant low-frequency energy and less high-frequency energy (e.g., clarinet), or it can be percussive in nature (e.g., drums). Music can be quiet, such as folk and jazz, or at a much higher sound level, such as rock and classical music. Table 1 is adapted from Chasin (2006a) and shows a sampling of the various sound levels in dBA (decibel A-weighted) for a wide range of musical instruments. Unless specified, all sound levels were measured on the horizontal plane and from a distance of 3 meters. In the case of the violin, the sound level is also provided at the musician's left ear.

Table 1. Average sound levels of a number of musical instruments measured from 3 meters. Also given is the sound level for the violin measured near the left ear of the players. Adapted from Chasin (2006a). Used with permission.

Musical Instrument	**dBA Ranges Measured From 3 Meters**
Cello	80–104
Clarinet	68–82
Flute	92–105
Trombone	90–106
Violin	80–90
Violin (near left ear)	85–105
Trumpet	88–108

Unlike speech, music can have a low-frequency emphasis with its fundamental being less than 50 Hz (for double bass and tuba) or have energy only above 1000 Hz (for cymbals and rimshots).

There are some musical instruments that share the mechanical and acoustic properties of the human larynx. These *one-half wavelength generators* (whose vibrating sources are held tightly at both ends like the human vocal cords) are stringed instruments such as the violin, viola, cello, bass, and guitar. More information on this can be found in any introductory book on acoustics or music.

There are some other musical instruments that function as *quarter wavelength generators* with harmonics at odd-numbered multiples of the fundamental. For a fundamental frequency of 125 Hz, such instruments would have harmonics at 375 Hz, 625 Hz, and so on. The harmonic structure of these instruments is less densely packed than that of human speech or the many half wavelength resonator instruments. This lack of "harmonic density" underlying a spectrum has been the subject of much debate over the years, and it is still of an unknown effect.

One feature that differentiates speech from music is that speech tends to be “serial,” in the sense that one phonetic segment will follow another but have minimal overlap—one does not utter a low-frequency sonorant at the same time as uttering a high-frequency obstruent. There is some overlap, called *co-articulation*, but it is rare to have a low-frequency element being co-incident with a high-frequency element. In contrast, there is nothing inherent in music that would prevent this co-occurrence between low- and high-frequency sounds. This has some ramifications for transducers that may be required to vibrate at both a low- and a high-frequency rate at the same time. To date, there is little research that has addressed this issue. Many manufacturers of in-ear monitors, which many musicians wear while playing and listening to music, are marketed according to how many receivers are used—some for low-frequency sound energy and other(s) for higher frequency sound energy. This may be a decision that is based more on marketing rather than one based on the acoustic and mechanical properties of small diaphragm receivers.

Crest Factors and Music

Another difference between speech and music is the *crest factor*, which is the difference in decibels between the average (or root mean square [RMS]) of a signal and its instantaneous peak.

For speech, the crest factor is typically taken as 12 dB (and, like the dynamic range, which is also a difference measure, no suffix is required for the dB value). This value serves as the basis of many of our hearing aid test protocols (e.g., ANSI S3.22-2003) and is physically related to the degree of damping in the human vocal tract. Speakers of all languages of the world have soft tissues such as the tongue, cheeks, lips, soft palate, and nasal cavity. The output of the human vocal tract has a significant degree of acoustical and mechanical damping, such that the peaks of the output are only about 12 dB greater than the average or RMS value of the signal (Cox, Mateisch, & Moore, 1988; Sivian & White, 1933).

In contrast, hard-walled musical instruments have less inherent damping than the vocal tact with a resulting higher crest factor. Typical measured crest factors with many musical instruments (and music in general) are on the order of 18–20 dB. This is roughly 6 dB greater than for speech and, as will be seen, has direct ramifications for setting the OSPL90 for a “music” program. Adding this 18–20 dB crest factor to the values found in Table 1 indicates peak-musical levels that are far in excess of 100 dB SPL.

Theoretical Limitations of a 16-Bit System

Although this is gradually changing, the vast majority of modern digital hearing aids have a 16-bit system. Without getting into too much detail, this refers to the resolution of being able to store precise amplitudes of a signal. A low-bit system would not be able to replicate the input signal in a digital system without significant information being lost. This lost information can be thought of as error and, specifically in the digital domain, is referred to as *quantization error*. The advantage of a 16-bit system is that the quantization error is below the audible threshold of a normal-hearing person and we are simply not aware of it.

Another aspect of a change in the resolution of being able to store the correct amplitude of a signal is the dynamic range- the difference in decibels between the lowest and the highest sound level that can be transduced through the digital system without distortion.

For those who like mathematics, the dynamic range of a system that has N bits is given by the equation in Figure 1, and a calculation is given for a 16-bit system.

Figure 1. The theoretical formula for the calculation of the dynamic range of an N-bit system, showing an example for a 16-bit system. Because of engineering decisions, the actual dynamic range tends to be slightly less than this figure.

DYNAMIC RANGE = $20\log_{10}2^N$

Example for a 16-bit system:

$$\begin{aligned}\text{Dynamic range} &= 20\log_{10}2^{16}\\ &= 20(16)\log_{10}2\\ &= 320 \times 0.3\\ &= 96\text{ dB}\end{aligned}$$

The calculations in Figure 1 show us that a 16-bit system has a dynamic range of 96 dB. Because the most intense element of human speech—the vowel [a] as in "father"—is on the order of 80–85 dB SPL, then such a system would be quite able to handle the more intense elements of speech. In fact, that is probably why a 16-bit system was initially chosen for the optimal transduction of speech—the quantization error is below the threshold of audibility for normal-hearing people, and the highest level sound that can be transduced is well above that of human speech. However, as we will see later, this 96 dB dynamic range is not necessarily 96 dB SPL. It is merely the difference (with no suffix to the dB scale) between the lowest and the highest sound level that can be transduced through the digital system without distortion.

In contrast to speech, as shown in Table 1, most musical instruments (and regardless of the type of music being played) have sound levels far in excess of 96 dB SPL. While a 16-bit system is more than sufficient for the requirements of speech, it falls short of the requirements of listening to music.

The architecture of a digital system has the number of bits specified in its A/D converter. This component is located after the hearing aid microphone(s) but before the (compression) software of the hearing aid. Overdriving the A/D converter with music inputs that are in excess of 96 dB SPL will result in distortion of the signal. No amount of software manipulation that occurs later in the processing will resolve the problem. This is also the case whether an *inductive*, or direct audio, input is used with the hearing aid—all inputs are required to be digitized by the A/D converter.

In reality, because of the various engineering decisions that need to be made, the actual dynamic range is typically slightly less than the 96 dB figure.

Can the Hearing Aid Handle the Higher Levels of Music?

Clinically, we have just selected an optimal hearing aid for speech, and we are concerned about how it may handle the louder components of music. Here is a quick clinical test that can be performed—which only needs to be performed one time for each hearing aid under consideration—that will provide immediate assurance that a hearing aid will also be optimal for music. The procedure is discussed in Chasin (2006b) and is based on the limitations of the A/D converters in modern digital hearing aids.

In a hearing aid test box, select a level that is typical of the input of the music. Depending on the manufacturer of the test system, levels of 90 dB SPL or higher may be selected. In cases where the stimulus level is restricted to 90 dB SPL, higher levels can be generated by either moving the reference microphone outside of the test box or moving the hearing aid closer to the loudspeaker in the test box. The exact level can be verified with a real ear measurement at the location of the hearing aid microphone. Ideally, 100–105 dB SPL is the desired input, since this is the range of many live performances.

Program the hearing aid for about 5 dB of broadband gain and set the OSPL90 to a level that is at least 10 dB greater than the input + gain. A level of 120 dB SPL or greater is typically selected.

With the 105 dB SPL input + 5 dB gain, there should not be any distortion measured if the A/D converter can handle inputs of 105 dB SPL. If there is distortion, it cannot be saturation-related, because the OSPL90 is set to 120 dB SPL or greater. Any measured distortion must be "front end" or A/D converter–related. If there is no measured distortion with this set-up, the hearing aid will be able to handle the full amplitude range of music.

Five Technical Innovations

Technology #1

Although the intent of this article is not to mention any one hearing aid specifically, it would be an error of omission not to discuss the analog K-AMP hearing aid, at least in a historical context. The K-AMP circuit was available through virtually all hearing aid manufacturers in the 1990s and early 2000s and is still commercially available in some markets. For years, it was the only hearing aid that was able to transduce inputs of 115 dB SPL with no distortion. And because it is an analog hearing aid, there is no A/D converter to be overdriven. In fact, most "vintage 1980s" hearing aids were able to handle the higher levels of music better than many of today's modern digital hearing aids, but none as well as the K-AMP. The K-AMP has been one of the most successful hearing aids in the last quarter century for musicians and for those who like to listen to music. Although it is no longer commercially available in many countries (including Canada), it is still available in the United States. Clinically, this almost 25-year-old hearing aid outperforms the vast majority of all digital hearing aids in the market.

Technology #2

A circuit called Head Room Expander (HRX) was a trade name of Sound Design (and now with the recent purchase, is owned by ON Semiconductors). Like the K-AMP, this is a "third-party" manufacturer of hearing aid technology whose customers are many of the large hearing aid manufacturers. HRX functions by "auto-ranging" the input to the hearing aid; this has the byproduct of always ensuring that the levels of music that actually reach the A/D converter are always within the operating range. It is a bit like ducking under a low-hanging doorway: HRX reduces the input (like ducking under the doorway) and then re-establishes its normal level after the A/D converter (like standing up again). In this way, HRX can provide distortion-free input of the more intense components of music. This technology also uses an architecture that is based on a 20–24-bit system, depending on its implementation, and subsequently can handle a higher dynamic range than 96 dB. There are several other manufacturers that use a form of static (non–auto-ranging) compression technique that reduces the input prior to the A/D converter and then re-establishes after the digitization process.

Technology #3

Recall that the dynamic range of 96 dB was the result of a 16-bit architecture. This was not necessarily 0 dB SPL to 96 dB SPL. This same 96 dB range has been shifted up by 15 dB to 15 dB SPL to 111 dB SPL. This new range is still 96 dB, but is now within the operating range to handle the higher sound levels found in music. To date, only one manufacturer has adopted this approach but others are sure to follow.

Technology #4

Most hearing aids utilize a broadband microphone in their design. This is also true of the newer breed of non-occluding behind-the-ear hearing aids for use with clients who have essentially normal hearing thresholds up to about 1000 Hz. The engineering reason for the use of a broadband microphone is to minimize the internal noise of the hearing aid. However, the use of "-6 dB/octave" microphone that is less sensitive to the lower frequency region below 1000 Hz can be quite beneficial (Chasin & Schmidt, 2009). As the name suggests, the hearing aid microphone has been made less sensitive to the more intense lower frequency components of music—specifically, -6 dB less sensitive at 500 Hz and -12 dB less sensitive at 250 Hz. This approach will not change the fidelity of the higher frequency elements of music, but because most of the intense components of music are below 1000 Hz, this "fools" the A/D converter into thinking that the input is well within its operating range. A drawback of this type of microphone is an increase in the internal noise floor of the hearing aid. However, if the expansion kneepoint is set above the internal noise floor, the noise level is then comparable to that found with broadband microphones.

Technology #5

A resistive network prior to the input of the hearing aid can be utilized. Because all inputs to a hearing aid are required to be digitized by the A/D converter, this includes not only the microphone input, but also the inductive and direct audio input routes. In the case of the inductive (neck loop or silhouette) and the direct audio input routes, a resistive network can be constructed and placed in an appropriate boot for the neck loop or the direct audio input cord. This reduces the input by 10 dB, thereby offering an additional 10 dB of headroom to allow more of the input to be within the operating range of the hearing aid's A/D converter. Contacting any manufacturer and asking for them to place a "10kohm resistor in series and a 1 kohm resistor to ground" in a boot will be sufficient. This boot can then be used between an inductive loop or silhouette (or built into a direct audio input cord) and the external assistive listening device.

If the external assistive listening device (e.g., an FM system) has a volume control, then simply reducing the volume slightly will also resolve the potential problem.

Of importance is that each of the above technical innovations are designed to configure the input either to not use an A/D converter (e.g., K-AMP) or to allow the input to be within the operating range of the A/D converter.

Four Clinical Recommendations

Assuming that one has been able to select or configure a hearing aid to receive the higher level components of music with minimal distortion, what are some of the optimal software and electro-acoustic settings for music? There are four general recommendations.

Recommendation #1

There should be similar wide dynamic range compression (WDRC) parameters for speech and for music. There is no inherent reason why modern day WDRC circuitry should be set any differently for music than for speech. The use of the WDRC circuitry is primarily to re-establish normal loudness growth due to (outer) hair cell damage. The use of this circuit is primarily to address damage of our auditory systems, rather than the nature of the input stimuli per se (Chasin & Russo, 2004). The outcome of the Chasin and Russo study was the result of a theoretical calculation of desired inputs. In an empirical study of this issue, Davies-Venn, Souza, and Fabry (2007) found that this was indeed the case. They commented that "Chasin and Russo (2004) suggested that WDRC . . . may be better for music. . . . That hypothesis was supported by the present data" (p. 696).

Recommendation #2

The "music program" should be set with about 6 dB lower OSPL90 and 6 dB lower gain than the client's "speech in quiet" program. This has been called the *-6 dB rule* and is based on the fact that many forms of music have a crest factor that is about 6 dB greater than that of speech. For example, if the crest factor of music is 18 dB and that of speech is 12 dB, then the peaks of music are 6 dB more intense (18 dB–12 dB) than those of speech for a given presentation level. Therefore, in order to prevent the peaks of music from causing discomfort, the OSPL90—assuming similar WDRC parameters for speech and music—and the gain should also be 6 dB lower than the "speech in quiet" program setting.

Recommendation #3

The optimal bandwidths for both "speech" and "music" programs derive from the work of several studies. Examining the work of Moore, Fullgrabe, and Stone (2011) and Ricketts, Dittberner, and Johnson (2008), some general recommendations can be made. These two studies were not specifically about music, but the results can be extended for both "speech" and "music" programs. If the hearing loss is mild (at most, up to a moderate level), then a broader bandwidth is better. If, however, the hearing loss is greater than a moderate level, then less may be more—a narrower bandwidth (which can avoid dead regions in the cochlea) may provide a more pleasant sound than a wider bandwidth that extends into the high-frequency region. The same can be said about the configuration of the audiogram: A person with a relatively flat audiometric configuration should have the widest bandwidth possible, given the limitations of their hearing loss. In contrast, if the audiogram has a precipitous high-frequency loss configuration, then again, less may be more—a narrower frequency response would be ideal.

There has been some discussion that a "music" program should have a wider bandwidth than for a "speech" program, but there no arguments that can be made to support this view. Bandwidth is an audiometric issue and nothing to do with the nature of the input stimuli.

Recommendation #4

Disable the feedback management and noise reduction systems, to the extent that they can be disabled. This is not a well-researched area, but clinical experience suggests that feedback management systems can "turn off" the hearing aid while listening to or playing music. The pure tone nature of harmonics in music can be confused with the pure tone–like nature of a feedback signal. This is especially true of the higher frequencies. To help resolve this in cases where the feedback management system cannot be disabled, some manufacturers have limited the feedback circuit to the higher frequency region, which is a reasonable solution to an otherwise problematic situation.

Four Clinical Strategies

Strategy #1

Turn down the input (stereo) and turn up the hearing aid volume (if necessary). If the excessive level of the input to the hearing aid causes distortion of the A/D converter, then turn down the input if at all possible. If traveling in a car, turn down the level of the radio and (if necessary) turn up the level of the hearing aid to compensate. The output will be the similar, but the input will be reduced to a level that is within the operating range of the A/D converter of the hearing aids.

Strategy #2

Removal of hearing aid for music. Given the higher level inputs of music, the required gain may be close to 0 dB for a desired output. Table 2 shows some data derived for a range of severities of hearing losses at 1000 Hz and the required gains for speech and for music. Even for an 85 dB sensorineural hearing loss at 1000 Hz, though a person may require 45 dB of gain for certain speech sounds (50 dB input), they may only require several decibels of amplification for many types of music (90 dB input). The best strategy for many hard-of-hearing consumers may be to simply remove their hearing aids when listening or playing music. Table 2 shows target gain data at 1000 Hz for a wide range of hearing losses. The data is based on FIG6 but is similar to other fitting formulae.

Table 2. Calculated amounts of gain required for a given hearing loss at 1000 Hz (column 1) based on FIG6. For average levels of music (95 dB A) inputs, virtually no amplification may be required, even for very significant hearing losses. Used with permission. Retrieved March 19, 2012, from www.hearinghealthmatters.org/HearTheMusic.

dB HL at 1000 Hz	65 dB input	80 dB input	95 dB input
15	0	0	0
25	2	1	0
35	8	4	0
45	14	7	0
55	20	10	1
65	28	15	2
75	36	20	3
85	44	24	4

Strategy #3

Use Scotch tape. This is the lowest technology level and is perhaps the easiest to implement clinically. Like the use of a less sensitive microphone (e.g., -6 dB/octave), using a temporary microphone covering such as Scotch tape shifts the hearing aid microphone sensitivity downward by about 10 dB, when three layers of tape are used. The A/D converter is therefore presented with a signal that is 10 dB lower, and the signal can often be within the A/D converter's optimal operating range. There needs to be some trial and error, and the hard-of-hearing consumer can be instructed to play with one, two, or three pieces of tape over both hearing aid microphones. The exact number does depend on the gauge and the brand of the tape. Attenuations of 10 dB, which are relatively flat across the frequency range, have been measured using this clinical "low-tech" approach.

Strategy #4

Change the musical instrument. This is a common strategy used by many musicians. Change to an instrument that has more of its energy in an audiometric region of better hearing. Many violin players have switched to the viola, which is a fifth lower in frequency. For many, this is a simple approach that has extended a musicians' enjoyment of their music for many years. This and other approaches that are acceptable to musicians are discussed in a book from the Association of Adult Musicians with Hearing Loss (Miller, 2011).

Conclusion

Most of the strategies and technologies that have been discussed here are related to the inadequacy of many modern digital hearing aids in handling the higher sound level inputs of music within their operating range. Like most areas of audiology, the realm of music as an input to a hearing aid and the technologies that are available are rapidly changing. New technologies are on the horizon (including the use of widespread post–16-bit architecture) and many similar ones may be implemented by various manufacturers under a score of different names.

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