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Music and Hearing Aids

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ABSTRACT

Music and speech have some differences which include spectral shape, intensity and "crest factors". Most modern digital hearing aids cannot handle the more intense inputs that are characteristic of music. Four new technologies and four clinical strategies will be provided to optimize hearing aids for music as well as for speech. These technologies are designed to circumvent some problems associated with the analog-to-digital conversion process- a major weak point with many modern hearing aids.

1 Summary

One of the misconceptions in the field of audiology is to restrict oneself to the various electro-acoustic and fitting parameters that are delineated in "Specification of hearing aid characteristics" (ANSI S 3.22, 1987, R. 2014). This is the current standard for assessing the quality control of hearing aid function in a test box. ANSI S 3.22 is merely a reporting standard that specifies, along with tolerances, the method to assess a hearing aid for a number of electro-acoustic parameters. These may include, frequency response, OSPL90, gain, distortion, equivalent input noise, and up until recently the attack and release times for compression.

There are no standards for a how a hearing aid should function for a number of differing inputs. ANSI S 3.22 makes no recommendations for whether a hearing aid is optimal or not, for speech, for music, and for non-English languages. It is merely a reporting standard.

In the most recent version of ANSI S 3.22 (and the equivalent IEC standard), attack and release times have been relegated to an optional annex. This does not mean that attack and release times are not important for speech and music, but that modern hearing aid technology has surpassed the need to measure these parameters which were based on the function of a capacitor which is no longer used with digital technology. Attack and release times can be very important for setting modern digital hearing aids for a client, but they no longer appear as important for the ANSI S 3.22 standard.

Similarly, there are some parameters that have never been part of the ANSI S 3.22 standard that are quite important, especially in the realm of hearing aids and music. One such measure includes the ability of the analog-to-digital (A/D) converter to handle inputs in excess of 90-95 dB SPL without appreciable distortion. The literature has referred to this parameter by several names, such as "front end distortion" or the "peak input limiting level". (Chasin and Russo, 2004; Chasin and Hockley, 2014). This refers to the highest "input" sound level that can be transmitted to the software digital algorithm portion of the hearing aid without distortion.

2 Peak Input Limiting Level and Clinical Strategies

The primary element of digital signal processing for hearing aids that has been problematic is the 16-bit architecture that, until recently, has been the mainstay of the hearing aid industry.

The selection of a 16-bit architecture is quite adequate for speech signals as an input to a hearing aid. With 16-bits, the quantization error noise floor is below that which normal hearing people can hear, and the associated dynamic range, in practice, is on the order of 90 dB. (Even though there is a 96 dB "theoretical" dynamic range, the real dynamic range tends to be slightly less due to other engineering design issues such as electrical noise.)

With the highest input sound levels of human speech being less far less than 90 dB SPL, a 16-bit architecture is more than adequate.

However, instrumental music is not constrained by the limits of the human vocal mechanism. Peak levels in excess of 100 dB SPL are routinely measured even for very quiet pieces, and levels in excess of 120 dB SPL are not unheard of. A 16-bit system cannot handle these higher input levels unless some alterations are made to this architecture. Because the "peak input limiting level" is associated with the A/D converter and other "front end" technology, this is prior to any digital software algorithms that may be applied. This issue is therefore considered a hardware issue and not something that can be resolved with software programming. Once an input signal is distorted, no amount of software manipulation can be used to resolve this front end distortion. A website with some audio files demonstrating this issue can be found at www.Chasin.ca/distorted_music.

The field of audiology has responded in several ways to this front end limitation of digital hearing aids: (1) reduce the input to the A/D converter, (2) increase the upper limit of the dynamic range of the A/D converter, and most recently, (3) developing post-16-bit hearing aid architecture with an inherently higher dynamic range.

1. Reduce the input to the A/D converter

Metaphorically this can be thought of as "ducking under a low hanging bridge or door". Given a certain height of a bridge or doorway- the peak input limiting level- some manufacturers of hearing aids have used a form of "analog" compression that reduces the input to the A/D converter (and then digitally re-establishes the original signal once in the digital domain). This "ducking under the bridge" approach has been used, starting about a decade ago by several manufacturers, but is now is the engineering standard that is used by virtually all companies (Chasin, 2017a).

Another innovation that can be thought of as "ducking under the bridge" is to use a hearing aid microphone that is less sensitive to the input signal, at least in some frequency range. The first approach (Chasin and Schmidt, 2009, Schmidt, 2012) was to use a hearing aid microphone that is "less" sensitive to sound below 1000 Hz. This "-6 dB/octave microphone" which reduces the input sensitivity by 6 dB at 500 Hz and 12 dB at 250 Hz, has been commercially available for years. The low-frequency, high sound level sounds of music would not be picked up as well by such a hearing aid microphone and instead would enter the vented or open hearing aid fitting directly to the ear canal. This low frequency sound would not overdrive the front end of the hearing aid as easily as would a hearing aid with a broadband microphone. Several small "patches" need to be made with this approach such as a reprogramming of some of the algorithms and the use of expansion to maintain a low noise floor in this hearing aid.

2. Increase the upper limit of the dynamic range of the A/D converter

Another approach to the resolution of this problem is based on the definition of "dynamic range". The dynamic range is not an absolute number, such as X dB SPL. It is a relative measure in dB (without any suffix) that is the difference between the quietest sound that can be reliably transduced and the highest level sound that can be reliably transduced. This range can indeed be 0-90 dB SPL or also, 15-105 dB SPL. In both cases, the dynamic range is 90 dB but the second one is better suited to the higher levels and peaks characteristic of instrumental music (Chasin, 2014; Chasin and Hockley, 2014; Hockley, Bahlmann, and Chasin, 2010; Hockley, Bahlmann, and Fulton, 2012).

Several hearing aid manufacturers have used "third party" hearing aid circuitry that has "auto-ranging" of the input signal hardware designed into the IC. In some cases these ICs sense the level of the input signal and adjust the lower and upper limits of the dynamic range automatically, and in other cases, use "stacked" A/D converters with one A/D being optimized for speech dynamics, and another being optimized for instrumental music.

3. Post 16-bit hearing aid architecture

The theoretical dynamic range of an "n-bit" system can be expressed in decibels as 20nlog2 or simply, 6n. A 16-bit system would have, at most, a 6 x 16 = 96 dB dynamic range. A 20 bit system would have, at most, a 6 x 20 = 120 dB dynamic range. While this second dynamic range does sound better, modern hearing aid microphones can only handle up to 115 dB SPL – 119 dB SPL (depending on the microphone manufacturer). A hearing aid manufacturer that markets a 24-bit system is really making a statement that is more marketing than clinical reality.

Having said this, a modest increase in the bit depth of a hearing aid architecture can indeed improve the fidelity of the instrumental music, although it would be false to generalize that "more is better". Most hearing aid manufacturers either currently have a post-16-bit architecture hearing aid already in the marketplace, or are planning to have one in the next year or two. It will be difficult to assess clinically whether any improvement would solely be related to this increase in the number of bits since most manufacturers also use some analog compression prior to the digitization stage (Chasin, 2017a). A manufacturer may choose to combine several of these hardware strategies into the same hearing aid platform such as utilizing a less sensitive microphone, analog compression prior to the digitization of a signal, and a post 16-bit architecture. Use of one approach does not obviate the use of another technology.

3 Some Clinical Strategies

Clinically, we have all been confronted with a client who is quite pleased with their hearing aids for speech, but are less than pleased while listening to, or playing music. Given that they may have a hearing aid with a low peak input limiting level, there are some clinical strategies that may be useful.
These strategies are all based on the "ducking under the bridge" metaphor. One can have the client place 4-5 layers of cellophane tape over the hearing aid microphones while listening to, or playing music- depending on the gauge of the tape, this may reduce the input by 10-12 dB thereby allowing the input music to be within the operating range of the A/D converter. Another strategy would be to use an assistive listening device with an external microphone and volume control. While it is true that any mode of input to a hearing aid (e.g., microphone, telecoil, direct audio input) still needs to go through their own A/D converter pathway, having an external microphone will allow the option of reducing the input to optimize the digitized signal. And finally, for those with only a mild to moderate level sensori-neural hearing loss, removal of the hearing aids may be useful- given the higher sound levels of instrumental music, very little or even 0 dB of gain may be required.

4 Smartphone control and Music

Smartphones and associated apps are beginning to be used in conjunction with hearing aids as these can be used to control some aspects of the amplified signal. Currently many manufacturers offer an equalizer that can be used to alter the frequency response- there may be slight adjustments that can be offered in the way of filtering certain frequency regions of the speech and music. While this can be quite useful, especially for the experienced hearing aid user, other types of self-control can be problematic. This is especially the case for setting and altering the maximum output of the hearing aid. Control over this parameter should not be provided to the player or listener of music. The maximum output of the hearing aid is a parameter that needs to be carefully set by the audiologist and verified using equipment that is not available to the general public. Further hearing loss may occur if the hearing aid output constantly exceeds a certain well-defined level, and this is true of speech as well as music.

Chasin, (2017b) discusses some of the limitations of using Smartphone microphones for listening to music (especially in conjunction with hearing aids). These involve the limited dynamic range of the MEMS microphones (on the order of 60 dB) used in Smartphone, an input compressor that may not be disabled on the Smartphone, and the directional characteristics of a Smartphone that may limit the full range of sounds unless held appropriately.

Alexander (2016) and Chasin (2017c) have questioned the longer latencies that some Smartphone manufacturers may provide. While this may not be an issue in listening to recorded music, this can be problematic for live performances. And this can be exacerbated by the use of some wireless transmission protocols such as Bluetooth. Caution needs to be exercised with the use of Smartphone technology, especially when used in conjunction with hearing aids.

5 All of the "other electroacoustic stuff"

The peak input limiting level remains the primary element in addressing which hearing aid would be most appropriate for a hard of hearing musician. There are a number of other electro-acoustic parameters that need to be specified for any one hearing aid fitting in order to optimize the sound for both speech and for music. These include the frequency response, non-linear processing, noise reduction and feedback management, multi-band compression, and frequency compression parameters. Many researchers and clinicians are still investigating these parameters, not only with speech versus music, but also for different languages and different forms of music.

In most cases, the parameters that are set forth to optimize speech in quiet, are similar to those for optimizing music. Following is a brief review of three parameters.

1. Frequency Response

The work of Brian Moore (1996, 2012) and Todd Ricketts (2008) among other have helped to delineate the optimal frequency response for hearing aids given certain audiometric hearing losses. While parts of these studies were not specifically about music, three generalizations could be made: (1) If the hearing loss is mild, a broadband frequency response is optimal; (2) if the hearing loss has a precipitous slope, then a narrower frequency response is optimal; and (3) if the hearing loss is greater than 60 dB HL (greater than a moderate sensori-neural hearing loss) then a narrower frequency response is optimal.

These conclusions are based on well-controlled laboratory experiments and can be explained in terms of severely damaged "cochlear dead regions" where too much amplification may be worse than less amplification. Of importance however, is that the prescribing of an optimal frequency response has more to do with the severity of the hearing loss and not whether the input is speech or music. Subsequently the optimal frequency response for music is similar to that which is prescribed for a person for their "speech-in-quiet" hearing aid program.

2. Amplitude compression

Amplitude compression (also known as Automatic Gain Control or AGC) can be set individually in any number of channels that modern hearing aids may have. This is a form of non-linear processing which serves to reduce the gain for higher level sounds while maintaining sufficient gain for quieter input sounds. The most common form of amplitude compression is known as Wide Dynamic Range Compression (WDRC) and is ubiquitous in the hearing aid industry.

Chasin and Russo (2004) and Moore (2008) have delineated some of the characteristics of amplitude compression for music. Chasin and Russo (2004) was confirmed experimentally by Davies et al. (2007). Similar to the realm of frequency response, it is the nature of damage to the cochlea of the hearing aid wearer and not whether the input is music or speech that defines the parameters of amplitude compression. The optimal amplitude compression for music is similar to that which is prescribed for a person for their “speech-in-quiet” hearing aid program.

3. Frequency compression
Frequency compression in many of the various formats that are commercially available, can be quite useful in the clinic. They can be a very useful tool to avoid cochlear dead regions that may be related to severe inner hair cell damage (Moore, 1996). Frequency compression serves to relocate higher frequency sound energy that may be difficult to hear, to a lower frequency, presumably healthier region of the cochlea. While this may be true for speech, it does not follow that this would also be the case for instrumental music. A “gain-reduction” strategy in the offending frequency region(s) would be the most appropriate clinical approach rather than any alteration in the spectrum of the amplified music.
For speech, frequency compression works well because it is only the higher frequency broadband sibilant sounds such as ‘s’ and ‘sh’ that are reduced in frequency and would therefore be represented in a healthier (lower frequency) region of the cochlea. Altering the center frequency of a broadband (say from 4000 Hz to 3500 Hz) would have only a slight alteration in perception.
However, changing the important relationship among harmonics in music can be disastrous. The tenth harmonic of A [440 Hz] needs to be at 4400 Hz for a stringed (or any half wave length resonator) musical instrument, and not at 4100 Hz.
Gain reduction (which maintains the exact frequencies of the input spectra) would be the appropriate clinical strategy to avoid any dead cochlear regions when listening to, or playing music. (Chasin, 2016). Frequency compression is useful for processed amplified speech, but not for music.

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