music TO THEIR ears

Using Clinical Strategies and Technical Innovation to Help Increase the Enjoyment of Music

BY MARSHALL CHASIN

Mrs. Jones, age 70, loves her hearing aids for listening to her grandchildren and friends, but she is less than pleased with them when it comes to listening to music. Mrs. Jones is not a musician, but likes to listen to music on occasion and to attend musical theatre productions from time to time.

Mrs. Smith, age 70, is a semi-retired oboe player with the local philharmonic and just can’t seem to hear the conductor’s instructions during rehearsals. In addition, she feels that the notes of her instrument sound “rather thin” in the upper ranges.

Both of these clients may walk in to our clinics and, despite the differences in their requirements to hear music, both may benefit from some rather similar clinical strategies and amplification technologies. And, as this article will show, optimizing hearing aids for music will improve the use of hearing aids for listening to speech.
Music and Speech as Inputs to Hearing Aids

Music and speech have many similarities and many differences. Both music and speech can have sounds that have rich harmonic structure with specific resonances (formants). Speech has sonorants—vowels, nasals, and liquids—that have regular harmonic structures and well-defined formant peaks. Music may include stringed, woodwind, and brass instruments with acoustics similar to those of the human vocal tract—rich harmonic structures with well-defined resonances. Speech has higher-frequency obstruents—affricates, fricatives, and stops—that have no harmonic structure and tend to be rather broadband. Music includes percussive instruments such as drums, cymbals, and tympani that also are devoid of harmonic structure and have energy spectra that can be quite broadband.

In contrast, speech is a relatively low-level signal, as compared with many forms of music. The long-term average speech spectrum (LTASS) of speech at one meter is approximately 65 dB SPL RMS (root mean square) with peaks around 12 dB higher (known as the crest factor). While there is no one “average LTASS” for music, the levels of music easily can exceed 90–100 dBA with peaks, or the crest factor, of around 18 dB higher. Both the differing sound levels and crest factors for speech and music have implications for the fitting and the design of hearing aids that can optimally transduce speech and music. TABLE 1 shows the sound levels of a selection of commonly played musical instruments (adapted from Chasin, 2006).

Peak-Input Limiting Level and A/D Converter

ANSI/ASA S3.22 (2003), the hearing-aid testing standard commonly used in North America, is under review and will be harmonized with the European IEC 60118-7 standard used in Europe and South America. These are both reporting standards, not performance standards. Once the harmonization is approved by various working groups, ANSI S3.22 (or IEC 60118-7) will focus only on the way the electroacoustic output of some hearing aids should be reported (such as on a specification sheet). The new harmonized standard will not say that a hearing aid should perform in a certain way.

Understandably, there is not a one-to-one correspondence between how a hearing aid performs in the real world and how it may be assessed in a hearing-aid test box. For example, in the new harmonized hearing-aid standard, measures of attack and release time for compression circuits have been removed, yet clinically we know that attack and release times can be quite important, especially when it comes to hearing in noisy situations.

One of the most important parameters not found in the new harmonized reporting standard (or in any of its predecessors) is the peak-input limiting level. This is the

<table>
<thead>
<tr>
<th>Musical Instrument (measured at 3 meters)</th>
<th>dBA</th>
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<tbody>
<tr>
<td>Normal piano practice</td>
<td>60–90</td>
</tr>
<tr>
<td>Violin/viola</td>
<td>80–90</td>
</tr>
<tr>
<td>Vocalist</td>
<td>70–85</td>
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<tr>
<td>Oboe</td>
<td>74–102</td>
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<tr>
<td>Saxophone</td>
<td>75–110</td>
</tr>
<tr>
<td>Piccolo</td>
<td>96–112</td>
</tr>
<tr>
<td>Amplified guitar</td>
<td>105–112</td>
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</table>

Note: Measured on the horizontal plane at a distance of 3 meters. Data for each instrument includes more than 30 samples (adapted from Chasin, 2006).

FIGURE 1. A metaphor of a low-hanging bridge: If the input is at too high a level, distortion will be created by exceeding the capability of the “front end” and A/D converter. The bridge height either has to increase or the (music) input has to decrease.
“ceiling” of the most intense sound that can be transduced through a hearing aid, just after the microphone. Modern hearing-aid microphones can transduce levels of 115 dB SPL (and have been able to do this since the late 1980s), but the current 16-bit analog-to-digital (A/D) converters used in modern digital hearing aids have a limitation in their dynamic range of 92–96 dB. Sounds input into hearing aids in excess of the operating capability of modern A/D converters will cause significant distortion. Once distortion occurs so early in the hearing-aid circuitry, no amount of software adjustments or reprogramming that occurs later in the system will improve the signal.

The peak-input limiting level can be thought of as a ceiling or maximum level that can get into a hearing aid without distortion. Modern day A/D converters typically are designed to be able only to receive inputs on the order of 96 dB SPL or less, and, unless some special technology or clinical strategy is used, such high-level inputs would have associated high distortion. Metaphorically, the input either has to duck under this low ceiling, or the ceiling needs to be increased in some manner. This analogy is shown in Figure 1.

Average speech levels are on the order of 65 dB SPL RMS and, even with the 12 dB higher-level peaks, the input to modern digital hearing aids will not exceed the peak-input limiting level of the hearing aid. The same cannot be said of music as an input to a hearing aid. Even medium levels of music at 85 dB SPL RMS, with a crest factor of 18 dB, arithmetically imply a peak input in excess of 100 dB SPL. Music will overdrive the “front end” (A/D converter) of hearing aids. The hearing-aid setting for gain, frequency response, or output doesn’t matter; this occurs before any programming. As a result, programming a “music program” will be a clinical waste of time, unless the peak-input limiting level issue is resolved.

An experimental hearing aid has been constructed that allows the peak-input limiting level to be altered from 115 dB SPL (the limit of modern day hearing-aid microphones) to 105 dB SPL, 96 dB SPL, 92 dB SPL, and then back to 115 dB SPL. All other parameters have been left constant, including the gain, the output, compression features, and the frequency response.

Recordings were made of speech at 60 dB SPL and music at 90 dB SPL and at 100 dB SPL. These files demonstrate that, once the music is so distorted by the front end of a hearing aid, no amount of software programming, such as a “music program,” will be able to improve things. The audio files can be accessed online at www.chasin.ca/distorted_music.

The Crest Factor and Implications for Speech

The crest factor, as a waveform in the time domain, is shown in Figure 2. This is the difference in decibels between the long-term average RMS signal and an instantaneous peak in the signal. The human vocal tract is a highly damped structure—soft cheeks, tongue, soft palate, narrow opening (velo-pharyngeal port) to the nasal cavity, narrow nostrils, and frequently narrowed lips. In short, the peaks are quite damped relative to the average speech output. This difference, with a 125 msec window of analysis can be shown to be 12 dB. In contrast, musical instruments are hard-walled structures with relatively low levels of damping. Subsequently, the peaks are higher than those found in speech, and tend to be on the order of 18 dB.

Dunn and White (1940) found that the crest factor for speech was 12 dB and, more recently, this number has been verified by Cox et al. (1988), using a 125 msec window. The 125 msec (1/8 of a second) window is used...
because it is approximately the shortest-level signal that we can perceive; it is a cochlear and central “time constant.” However, when it comes to inputs and crest factors for electrical and mechanical components such as A/D converters, there is no such cochlear and central limitation. Microphones and A/D converters can receive signals on the order of 20 msec, so the 125 msec window of analysis no longer makes sense. Figure 2 shows the measured crest factors of some speech and music samples as a function of a shorter window of analysis. Note that, with these shorter analysis windows, crest factors of 16 dB or greater can be measured for speech, and crest factors in excess of 20 dB can be measured for music.

This larger crest factor has implications for loud speech uttered by a hard-of-hearing individual. While the level of the LTASS at one meter is 65 dB SPL RMS (with higher-level peaks), the level of a hard-of-hearing person’s own voice may be on the order of 84 dB SPL. Adding in the crest factor of speech at the level of their own microphone with an arbitrarily short window of analysis, the input levels of a person’s own speech can easily exceed the peak-input limiting levels of hearing-aid front ends. Modern digital hearing aids using the commercially available 16-bit architecture will cause a hard-of-hearing person’s own voice (specifically, the lower-frequency sonorants such as the vowel /a/) to be distorted.

To resolve the distortion, four clinical strategies—approaches that can be implemented in the clinic or in suggestions to the clients—appear below, along with four innovations in technology that offer a response to the low peak-input limiting level found with 16-bit architecture.

### Four Clinical Strategies

Mrs. Smith and Mrs. Jones, mentioned at the start of the article, arrive at your clinic. Both have a presbycusic type of hearing loss. One likes to listen to music on occasion, and the other is a musician who has some difficulty hearing the conductor during rehearsals and can no longer appreciate the higher-frequency components of music. Both are quite happy with their current amplification for speech, but both feel that their hearing aids fall short when music is involved. What can be done or suggested for these (and other) clients?

**Clinical Strategy #1: Turn down the input and turn up the aid volume (if necessary).**

If the excessive level of the input to the hearing aid causes distortion of the A/D converter, turn down the input, if at all possible. If traveling in a car, turn down the level of the sound system and, if necessary, turn up the level of the hearing aid to compensate. The output limiting of the device will ensure that the output will be the same, but the input will have been reduced to a level well within the operating range of the front end of the hearing aid.

**Clinical Strategy #2: Remove the hearing aid to hear 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 HL sensorineural hearing loss at 1000 Hz, while a person may require 45 dB of gain for certain speech sounds, the individual may only require several decibels of amplification for many types of music. The best strategy for many hard-of-hearing people may be to simply remove their hearing aid when listening to or playing music.

**Clinical Strategy #3: Use tape to cover the hearing-aid microphone(s).**

This strategy, using the lowest level of technology, is perhaps the easiest to implement clinically. Using a temporary microphone covering, such as transparent tape, attenuates the sensitivity by about 10 dB for three or four layers of Scotch brand tape (made by 3M Corporation, Minneapolis, MN) (Chasin, 2010). The A/D converter is

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**TABLE 2. FIG6 Fitting Formula at Various Input Levels**

<table>
<thead>
<tr>
<th>dB HL at 1000 Hz</th>
<th>65 dB input</th>
<th>80 dB input</th>
<th>95 dB input</th>
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<tbody>
<tr>
<td>15</td>
<td>2</td>
<td>1</td>
<td>0</td>
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<tr>
<td>25</td>
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<td>1</td>
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<td>3</td>
</tr>
<tr>
<td>85</td>
<td>44</td>
<td>24</td>
<td>4</td>
</tr>
</tbody>
</table>

Using the FIG6 fitting formula (www.etymotic.com), at input levels of 65 dB (quiet music), 80 dB (medium-level music), and 95 dB (loud-level music), for an individual with a given hearing loss at 1000 Hz, very little gain may be required for loud music, even for moderate-level hearing losses. A strategy for some people would be to simply remove their hearing aids while listening to and playing music.
therefore presented with a signal that is 10 dB less intense and often can be within its optimal operating range. This approach requires some trial and error; the hearing-aid user can be instructed to play with one, two, or three pieces of tape over both hearing-aid microphones. The exact number depends 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.

Clinical Strategy #4: Change the musical instrument.
While I would never recommend (and have never recommended) that a musician stop playing music, a common (and acceptable) strategy used by many musicians is to play a more bass-oriented instrument—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 musicians, this simple approach has extended their enjoyment of their music and, in some cases, their career, for many years.

Four Technical Solutions
The hearing-aid industry has responded to the problems that occur when a high-level signal, such as music, is presented to a modern hearing aid. Four technological approaches that are commercially available in the hearing health-care industry are listed below. This is not an exhaustive list, but includes technologies that, to date, have been found to be clinically useful.

Technical Solution #1: K-AMP analog processing
Although no longer widely commercially available, the analog K-AMP was first manufactured a quarter of a century ago (Killion, 1988, 1993) and was the mainstay for musicians from its inception until the advent of second-generation digital hearing aids. The K-AMP was designed with the ability to transduce very intense inputs with virtually no distortion. And, because it is analog, there is no A/D converter to be overdriven. Recently, a Personal Sound Amplification Product (PSAP) has become available with the analog K-AMP at its heart. This is the Bean from Etymotic Research, Inc. (www.etymotic.com). To my knowledge, this is the only PSAP that can handle the higher levels of music without distortion.

Technical Solution #2: Change where the dynamic range of the hearing aid operates
This approach is based on the actual definition of dynamic range. The theoretical dynamic range of current 16-bit hearing aids is 96 dB (and not 96 dB SPL). It is a range between the least intense signal and the most intense signal, and is 96 dB (without any dB scale). In this approach, when the circuit is implemented (by the selection of a music program), it transduces all inputs between 15 dB SPL and 111 dB SPL—still a 96 dB dynamic range—but it has been shifted up by 15 dB. Levels of 111 dB SPL can be transduced distortion-free and are much better for the listening to, or the playing of, live music. (for example, Hockley, Bahlmann, and Chasin, 2010).
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Many similar technical solutions may be implemented under a score of different names.

Technical Solution #3: The use of “-6 dB/octave microphone” instead of a broadband microphone
This sensitivity in the lower-frequency region has been shown to be quite beneficial with many forms of music (Schmidt, 2012). 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, since 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 using a -6 dB/octave microphone is that it increases the internal noise floor of the hearing aid. However, expansion can be used successfully in its maximum setting to completely offset this change in noise floor.

Technical Solution #4: Reduce the music input, and then digitally re-expand the signal
This technology is similar to ducking under a low-hanging doorway and then standing up after you pass through it. Depending on the implementation, there may be an analog compression of the signal prior to the A/D converter, and then a digital re-expansion after the A/D converter. The digitized signal is identical to the initial analog input, but in a distortion-free digital format. Some manufacturers have used a form of the transformer effect (maintaining a low noise floor), and others are contemplating the analog/digital strategy described above (See, for example, Chasin, 2014).

It is important to note that none of these strategies or approaches are software adjustments. Software changes occur after the A/D converter; once a high-level signal is distorted by a poorly configured front end, no amount of software manipulation will ameliorate the situation. Fitting software modifications are simply not the approach that should be taken when dealing with the more intense components of music.

Conclusion
Most of the strategies and technologies discussed here are related to the finding that most currently available digital hearing aids cannot handle the higher-level inputs of music within their optimal operating range. A study of crest factors relevant to the input of a hearing aid, rather than the output to the human auditory system, may have far-reaching implications for music listening. Like most areas within the field of audiology, music as an input to hearing aids, and the available related technologies, are part of a rapidly changing realm. New technologies are on the horizon, and many similar technical solutions may be implemented by various manufacturers under a score of different names.

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Notes
References


