



**HEARING
AIDS & MUSIC**

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Some features of instrumental music and speech

Music and speech are rather similar in that they both are vibrations in the air, and whose energy is constrained to a well-defined bandwidth. Musical cues and fundamentals do extend down into a lower frequency region than does speech, and speech has no energy below the fundamental frequency which is about 125 Hz in males, but both have a rather large bandwidth. Both music and speech have peaks in their spectra and times where there are silences. Both have rapid changes from loud to soft- speech being governed by the mechanics and neurology of the vocal tract, and music just being governed by the musical instrument being played.

In short, music and speech can have similar, albeit not identical, spectra and time based waveforms. The differences lay in their intensities and their crest factors. Instrumental music—even quiet music—can be in excess of anything produced by the human vocal tract. And crest factors can become quite significant as well. Table 1 shows the typical outputs in dBA of a number of musical instruments. In most cases these values were obtained from a horizontal distance of 3 meters. In one case, the values have been measured near the musicians’ ear. These are all from Chasin (2006).

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

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 (2006). Used with permission.

The crest factor is the difference between the peak of a signal and its average (or root mean square, RMS). For a pure tone it is 3 dB. For speech it is usually taken as 12 dB. That is, for speech, the difference between the peak output in the time based wave form is about 12 dB more intense than the average of the speech signal. We deal with this crest factor concept every day when we assess the function of hearing aids in a test box according to ANSI S3.22 (2003). The reference test gain is the OSPL90- 77 dB. Well, 77 dB is 65 dB (average speech) + 12 dB. And 12 dB is the crest factor of speech. This value is based on years of research going back to Sivian and White (1933) and also Cox et al. (1988) who found that peaks for speech were on the order of 10-12 dB above the estimated long term speech spectrum. In contrast, musical instruments can have crest factors in excess of 20 dB and this has implications for designing and setting hearing aids. However, as we will see later, this crest factor data may not be as clear cut as first suspected. Table 2 shows some crest factors that have been measured in some common types of music and also from some speech samples.

Stimulus	Peak Amplitude -Total RMS Power	Crest Factor
Speech #1	-0.92 - -21.97	21.05
Speech #2	-5.53 - -17.99	12.46
Speech #3	-3.65 - - 17.6	13.95
Music #1	-8.62 - - 19.35	10.73
Music #2	-5.0 - - 15.28	10.28
Music #3	-0.98 - - 22.65	21.67
Music #4	-2.45 - - 21.88	19.43

Table 2: Using a 125 msec analysis window, the differences between the peak amplitude and the total RMS power are calculated to give the crest factor for a selection of speech samples and music samples.

Hearing aid elements and music

The hearing aid has several components, each with their own set of optimal operating characteristics. The hearing aid microphone has been able to transduce 115 dB SPL with virtually no distortion, at least since the late 1980s. The hearing aid receiver has gradually improved over the years and with an appropriate acoustical load (and acoustical plumbing), can transduce significant high frequency output without sacrificing battery life too much. The same can be said about the digital to analog (D/A) converter- both music and speech are well within its operating range. A difficulty arises when we are talking about the front end of the hearing aid- the analog to digital (A/D) converter.

The current implementation of the A/D converter, given a 16 bit architecture, can only handle a 90-96 dB dynamic range (the range between the least intense and the most intense signal). The actual dynamic range of the A/D converter for a 16 bit system is theoretically 96 dB however frequently engineering and design decisions need to be made that reduce the effective dynamic range slightly. A 96 dB dynamic range is quite sufficient for the range of speech sounds that are encountered but can easily fall short with many forms of music. With the joint conspiracy of both intensity and a high crest factor, instrumental music can be far in excess of 96 dB SPL. Adding the various values from Table 2 to levels of 80-90 dBA clearly shows that a problem will exist. (Or simply add values from Tables 1 and 2 will demonstrate that peak outputs can easily be in excess of 96 dB SPL). A quiet piece of music played at 80 dBA with instruments that have a 20 dB crest factor suggests that the peaks of the input will be on the order of 100 dB SPL.

Crest factors revisited

Sivian and White (1933) and Cox et al. (1988) both used a sample length of 125msec for their research. This makes sense because the 125 msec chunk is fairly close to our perceptive limits so using smaller temporal chunks would not be meaningful. However, when it comes to hearing aids and music, we are talking about the input to the hearing aid, and not the output to the hearing aid wearer. The A/D converter is not “aware” of the limits or operating range of the human ear- this occurs much later in the transduction process.

To make matters worse, consider the well-agreed-upon crest factor for speech: namely 12 dB. This value is the result of a

125 msec analysis window. Table 3 shows the varying crest factors measured for the same speech sample with varying time windows for its analysis. Indeed, a crest factor of 12 dB is obtained when using a 125 msec or longer window, but higher crest factors than 12 dB are obtained when using shorter windows of analysis. If a hard of hearing person’s own voice impinges on their own hearing aid microphone, regardless of whether humans can perceive short time constant inputs, the A/D converter will be subjected to a signal that has a high intensity plus a “higher than 12 dB” crest factor- a level that may also force the A/D converter to operate in its non-optimal range. That is, the analyzing window for the stimulus should reflect the input parameters for a hearing aid and not the output parameters for our auditory system.

Some technologies and strategies to handle the more intense inputs of music

There are a number of creative strategies and electro-acoustic techniques that are being used in the industry to resolve this “high level input- front end” problem for music. Following is a list of eight of the more commonly used approaches that have found it be useful. The first four are strategies that can be shared with patients who already have hearing aids that are optimal for speech but require additional fidelity for listening or playing of music. The next four are technical innovations that are currently available in hearing aids or can be implemented within a clinical setting.

Strategy #1: Turn down the input (stereo) and turn up the aid volume (if necessary). If the excessive level of the input to the hearing aid does cause 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 same, but the input would have been reduced to a level that is well within the operating range of the front end 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 4 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 sensori-neural hearing loss at 1000 Hz, while a person may require 45 dB gain for certain speech sounds, they may only

Stimulus	500	400	300	200	125	100	50	25
Speech #2	12.46	12.48	12.46	12.45	12.46	13.22	16.68	16.68

Table 3: Crest factor calculations for speech stimulus #2 measured with varying time analysis windows from 500 msec, down to 25 msec.

require several decibels of amplification for many types of music. The best strategy for many hard of hearing consumers may be to simply remove their hearing aids when listening or playing music.

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

Table 4: 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. www.hearinghealthmatters.org/hearthemusic blog, Chasin, 2011). Downloaded March 19, 2012.

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 its ability to transduce sound downwards by about 10 dB for three layers of Scotch tape. The A/D converter is therefore presented with a signal that is 10 dB less intense and can often be within its 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.

The above strategies are just a few of the many that have been found to be useful over the years. The Association of Adult Musicians with Hearing Loss (www.AAMHL.org) is an

organization of hard of hearing and deafened musicians as well as interested hearing health care professionals who work with those in the performing arts. In addition to their very active listserv blog, they have recently come out with a book entitled, “Making Music with a Hearing Loss” edited by Cherisse W. Miller (2011). Chapter 4 of that book is aptly entitled “Personal Stories and Strategies” where many musicians talk about what works best for them. Copies of this excellent book can be obtained through the AAMHL.org site.

The next four sections are about hearing aid technologies that have been shown to be quite useful. This review is not meant to be technically complete. In some cases I have mentioned actual manufacturer names that are associated with each of the technologies and the reader is encouraged to contact the respective manufacturer for more information. This is not an exhaustive list, but ones that I have personally found, to date, to be useful. Other clinicians may have found other approaches to also be useful.

Technology #1. The analog K-AMP has been available since 1988 and through the 1990s and 2000s has been the mainstay for musicians. It was designed with the capability of being able to transduce very intense inputs with virtually no distortion. And because it is analog, there is no A/D converter to be overdriven. The K-AMP is still commercially available (at least in the United States) and is marketed through General Hearing (www.generalhearing.com). I frequently have sent many musicians south of the Canadian border to purchase the K-AMP and the function today is still on par, if not better, than some of the other technologies that have been designed with the more intense components of music in mind.

Technology #2. Head Room Expander (HRX) is a tradename of Sound Design (and now with the recent purchase, is owned by ON Semiconductors). 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 and this has the by-product of always ensuring that the levels of intense music that actually reaches the A/D converter is always within the operating range. It is a bit like ducking under a low hanging door way. HRX reduces the input (like ducking under the door way) and then re-establishes its normal level after the A/D components (like standing up again). In this way, HRX can provide distortion free input of the more intense components of music. Those who are interested should contact the various manufacturers directly to determine whether HRX is used in any of their hearing aids. The Digi-K, which is the digital successor of the K-AMP, uses this approach and again, is

available from General Hearing. The HRX technology is also widely used by other manufacturers, perhaps without being explicitly mentioned in their product literature.

Technology #3. “Live Music Plus” is a proprietary technology that is available from Bernafon Hearing Instruments (www.bernafon.com). This is a clever approach that is based on the actual definition of dynamic range. The dynamic range over modern 16 bit hearing aids is not 96 dB SPL. It is a range between the least intense signal and the most intense signal that is 96 dB (without any scale). The Live Music Plus technology when implemented 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, as the name of the technology suggests, is ideal for the listening to, or the playing of, live music.

Technology #4. The use of “-6 dB/octave microphone” instead of a broadband microphone has been shown to be quite beneficial with many forms of music. 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 draw-back of using a -6 dB/octave microphone is that it does increase the internal noise floor of the hearing aid. However, expansion can be used successfully in its maximum setting to offset this change in noise floor. Although this can be implemented by any manufacturer, the only one to date that has implemented it is Unitron Hearing

(www.unitron.com) and they should be contacted directly for more information.

Of importance is that none of these strategies or approaches are in, or can be altered by software adjustments. Software changes occur after the A/D converter and once an intense signal is distorted by a poorly configured front end, no amount of software manipulation will ameliorate the situation. Software modifications are simply not the approach that should be taken when dealing with the more intense components of music.

General recommendations for an “optimal hearing aid for music”

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

Recommendation #1- Similar 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. This was perhaps the case in the past because some circuits used a peak detector versus an average or RMS detector. Because of the differing crest factors of music and speech this threshold kneepoint for activation should have been different for the two stimuli, but with the current trend to design primarily with an RMS or average level detector, any differences are obviated. The use of the WDRC circuitry is primarily to re-establish normal loudness growth due to outer hair cell damage, and indeed that what it does. The use of this circuit is primarily then for damage of our auditory systems rather

How to expand your business with musicians



Musicians are interesting people—they frequently cancel their appointments at the last minute; are chagrined that we don’t fully appreciate that their tinnitus is F# and not F; and sometimes don’t pay their bills on time (at least until their next gig). But, if you still want to see musicians in your practice, here is how:

Start by giving a talk or two at the local College or university music school. Offer to give a talk during the intermission to the musicians at the local philharmonic, and leave your business cards at local music stores. Write a lay article or two in the local newsletter in your community, and then sit back and wait.

You won’t have to wait long. First the rock and jazz bass players come in, then the drummers, next the vocalists and finally the lead guitar—it’s a pretty universal order. In the classical domain woodwinds and violins are the first to show up at your door, then the other instruments, and finally, if at all, the brass players show up.

Whatever the order, there are some things that we can offer musicians- musicians’ earplugs, in-ear monitors, and of course, the subject of this article, hearing aids for music.

than the nature of the input stimuli per se. Davies-Venn et al. (2007), “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 noise” program. This is 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, however as seen in Table 2, this can vary significantly. 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 and, assuming similar WDRC parameters for speech and music, the gain should also be 6 dB less intense than for the “speech in quiet” program setting.

Recommendation #3. Bandwidth for a music program. Examining the work of Moore et al. (2011) and Ricketts et al. (2008) several general recommendations can be made. If the hearing loss is mild (and at most up to a moderate level) then a broader bandwidth for music 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. Table 4 shows some suggested frequency responses from a number of manufacturers, but these should only be for those with a relatively flat audiometric configuration and a mild to, at most, a moderate sensori-neural hearing loss.

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 puretone 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. Siemens Hearing Instruments limits their feedback system to signals over 2000 Hz, and

Oticon Hearing Aids limits their system to signals over 1500 Hz. Both are reasonable solutions to an otherwise problematic situation.

Conclusion

Most of the strategies and technologies that have been discussed are related to many hearing aids not being able to handle the more intense inputs of music within their operating range. A study of crest factors that are relevant to the input of a hearing aid, rather than the output to our auditory systems, may have far reaching implications for speech as well as music, and this is especially true of a hard of hearing person’s own voice at the level of their own hearing aid microphones.

Like most areas of the field of audiology the realm of music as an input to hearing aids and the technologies that are available is a rapidly changing one. New technologies are on the horizon and many similar ones may be implemented by various manufacturers under a score of different names. ■

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