Music and hearing aids

By Marshall Chasin

Hard-of-hearing musicians have long complained about the poor sound quality they experience while playing their instruments or when listening to music through hearing aids. Indeed, many non-musicians also complain of the reduced sound quality of music heard through their personal amplification.

This article is a study of one electroacoustic parameter that, together with three other less important parameters, delineates the optimal amplification scheme for hard-of-hearing musicians and for others who simply like to listen to music. In addition to the experiment reported here, this study is based on clinical experience with more than 350 hard-of-hearing musicians who have been successfully fitted with hearing aids over the last 15 years at the Musicians' Clinics of Canada (www.musiciansclinics.com).

Traditional approaches by the hearing aid industry to optimize a hearing aid or a program within a hearing aid for music are varied. Some manufacturers have sought to reduce the low-frequency amplification and output; others have tried to increase this gain and output. Still other manufacturers have used a strategy of increasing mid-frequency gain and output to optimize the "long-term" spectrum of music.

These approaches have met with only limited clinical success. One reason is that music spectra, unlike those of speech, are quite variable. Another reason is that the input to the hearing aid of speech differs from that of music.

HOW SPEECH AND MUSIC DIFFER

The most intense components of speech are the "low back" vowels. e.g., [a] as in "father." At the level of the listener's ear, these sounds, even if shouted, rarely exceed 85 dB SPL and are more typically 75-82 dB SPL. In contrast, the more intense elements of music measured at the same ear level location are on the order of 100-110 dB SPL, with occasional peaks about 118 dB SPL. This is true not only of rock, but of classical music as well.

Since most engineers have had speech input in mind when designing hearing aids, it is understandable that the "peak input limiting level" of hearing aids has been set to about 85 dB SPL. Peak clipping or limiting may occur at various points in a hearing aid, but if it occurs before the gain-controlling elements, or the A/D converter, adjusting hearing aid settings will not prevent it. Hearing aids are normally designed so that this does not happen for the peak levels encountered in speech, but the peak levels encountered in music may well cause clipping in the early stages of the hearing aid amplifier. This parameter is not found on hearing aid specification sheets.

The usual peak input limiting levels are very reasonable for speech since the most intense components are less than 85 dB SPL. Anything of greater intensity is presumed not to be speech and is therefore limited. The peak input limiting level is not reported on ANSI hearing aid specification sheets, so one needs to speak with someone in the

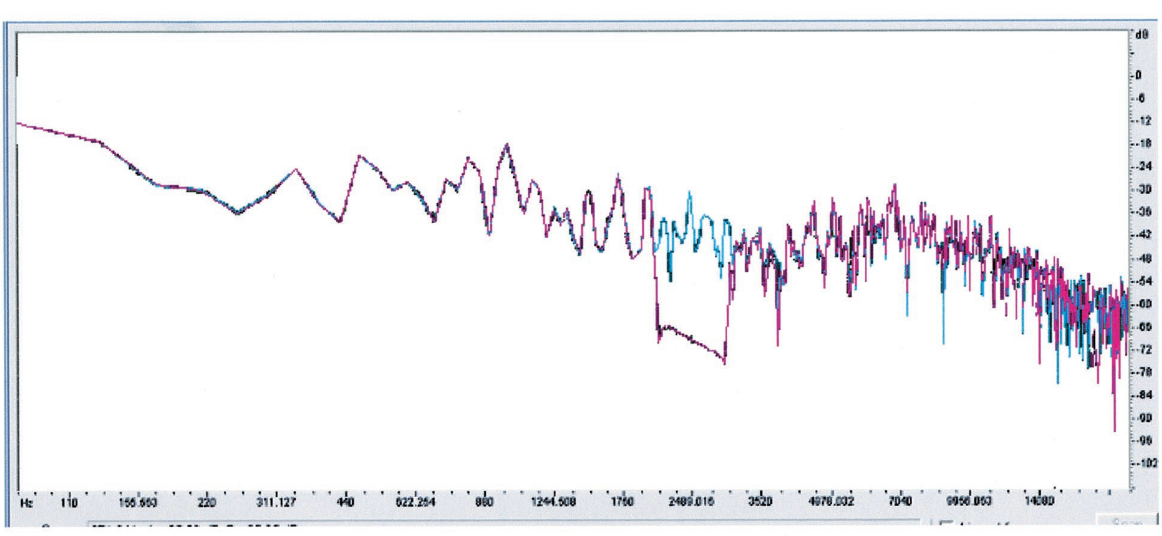

Figure 1. *Through the use of Cool Edit 2000, a spectrum of intense clarinet music was measured and a 30-dB "square" notch was created between 2000 and 3000 Hz. The degree of "fill-in" in the notch is a measure of fidelity. Perfect fidelity would be 0 dB. Specifically, perfect fidelity would be signal–distortion +30 dB (since the notch is 30 dB deep).*

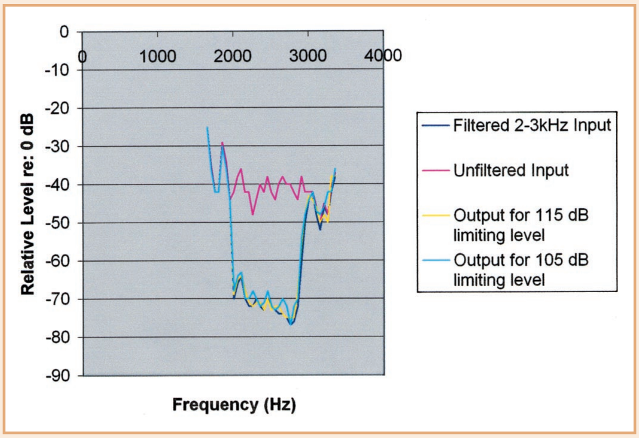

Figure 2. *Showing only the notched portion of the input and output spectra, data are shown for peak input limiting levels of 115 and 105 dB SPL. Note that there is very little difference between the filtered input and the two output conditions, implying fairly good sound reproduction.*

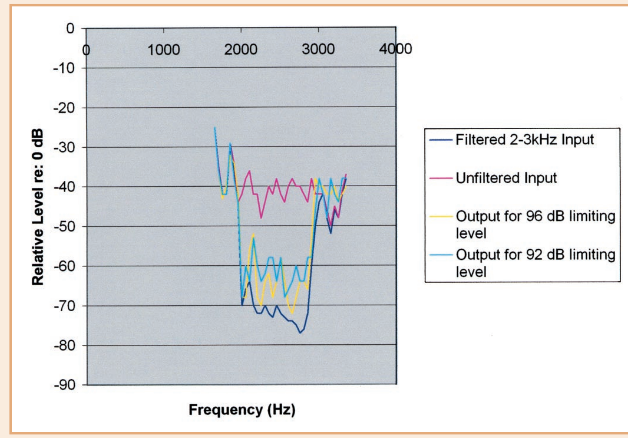

Figure 3. *Outputs for peak input limiting levels of 96 and 92 dB SPL. Note the "fill-in" in the input spectrum notch indicating poorer sound reproduction.*

manufacturer's engineering department to find out what this level is for a particular hearing instrument.

A design rationale with an 85-dB peak input limiter level works well for speech of all languages. However, it can cause a problem for music. This raises an important question: For transducing music, should the "peak input limiting level" of the hearing aid be elevated above 85 dB SPL to a level more in line with the inputs typically found in music? Modern hearing aid microphones are able to transduce up to about 115 dB SPL with limited distortion.

VARYING THE PEAK

To study this issue, we constructed a wearable, experimental hearing aid in which the peak input limiting level could be set at any of four discrete steps: 115, 105, 96, and 92 dB SPL. The gain, frequency response, and output of these four conditions were within 3 dB, regardless of peak input limiting level.

We used measures of distortion (signal-to-distortion ratio) and patient quality judgment scales. Many electroacoustic features, especially in non-linear devices, can affect distortion measurements. Cross-correlation (and auto-correlation) has been used widely in this subject area, but, as Kates et al. have pointed out, this is valid only if the amplification scheme is linear.[1,2]

Kates demonstrated that for non-linear hearing aids, a "notch paradigm" works better.[3] Specifically, he showed that "debris or filling-in" of a spectrum with a well-defined notch could be used as a measure of fidelity. If there is no difference (or "fill-in") within the notch between the input and the output of a hearing aid, then this could be construed as perfect fidelity.

While Kates suggested the notches found in a comb filter,[1] such notches can also be created by using software such as Cool Edit 2000 (see www.syntrillium.com). The "signal-to-distortion" ratio can then be measured. A signal-to-distortion ratio of 0 dB means no fill-in of the notch in the output spectrum, which implies perfect fidelity. More negative value of signal-to-distortion ratio (e.g., –10 dB) implies worse fidelity.

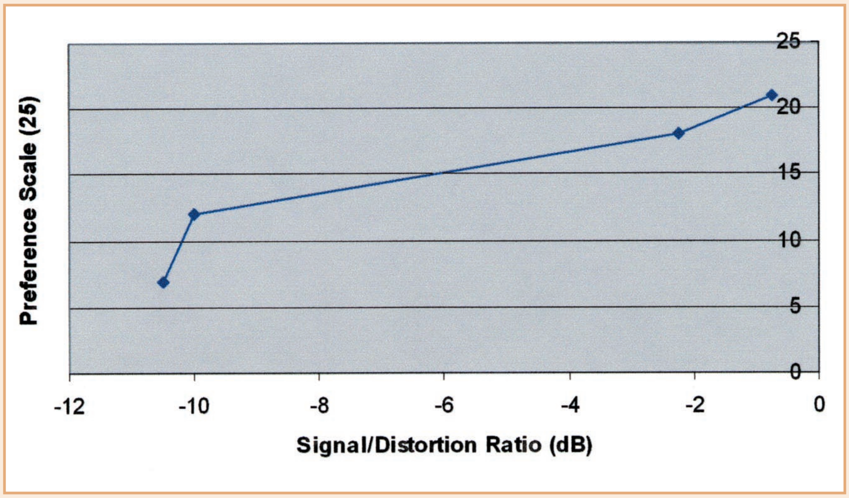

Figure 4. *The sum of the five five-point, perceptual attribute scales for the 23 subjects, plotted against the signal-to-distortion ratio for the four peak input limiting levels [sum for preference scales] (115 dB SPL [21], 105 dB SPL [17], 96 dB SPL [12], and 92 dB SPL [7]).*

Table 1. *Comparison of peak input limiting level (dB SPL) with measured signal-to-distortion ratio (dB). A 0-dB ratio is perfect fidelity. These data were measured in a 2-cc coupler.*

Peak input limiting level (dB SPL)	Signal-to-distortion ratio (dB)
115	-0.75
105	-2.25
96	-10.0
92	-10.5

Figure 1 shows the original input spectrum with a 30-dB square-shaped notch created between 2000 and 3000 Hz. Specifically, the "signal" was measured by eight points on either side of the notch (four lower and four higher) and by eight "distortion" points in the notch. Since the notch was 30 dB deep, we added 30 dB to the distortion average, making perfect fidelity a 0-dB signal-to-distortion ratio.

To obtain measures of sound quality, five five-point, perceptual scales relevant to music were used. This approach is a modification of the work of Gabrielsson and colleagues[4,5] and has been used extensively in the hearing industry (e.g., Cox and Alexander[6]). Specifically, hard-of-hearing musicians were asked to rate the sound from 1 (poorest) to 5 (best) on five perceptual scales: Loudness, Fullness, Crispness, Naturalness, and Overall Fidelity. A perfect perceptual reproduction score was 25 points (5 X 5 scales).

Subjects were given the following definitions of the five perceptual parameters:

Loudness: "The music is sufficiently loud, in contrast to soft or faint."

Fullness: "The music is full, in contrast to thin."

Crispness: "The music is clear and distinct, in contrast to blurred and diffuse."

Naturalness: "The music seems to be as if there is no hearing aid, and the music sounds as 'I remember it.'"

Overall Fidelity: "The dynamics and range of the music are not constrained or narrow."

Measures of signal-to-distortion ratio and of sound quality were assessed in 23 professional musicians who had music-induced and/or presbycusic hearing losses. The musicians ranged in age from 42 to 81 years and included 17 males and 6 females.

RESULTS—AND A WEB SITE

Figures 2 and 3 show the "fill-in" in the square notch as the peak input limiting level dropped from 115 dB SPL, to 105 dB SPL, to 96 dB SPL, and finally to 92 dB SPL. For clarity, Figure 2 shows the data for 115 and 105 dB SPL, while those for 96 and 92 dB SPL appear in Figure 3.

Quantifying Figures 2 and 3, Table 1 shows the signal-to-distortion values. These values were measured in a 2-cc coupler, but data were also measured in the real ear for all 23 subjects. To hear the difference as the peak input limiting level decreased in steps from 115 dB SPL to 92 dB SPL, we generated wave files for both average conversational speech (65 dB SPL) and typically intense music (102 dB SPL). The wave files can be seen at www.musicandhearingaids.cjb.net.

Figure 4 shows the sum of the five perceptual scales (maximum value of 25) plotted against the measured signal-to-distortion ratios for the four peak input limiting levels.

Statistically, there is no significant difference between the sum totals of the five five-point, perceptual attribute scales for peak input limiting levels of 115 dB SPL and 105 dB SPL. Neither is there any statistical difference between the two lower peak input limiting levels of 96 dB SPL and 92 dB SPL. There is, however, a statistically significant difference between the two upper levels and the two lower levels.

In addition, to the above measures, all subjects reported that they preferred the 115 dB SPL and 105 dB SPL levels to the lower levels. Comments included: "Music was so much more natural at the higher peak input limiting levels." "My horn sounded like it should at the higher levels." "While conducting, I could hear the orchestra beautifully." Still, one subject noted, "...occasionally the drums and a couple of notes on the trumpet, sounded odd."

OTHER PARAMETERS

Clearly, the peak input limiting level is a major determining factor in the optimal reproduction of music. However, three other factors are also necessary. This study did not examine these other factors in detail. Rather, they are derived from my 20 years' experience in dealing with hundreds of hard-of-hearing musicians and non-musicians who enjoy listening to music. The following parameters will be the subject of future research.

Parameter #1

For listening to music, one channel is probably best. In music, unlike speech, the relationship or balance between the lower frequency fundamental energy and the higher frequency harmonic energy is crucial for the perception of optimal sound quality. This is especially true for violinists and violists, as well as for hard-of-hearing people who regularly listen to classical music. Therefore, it is necessary to use a single-channel hearing aid that maintains this balance.

On the other hand, for woodwinds and quieter music or for patients with precipitous audiometric configurations, a multichannel hearing instrument may be acceptable, since woodwinds rely perceptively on the lower frequency information in their music.

Parameter #2

The kneepoint on the input-compression circuit should be set approximately 5 to 8 dB higher for music than for speech. Speech has a "crest factor" of about 12 dB. The crest factor is the difference between the RMS of the signal and the peak.

Because the human vocal tract is so inherently damped (soft lips, nasal cavity, soft palate, etc.), the peaks of speech are closer to the average RMS. In contrast, non-vocal music is generated by electrical instruments or by instruments with hard-walled cavities. As a result, there is less inherent damping. The difference between the average RMS and the peak levels in a music spectrum is therefore greater. Typical crest factors for music are on the order of 18 to 20 dB. Subsequently, the compression circuit should be prevented from entering its non-linear phase prematurely.

Parameter #3

A good argument can be made for giving musicians a wide dynamic range compression (WDRC) circuit. Most musicians, especially those who suffer from the effects of excessive exposure to music and/or presbycusis, have a mild to moderate hearing loss with poorer acuity in the mid- to high-frequency ranges. This type of hearing loss reflects predominantly outer hair cell damage; subsequently, the WDRC circuit is typically optimal.

CONCLUSION

Given the characteristics of today's hearing aid technology and the types of input spectra that musicians and regular listeners are subject to, the peak input limiting level should be at least 105 dB SPL. Which precise setting is optimal probably depends on the type of music as well as the particular instrument(s) that the musician plays.

The vast majority of hearing aids on the market have a restricted peak input limiting level (typically 85-90 dB SPL). However, since modern microphones can safely transduce 115 dB SPL without appreciable distortion, there is no audiologic reason to limit the input range of today's hearing aids. Of course, the output should be limited in accordance with the individual's hearing loss.

The problems associated with too low a peak input limiting level are unrelated to the hearing aid's signal processing method. That is, this is as much a problem for analog instruments as for digital.

The optimal hearing aid for hearing-impaired musicians and other music lovers includes a high peak input limiting level (at least 105 dB SPL), WDRC with a higher threshold kneepoint than is prescribed for speech, and, for most musicians, a one-channel system. One popular hearing aid circuit that comprises elements of all four parameters is the K-AMP.

There are other technologies that can simulate a wider input range of sounds to the hearing aids without elevating the peak input limiting level, and these should be assessed clinically. However, merely altering the frequency response of the hearing aid will not benefit musicians or those who regularly listen to music. The input spectrum of music is inherently quite variable and significantly different from that of speech. Hearing aids need to be designed with this and the signal dynamics of music in mind from the very onset of development. HJ

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Marshall Chasin, MSc, Reg. CASLPO, Aud(C), is Director of Auditory Research, Musicians' Clinics of Canada. Correspondence to him at #340- 340 College Street, Toronto, ON, Canada, M5T 3A9; Marshall.Chasin@rogers.com.

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