

would be amplified by the hearing aid. However, most manufacturers of non-occluding behind-the-ear hearing aids use a broadband microphone to minimize the hearing aid noise level. A broadband hearing aid microphone would still allow the A/D converter to be overdriven, resulting in poor fidelity for music, even though the lower-frequency components would be lost through the non-occluding coupling to the ear. Higher-frequency harmonic distortion components would still be transduced through the hearing aid system and thus be perceived by the hard-of-hearing listener.

Using a less sensitive low-cut microphone would minimize the chances that the intense low-frequency sound components of the music would overdrive the A/D converter. Internal noise would increase with this type of microphone, so judicious use of expansion would be necessary. Commercially available -6 dB/octave, low-cut microphones are available in almost all mechanical sizes and electrical configurations as broadband microphones, so there is no inherent reason why any hearing aid manufacturer would be unable to perform this modification.

**(5) Taping down the sound:** Scotch tape has many uses, and one of them is for anyone who wants to enjoy listening to and/or playing music. Here's something I often recommend to my clinical patients and it works very well. Simply stated, placing one or two layers of Scotch tape over the hearing aid microphone(s) will reduce the microphone's sensitivity by 5-10 dB. This results in an additional 5-10 dB range

for the A/D converter. Some experimentation is required, but simply placing the tape over the microphone opening(s) prior to a concert or even in a noisy movie theater can significantly improve the listening experience. You should counsel your clients to remove the tape once they leave the musical venue.

**(6) Using an electric network:** It is not difficult to have a hearing aid manufacturer reduce the sensitivity of a microphone by 10-12 dB using an electric network. This has the same benefits as the Scotch tape method, and is typically programmed to be "on" (a -10-dB reduction in sensitivity) or "off" (normal function) with either a pushbutton or remote control. This may be implemented differently by different manufacturers, but a goal of a uniform reduction of 10-12 dB is reasonable. Depending on the instrument, the required part of the circuitry may not be accessible, so this approach may not be an option with every manufacturer. There are several ways it can be implemented, including by reducing the electrical charge on the back part of the microphone capacitive sensor.

## CONCLUSIONS

These six clinical and manufacturer-based modifications work well for listening to more intense music, and I routinely recommend some of them to my clients. I should add that at no time have I referred anyone for routine "software" adjustments. Listening to music is not a software issue; it is a front-end hardware issue. "Programs" that successfully alter the dynamic char-

acteristics of the input, such as Live Music Plus, are not simple software programs, but are actually hardware changes that can be implemented by software changes.

Simply altering the frequency response, compression, gain, and output characteristics will be of very little benefit in listening to music unless the analog-to-digital converter is presented with an input within its operating characteristic. Not all manufacturer-based modifications will be simple. That will depend on many factors, including whether or not a particular portion of a circuit is accessible in any given hearing aid.

## Acknowledgments

I would like to acknowledge the valuable input and discussions of Steve Armstrong and Jim Ryan on earlier drafts of this paper. I would also like to point out that any errors in this paper are mine alone.

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may react inappropriately when music is present, since there are many acoustic differences between speech and music.

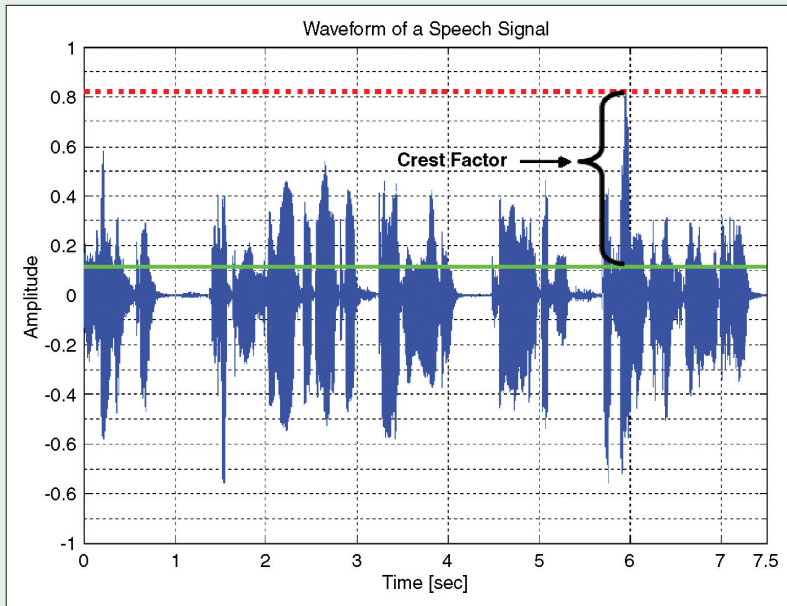
A hearing aid that has been optimized to handle music as an input should have both software and hardware differences from other instruments. Bernafon has developed Live Music Plus, a software program with a dedicated combination of features for live music processing, which is available in its Veras and Vérité 9 hearing instrument families. In this

# Programming hearing instruments to make live music more enjoyable

By Neil S. Hockley, Frauke Bahlmann, and Marshall Chasin

While concentrating our clinical efforts on the perception of speech in many different environments, hearing healthcare providers may sometimes overlook other signals, such as music, that may be very meaningful to the patient. Because hearing instruments are

designed to focus on speech, music lovers and musicians are often disappointed by the sound quality of music. Settings and electroacoustic characteristics of hearing instruments may be ideal for speech signals, but not for music.<sup>1</sup> As a result, hearing instruments



**Figure 1.** The RMS and peak level of a speech signal to show the crest factor.

paper we will first review some of the differences between music and speech signals. We will then explore the four elements that make up Live Music Plus, and finally we will report on the reactions of some professional musicians who have tried hearing aids with this program.

### MUSIC IS DIFFERENT

Chasin<sup>2,3</sup> and Chasin & Russo<sup>1</sup> have pointed out a number of differences between music and speech, including these three:

#### (1) Speech vs. music spectra

Speech has a relatively uniform spectrum (the range of frequencies produced), since the human vocal tract is the source. The sound source is similar, even though there are differences between the voices of men, women, and children. This speech spectrum has been extensively characterized by Byrne et al.,<sup>4</sup> and standardized as the Long Term Average Speech Spectrum (LTASS).<sup>5</sup> The speech spectrum is used as a foundation of fitting rationales to restore the audibility of speech via amplification. Music, on the other hand, has many, highly variable sources, and the resulting spectrum can resemble noise in some cases and speech in others.<sup>1</sup> Therefore, there is no truly representative long-term music spectrum.

#### (2) Different intensities

Soft speech is generally considered to be about 50 dB SPL, conversational speech around 65 dB SPL, loud speech about 80 dB SPL, and shouted speech around 83 dB SPL.<sup>2</sup> Music, on the other hand, is quite different and can easily reach 105 dB(A)\* and have peaks of up to 120 dB(A). Killion has measured peaks of a symphony orchestra in a concert hall at 114-116 dB (C).<sup>6</sup>

Speech has a well-defined relationship between loudness (the psychological impression of the intensity of a sound) and intensity (the physical quantity relating to the magnitude or amount of sound). For music this relationship may be variable and greatly

\* The dB A scale is used to approximate what we hear as opposed to the physical sound pressure level (SPL). The dB C scale is used to measure the peaks of a signal. Both dB A and dB C filters are found on most sound level meters.

depends upon the musical instrument being played.<sup>1</sup> For example, for bass string instruments such as the cello and the acoustic bass, less gain should be applied for the lower frequencies than for speech.<sup>2</sup>

#### Crest factor

The *crest factor* is the difference between the peak level and the average (RMS) level. More specifically, the crest factor is the instantaneous difference between the peak of a signal and the overall level. This can be seen in Figure 1 for a waveform of a speech signal. The dotted red line represents the peak of the signal, while the green line represents the RMS level; the resulting crest factor can be seen in black.

Speech has a fairly consistent crest factor of 12 dB, while music has a crest factor of up to 18-20 dB for many instruments.<sup>2</sup> This acoustic characteristic is very important for the dynamic impact of music.

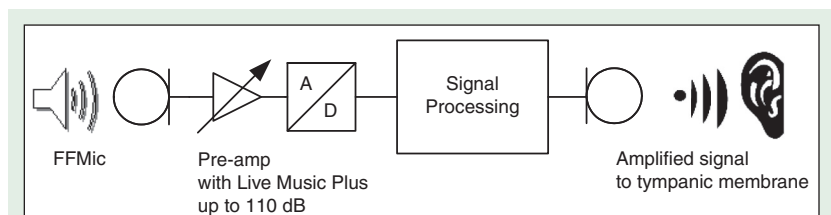
From this very brief discussion of the differences between speech and music, it is quite easy to see why these signals must be processed differently within the hearing instrument. Now we will explore the four systems that Bernafon has implemented to improve live musical sound quality.

#### LIVE MUSIC PLUS

The four systems that make up Live Music Plus are: (1) Live Music Processing, (2) ChannelFree™ Compression, (3) wideband frequency response, and (4) microphone settings. Now let's look at each of these systems individually and how they work together.

#### Live Music Processing

As we discussed earlier, music has different intensities and crest factors from



**Figure 2.** A basic block diagram illustrating the signal path from the microphone, to input compressor, to A/D converter, to amplifier, and finally to the receiver.



**Figure 3.** A comparison between the input and output functions with Live Music Processing on and off.

speech. These dynamic characteristics create a challenge to digital hearing aids. Typically, before converting the signal from the analog to the digital domain, a digital hearing aid compresses or clips the peaks of the signal when they reach 95 dB. While this is more than adequate for even loud speech, for the peaks of live music this is too low and the music will sound compressed, unnatural, even distorted. This is especially a drawback for musicians who need to hear their colleagues to play correctly. Live Music Processing increases the level to 110 dB to preserve the peaks in music before they reach ChannelFree™ processing.

Figure 2 shows a typical signal path of a hearing instrument. The acoustic signal is received by the microphone, amplified slightly, converted to the digital domain (A/D converter), and then processed. In the pre-amplifier (an AGCi), the signals are typically limited to 95 dB, while with Live Music Processing, signals up to 110 dB are allowed through to be processed by the A/D converter.

An easy way to look at the difference between the pre-amplification for a standard hearing instrument and Live Music Processing is to look at an input/output function. In Figure 3 we can see an input/output function for a 1000-Hz

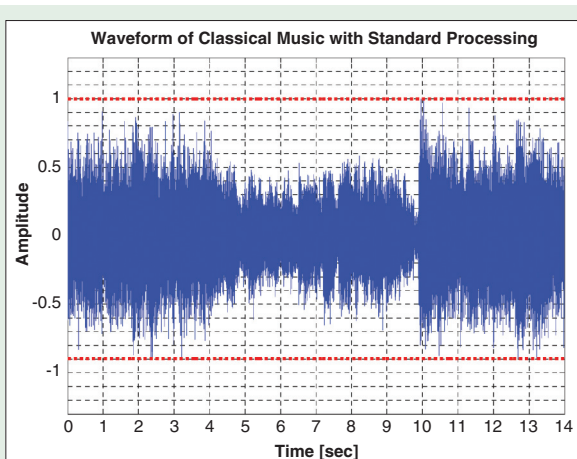
sinusoidal signal, with the red line representing a standard hearing instrument program designed for speech with a 95-dB SPL limit. After the 95-dB SPL input, the curve begins to level off, indicating that the instrument is compressing this signal. The blue line represents the same instrument but with Live Music Processing. In this case, the hearing aid is not compressing the signals until they exceed 110 dB SPL.

Pure sine wave signals are not so common in music (except electronic music), so it is important to look at the effects that have been seen so far with music. Figures 4 and 5 show amplified music displayed as waveforms with amplitude on the y-axis and time on the x-axis. In Figure 4, we see a signal processed without Live Music Plus. The peaks of the waveform are cut off, as indicated by the top red dotted line. This line signifies the maximum level that the hearing instrument will permit to be converted to the digital domain. The same signal can be seen in Figure 5 with Live Music Processing. However, here the peaks of the musical signal are preserved and the dynamic range is higher, demonstrating that the natural dynamic characteristics will be converted into the digital domain.

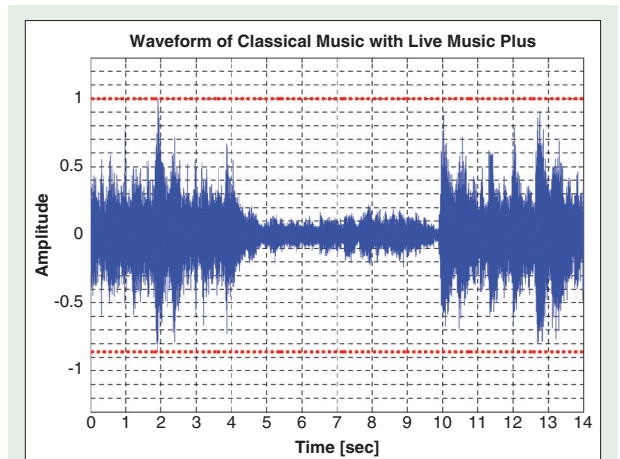
### Compression system

The ChannelFree compression system has a fast processing time and treats signals as a whole to maintain the balance between low- and high-frequency harmonic energy.<sup>7-9</sup> The high-frequency harmonics, for example, are especially important for judging the timbre (the difference between musical instruments, e.g., a trumpet and a violin, playing the same note at the same intensity). This balance is crucial for musical sound quality.

ChannelFree compression is designed to maintain the level differences between the sounds of music, thus resulting in a natural perception of the musical signal. The peaks of musical signals may be sharper than speech, as described earlier in our discussion of the crest factor, and may send a standard hearing aid into too much compression too early. However, ChannelFree compression can quickly follow the level of the signal



**Figure 4.** Amplified music waveform without Live Music Processing.



**Figure 5.** Amplified music waveform with Live Music Processing.

to preserve the relationships between different levels of the musical signal, which results in a signal that is amplified to a comfortable level for the patient.

Bernaon's ChannelFree compression system has been judged to have high sound quality. A 2003 study by Dillon et al. with hearing-impaired listeners found that Symbio, a first-generation ChannelFree processing hearing instrument, received higher ratings than any of the other digital hearing aids tested, for the sound quality of piano music.<sup>10</sup>

### Wideband frequency response

It is well known that a wide frequency response contributes to the perceived naturalness of music.<sup>6,11</sup> Hearing instruments with Live Music Plus have a frequency response up to 10,000 Hz, sufficient to convey most musical sounds accurately. For example, the highest C note on a piano is 4186 Hz, while the highest C note on a violin is 2093 Hz.<sup>12</sup>

### Microphone settings

For listening to music, all automatic features such as noise reduction and adaptive directionality need to be turned off. This is to prevent these systems from interpreting the music as noise or feedback, which may affect the sound quality.<sup>1</sup> When one is sitting in a concert hall, the people seated around you often make extraneous noise. Perhaps they are explaining what is happening on stage to their neighbor or opening a candy wrapper.<sup>13</sup>

Applause can also be very disruptive for a hearing aid wearer. But Live Music Processing allows the hearing aid user to select a fixed directional-microphone setting (hypercardioid) so as to place the focus more on the stage and less on the people seated around you.

### Four elements work together

To summarize, Live Music Plus combines four elements to present live musical signals accurately and enhance the experience of music for the hearing aid wearer:

- ❖ Live Music Processing to preserve the dynamic characteristics of music,
- ❖ ChannelFree processing to amplify music accurately so that it is within the wearer's dynamic range,
- ❖ A wideband frequency response to help make the music sound natural, and
- ❖ A fixed directional setting to focus on the performing musicians.

### EXPERIENCE WITH LIVE MUSIC PLUS

A trial was conducted in which nine professional musicians (eight males and one female) were asked to rate how the music they played sounded to them when they wore hearing aids with Live Music Plus. Four of the musicians were woodwind players (clarinet, sax, and flute), of whom three played jazz and one played classical music. Three of the musicians were classical violinists who also played the viola. The final two musicians were both rock (electric) guitarists. All were current users of occluding in-the-canal instruments with an analog K-AMP circuit.<sup>14,15</sup>

Previously, these musicians had not worn digital hearing instruments because they found the sound quality unnatural. Many musicians use analog hearing aids, such as the K-AMP, because they can handle higher level inputs and do not have an A/D converter.

The nine musicians were fitted with Bernafon Veras 9 micro-BTE hearing instruments programmed with Oasis fitting software. Eight wore non-occluding earmolds while one used fully occluding earmolds because of the degree of hearing loss.

The attribute scales used with the subjects were based on the work of Gabrielsson et al.<sup>16,17</sup> and Cox and Alexander.<sup>18</sup> The scales consisted of qualitative descriptions of sound quality, as shown in Table 1. Based on what he or she experienced with the hearing aids, each client gave a numerical rating to each of the five attributes in Table 1.

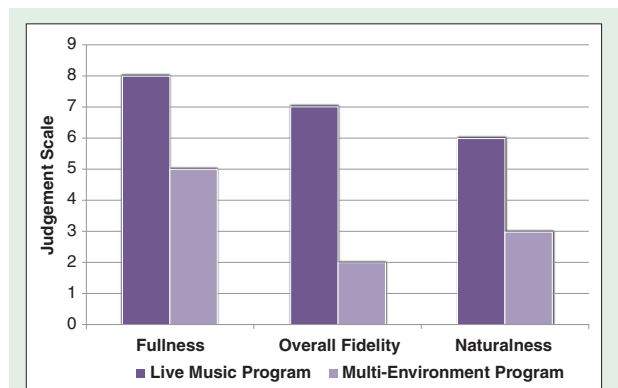
The results for loudness and crispness showed no significant difference between the Live Music Plus program and a multi-environment program. However, there were clear differences in the results for the other three attributes: fullness, overall fidelity, and naturalness. These are shown in Figure 6.

A program with Live Music Plus was judged significantly fuller ( $p < 0.05$ ) than the multi-environment program. Overall fidelity with Live Music Plus was judged as significantly better ( $p < 0.05$ ) than with the multi-environment program. There was no significant difference for naturalness between the two programs due to a large variance in the response data; however, a trend was observed (Figure 6).

The fidelity to the input signal can also be seen by measuring distortion with the musicians' instruments in a 2-cc coupler

**Table 1.** Examples of attribute and perceptual dimension of sound quality judgments.

Attribute	Perceptual dimension
Loudness	Loud vs. Faint
Fullness	Full vs. Thin
Crispness	Crisp vs. Blurred
Naturalness	True to the source vs. Artificial
Overall fidelity	Wide dynamics vs. Limited and compressed



**Figure 6.** Judgment of test clients.



and test box. In this test, a 100-dB SPL stimulus was the input. The hearing aid was set to 5 dB of gain and the OSPL-90 was set to a very high level such that the [input + gain << output]. Any measured distortion would have been from an A/D converter that was receiving input that was outside of its operating characteristic. For more information on this test, see Chasin.<sup>19</sup>

With Live Music Plus, less than 3% of total harmonic distortion (THD) was seen, in contrast with the multi-environment program where 18%-24% (THD) was seen for very loud input signals.

## CONCLUSIONS

Music differs dramatically from speech and is therefore a potential challenge for hearing instruments. Bernafon's Live Music Plus program uses a combination of elements to present live musical signals accurately to the wearer. A trial of this technology involving professional musicians suggests that music lovers, whether performers or listeners, may appreciate this program.

## Acknowledgments

The authors would like to sincerely thank Jesko Lamm (Bernafon Switzerland) for his assistance with the graphics.

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# Enhancing music with virtual sound sources

By Pauli Minnaar

For many people, listening to music is an important part of life. Most often the music is recorded and played on a CD player, the radio, the television, an mp3 player, or a computer. Listening to music from such devices was long out of reach for hearing aid users. But recently, the development of devices, such as the Oticon Streamer, that can send music wirelessly to hearing aids enables people to enjoy listening to music directly in their hearing aids with a good signal-to-noise ratio.

However, listening to music sent directly to hearing aids is not optimal. Specifically, the sound image appears to be inside the listener's head. This is referred to as "in-the-head locatedness."<sup>1</sup> When the signal is the same at both ears (monophonic), the listener perceives it as being in the middle of his or her head. When the signal is stereophonic, the sound is perceived as being on a line between the ears. By changing the level of the signal in either ear, the sound can be moved between the ears. This is referred to as "lateralization of the sound image."

Thus, with a stereophonic signal the sound image can be lateralized, but it is still perceived as being inside the head.

Users generally experience this as unpleasant and unnatural since it is not what occurs in real-life listening, where sound sources are placed at a distance in the space around the listener. Therefore, it is desirable to enhance this sound image to make it more natural and pleasant to listen to.

The problem of in-the-head locatedness also occurs when people listen to stereo music through ordinary headphones. This is because stereo music is designed to be played through two loudspeakers placed in front of a listener in a room. Specifically, the loudspeakers have to be placed at  $\pm 30^\circ$  if the listener is to perceive the correct stereo image. With this setup, the sound image is perceived as between the loudspeakers, which creates the correct spatial sound stage. Since the sound image is out in the room, not in the head, normal binaural hearing can be used to localize the sound.

Since recorded music is intended to be listened to through loudspeakers in a room, it is desirable to simulate this when listening through hearing aids (or headphones). This can be done by filtering the hearing aid signals similarly to how they