Better Hearing Through DSP

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There have been more technological advancements in the hearing aid industry during last 10 years of the 20^{th} century than during the first 90. The commercial success of multiband compression, multimicrophone directional processing, and commercially viable digital signal processing have made the industry unrecognizable from it's state 15 years ago. Signal processing—be it electrical, acoustical, or digital—has become the latest frontier for growth in the field, and as DSP chips designed for hearing aids become more powerful, more sophisticated sound processing algorithms will be produced, since all domains of sound can be manipulated or altered. Amplitude, spectral and temporal characteristics of sound can be re-configured as necessary, offering a whole new range of sound processing solutions. Incoming signals can be “analyzed” so that intelligent sound processing decisions can be made. This “multi-dimensional tool box” that only DSP technology can expose, is the true value of DSP as the premier design platform. and the person fitting these algorithms on patients must maintain current in their understanding of how different signal processing algorithms work in order to properly fit them to the hearing loss of their patients.

With many of the currently available DSP products, however, the signal processing schemes that these DSP devices are executing remain amplitude domain manipulation schemes. With these designs, all of the tools in the DSP toolbox are not being fully utilized. In some cases, these products are merely DSP duplicates of analog signal processors. It stands to reason then, that comparisons of DSP and analog signal processing schemes that are similar in nature, should yield similar performance and benefit results. However, as Robert Sweetow has said, “It is the potential of DSP, rather than the current state of the art, that is so exiting.” With that in mind, several areas that DSP can exploit successfully will now be reviewed.

FFT Processing and Multiband Compression

The basis for many digital signal processing technologies is the Fast Fourier Transform (FFT), which is found in, for example, DVD and digital cell phone technology. The FFT provides a convenient way of calculating the spectrum of a signal and extracting its frequency information and is a capability that does not exists in analog technology. The usefulness of the FFT to digital hearing aid processing is obvious given the fact that all hearing aid processing is dependent on the frequency of the signal, such as increasing the compression ratio at high frequencies. The healthy cochlea can be viewed as a biological Fourier Transform since it separates sound into frequency regions along the basilar membrane, with high frequency sounds vibrating the basal end of the basilar membrane and low frequencies vibrating the apical end. While the FFT is not an end in itself, it allows complicated signal processing to be performed that do take advantage of the rich frequency information provided by the FFT.

Two examples of the way in which hearing-loss compensation can be improved using FFT-based processing are provided in the Digital 5000 and Danalogic lines. Products in both lines provide the ability to adjust the frequency response in 64 independent bands. This ability is used at the manufacturing level to fine-tune the response of the device such that smooth insertion gains can be achieved, and is a level of refinement not seen in the industry until the introduction of the FFT to hearing aid processing. In addition to the 64-band processing, 14-channel compression is implemented in a manner similar to the compression processing in a healthy cochlea. As with auditory filters, the bands are heavily overlapped to eliminate distortions that can occur in the channel transition regions. Additionally, the high number of channels allows for precise fitting of gain and compression to the specific hearing loss of each individual patient. Digital technology allows this 14-channel, 64-band processing to be achieved on a small low-power hearing aid chip.

Directionality

Improving speech understanding in noisy environments is another area where digital technology can improve current hearing aid solutions. Digital processing has the capabilities to make may features of directionality adaptive and can provide for easy programmability when directional processing is implemented digitally. The directional pattern—the characteristic that describes how much gain is applied for each direction—is controlled by the delay applied to the rear microphone in a dual-microphone directional hearing aid. By implementing this delay digitally, as is done in the Danalogic and Digital 5000 lines, the directional pattern can be precisely controlled at all frequencies, maximizing the improvement in the signal-to-noise ratio and thus maximizing speech understanding in noisy environments. Since different noise environments may require different directional patterns, there may be a need to provide the option for programming different directional patterns into the hearing aid. Because of the digital implementation, changing the directional pattern can be easily done with fitting software, such as with the Digital 5000's ReSource software.

Noise Reduction

Signal perception in the presence of background noise can also be enhanced using a modulation-based noise reduction algorithm. This technique, implemented with DSP technology, is based on the principle that the amplitude modulation pattern of speech over time is distinctively different than the amplitude modulation pattern of noise. In Figure 1 below, the typical amplitude modulation of a speech signal over time is depicted. Note the rather large fluctuations of amplitude that occur as speech progresses through the vowels, consonants, voiced and unvoiced segments that make up speech.

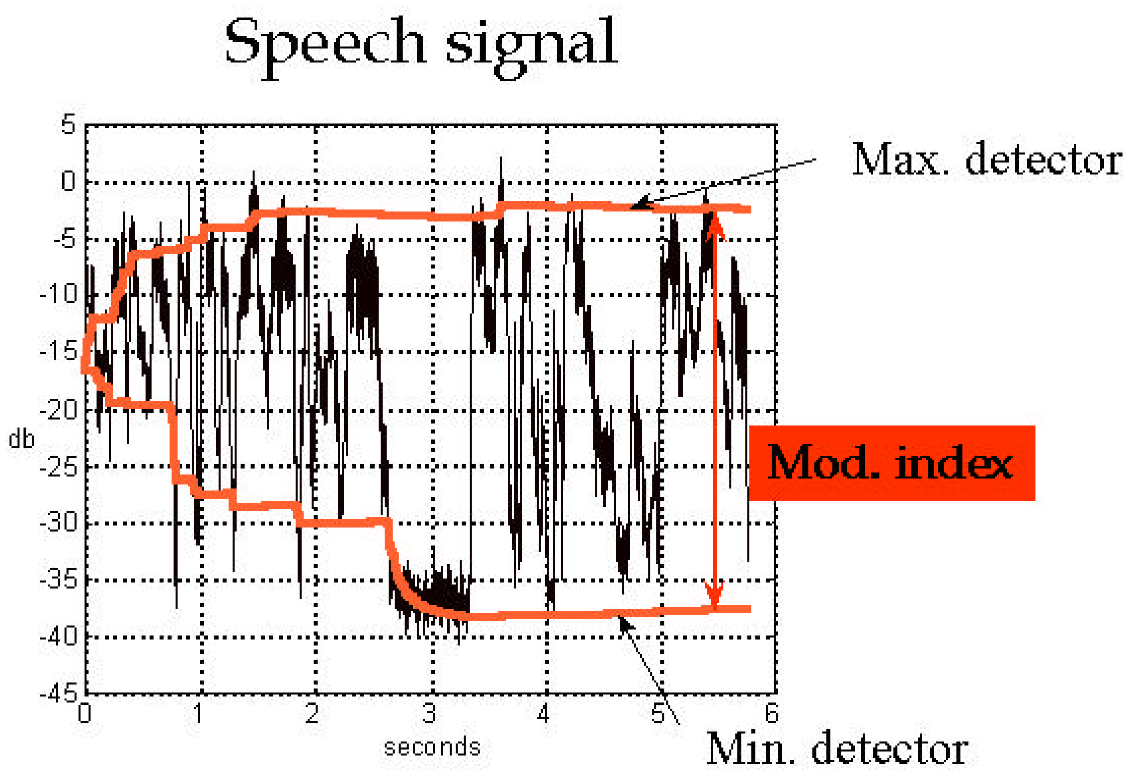

Figure 1

In contrast, Figure 2 shows the typical modulation pattern over time that noise produces. Even though the noise signal may contain many of the frequencies that make up speech, its temporal pattern over time is distinctively different. The large amplitude changes so characteristic of speech energy are not evident when noise is the input signal, even if that noise is multi-talker babble.

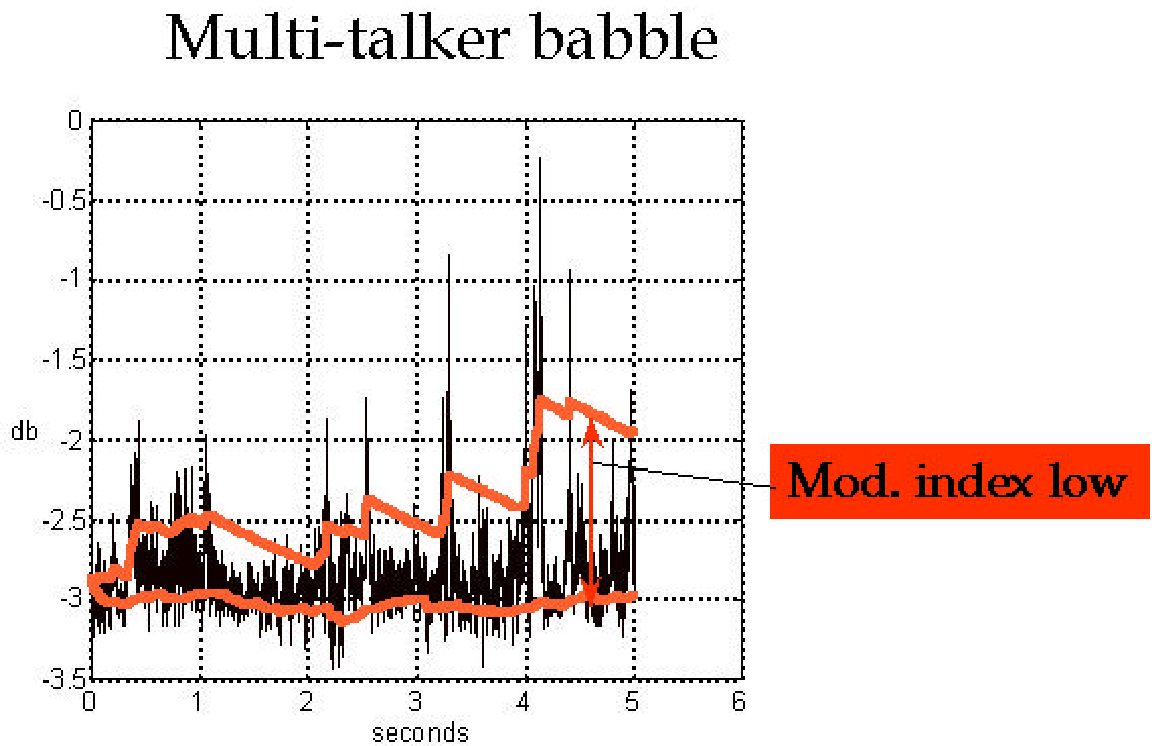

Figure 2

In digital systems that incorporate multi-band amplification, it is possible through modulation analysis to determine which of those bands likely is processing speech energy, and which of those bands is processing noise energy. Once this difference has been determined, an algorithm can be designed to reduce the gain applied to those bands that appear to be processing predominantly noise energy.

If speech energy happens to be located in a band that has been diagnosed as containing dominant noise modulation patterns, then the gain applied to that speech energy will be reduced as the gain in that band is reduced. The risk of this happening can be reduced as the number of bands involved in the noise modulation detection scheme is increased. An example of this is depicted in Figure 3. In example A, a three band amplifier is depicted, amplifying all input frequencies equally. The resulting time-based output pattern is to the right. The gray components of the output wave form are speech energy and the black components of the output waveform are noise. If, for example, noise modulation patterns dominate the content of the energy driving the gain in the third, high frequency band, then a gain reduction based on this modulation information would be triggered. In example B, this gain reduction is depicted, and the resulting effect on the output signal is to not only reduce the noise energy in that signal, but also some of the speech energy as well. Since the speech energy that was reduced was critical high frequency energy, then there can potentially be a very significant degradation of speech intelligibility as a result of this noise reduction technique.

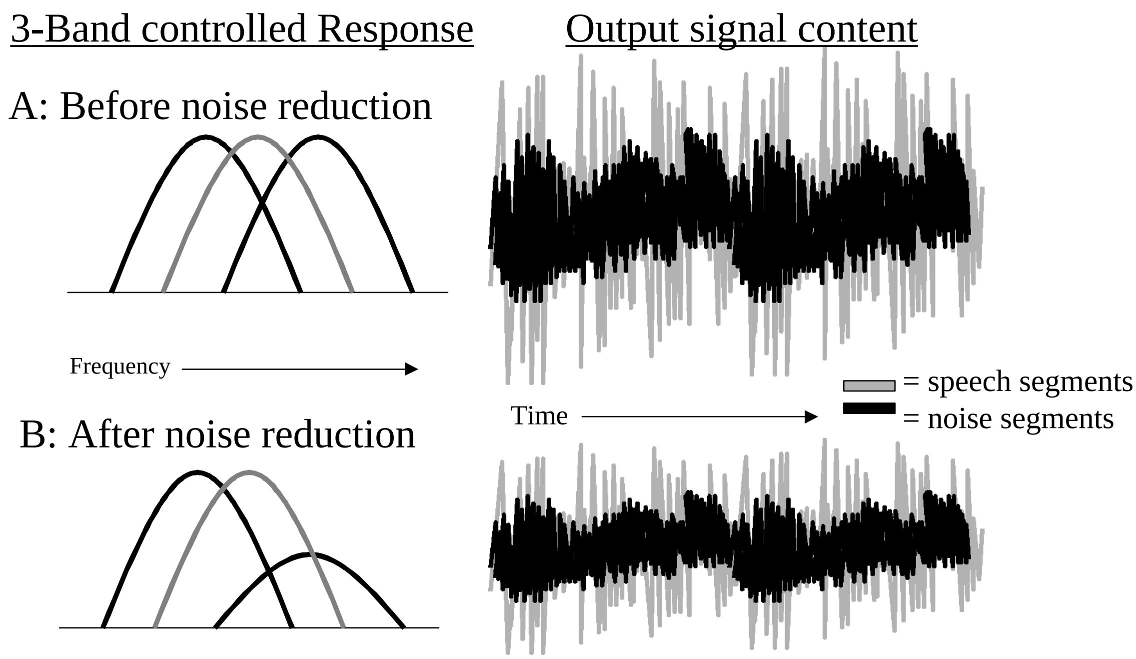

Figure 3

By implementing this algorithm in 14 bands, as is done in the Digital 5000 and Danalogic lines, spectral mismatches between speech and background noise can be better exploited by the algorithm than one with fewer bands. By using this high-resolution noise reduction system, the noise can be more effectively isolated from frequency regions with clean speech, and the long-term and broadband speech-to-noise ratio can be better enhanced.

Digital Feedback Cancellation

In many hearing aid fitting cases, the amount of high frequency gain that can be effectively delivered is restricted by that systems ability to contain acoustic feedback. Even with several of the digital systems available today, digital feedback management techniques often reduce critical high frequency gain in an effort to contain feedback oscillation.

In contrast, a unique digital feedback suppression approach, first introduced in 1993 in the Danavox DFS Genius, offers the ability to contain feedback oscillation without reducing (and even accessing more) high frequency gain.

The principle behind this feedback suppression approach is phase canceling. By introducing an oscillation signal into the feedback pathway that is the same as the actual feedback oscillation present in the fitting, but 180 degrees out of phase with it, the actual feedback oscillation signal will be rendered inaudible.

Figure 4 depicts a processing diagram of the digital feedback suppression technique used in the Danalogic and Digital 5000 digital product. The digital algorithm used for digital feedback suppression in this product incorporates both a static and an active feedback suppression filter. The static filter is used to suppress feedback oscillation that may be present in the environment in which the hearing instrument is being worn. The active filter is used to suppress feedback oscillation that may be induced by changes in the instruments acoustic environment, such as when a hand or a telephone is placed near the instrument.

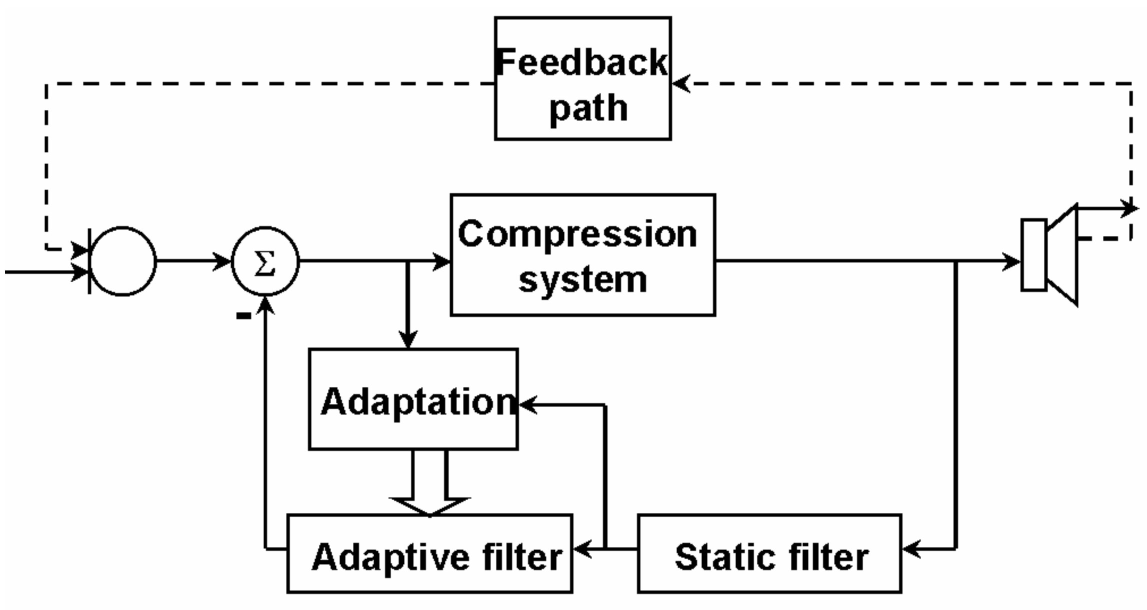

Figure 4

In order for the appropriate 180-degree out-of-phase duplicate of feedback oscillation to be created by these filters, a calibration procedure must be incorporated into the fitting procedure. This calibration procedure involves the presentation of a digital test signal that is generated by the hearing instrument while it is worn in the patient's ear. This test signal, (which sounds like a loud buzzing sound to the patient) is used to analyze the acoustic conditions the hearing instrument is being worn in, and to establish the necessary reference point for digitally creating the phase canceling duplicate. The calibration procedure takes approximately 15 seconds to complete. Once the necessary phase canceling duplicate has been introduced into the aided acoustic environment, feedback oscillation can be controlled without the need for gain reduction.

Clinical data has shown that activation of the DFS algorithm can allow high-frequency gain to be increased by 10-15 dB beyond the point where feedback normally occurs when DFS is inactive. So, besides providing patients with more safety from feedback difficulties, DFS also allows more gain to be programmed for patients who need significant high-frequency gain, which can translate to better speech understanding. Clinical studies in both the United States and Europe have confirmed this finding, showing increased intelligibility with the proper use of the DFS algorithm.