Vol. 1 No. 1 / April 4, 1998

SONIC innovations

Digital Signal Processing (DSP) Derived from a Nonlinear Auditory Model

Victor H. Bray, Jr.
Douglas M. Chabries
Keith L. Davis
Jeannette S. Johnson

Abstract

Digital signal processing (DSP) hearing aids are producing excitement in audiology, but DSP does not necessarily deliver increased user benefits. To be successful, DSP performance must match the specific auditory needs of the hearing-impaired individual. Stockham and Chabries (1996) have patented an amplification design based upon a non linear cochlear model. A DSP-based hearing aid employing the patented solution, Multiplicative DSP™, has been developed by SONIC innovations.

Presented at the American Academy of Audiology in Los Angeles, California, on April 4, 1998.

The Auditory Model

A model describing the human auditory response to sound is pictured in Figure 1. The model includes (A) a set of bandpass filters, similar to basilar membrane

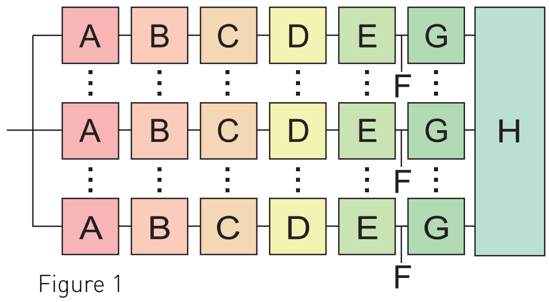


Figure 1

function; (B) an envelope detector, which estimates the signal intensity from each filter, similar to cell detection; (C) a nonlinear intensity gain, which logarithmically compresses the signal, similar to outer hair cell gain and inner hair cell responses; (D) a high pass filter, which provides loudness adaptation; (E) a hyperbolic tangent, which mimics the inner hair cell firing and associated neural network; (F) a multiplicative intrinsic noise source; (G) an exponentiator; and (H) detector to mimic loudness detection in the brain. This model of the normal auditory system may be combined with the equivalent inverse model of a damaged auditory system to enable near normal hearing perception for a hearing-impaired ear[1].

Multiplicative AGC

An acoustic signal s(t) may be represented as s(t) = e(t) v(t) where e(t) is a slowly-varying positive-valued envelope and v(t) is a rapid varying vibration. The neural firing rate in the human ear is determined by the intensity of the signal envelope, log e(t), and encoding of frequency information, log /v(t)/. Thus the model independently maps the intensity cues in the envelope and processes the frequency cues in the vibration, which is believed to be the way the auditory cortex perceives sound[1].

An inverse model for the cochlear-damaged ear leads to a new method of processing auditory signals, termed "Multiplicative AGC." This system provides little or no gain for signals at the upper sound comfort level and gain nearly equivalent to the hearing loss for signal intensities associated with normal hearing[2].

When realized in a multichannel configuration, Multiplicative AGC allows control of the audio signal with narrowband precision. The block diagram below (Figure 2) displays the 9-channel implementations by **SONIC innovations**, designated Multiplicative DSP™, enabling programming control with half-octave resolution. Testing of the signal processing algorithm against high-performance hearing aids has produced favorable results[3].

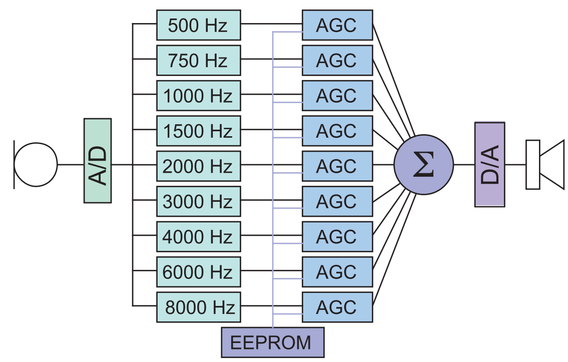


Figure 2

Multiplicative DSP

The nine AGC channels can be configured independently to match the individual needs of the hearing-impaired person. Figure 3, containing both an output curve and a gain curve, illustrates typical setting for the 2000 Hz channel.

Each compression channel has two kneepoints (identified by the arrows above). The kneepoints can be adjusted horizontally in 1-dB increments for compression threshold and vertically for desired amount of gain at

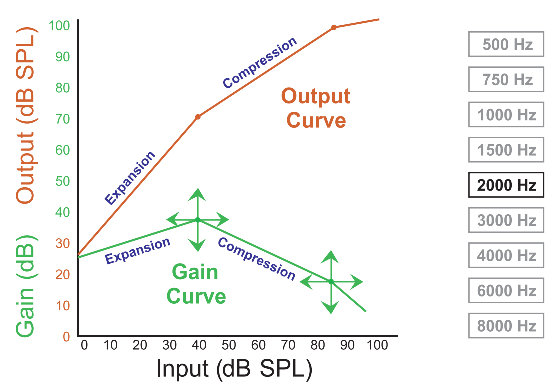


Figure 3

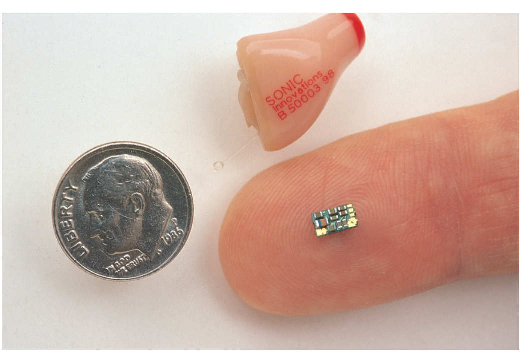
the kneepoints. When the gain curve is configured with a rising slope, the gain increases as the input level increases. This is termed "expansion" and is useful in reducing low-level background noises. When the gain curve has a falling slope, the gain decreases as the input level increases. This is termed "compression" and is beneficial for individuals with narrowed dynamic ranges, or recruitment. A horizontal gain curve would represent linear mode.

A demonstration of the extraordinary power and programming flexibility of this system can be seen in the companion paper "Optimized Target Matching: Demonstration of an Adaptive Nonlinear DSP System" by Bray, Harris, and Johnson (1998)[4].

Summary

The advent of low-power, miniature, digital signal processing (DSP) circuitry is creating exciting change in the hearing industry. However, the use of DSP, per se, does not necessarily mean improved user benefits when treating hearing loss with amplification. To be successful, DSP amplification must be congruent with the specific auditory processing needs of the hearing-impaired individual.

Stockham and Chabries (1996)[2] recently patented the design for an amplification system based upon a nonlinear model of the cochlear performance. One major benefit of the system is superior frequency-specific, level-dependent multichannel compression matched to cochlear deficits. Using these ideas, **SONIC innovations** has designed, developed, and manufactured an advanced integrated circuit (pictured below) and employed it to drive a unique and powerful DSP-based hearing aid.



References

1. Chabries, D., Anderson, D., Stockham, T., & Christiansen, R. (1995). Application of a human auditory model to loudness perception and hearing compensation. IEEE ASSP, 3527 – 3530.
2. Stockham, T., & Chabries, D (1996). US Patent 5,500,902.
3. Anderson, D., Harris, R., & Chabries, D. (1995).Evaluation of a hearing compensation algorithm. IEEE ASSP, 3531 – 3533.
4. Bray, V., Harris, R., & Johnson, J. (1998). Optimized target matching: Demonstration of an adaptive nonlinear DSP system. AAA – Los Angeles Poster Session.

Presented at the American Academy of Audiology in Los Angeles, California, on April 4, 1998.

SONIC
innovations