ADVANCED SIGNAL PROCESSING TECHNOLOGIES FOR INTELLIGENT HEARING AIDS

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Introduction

Digital signal processing technology has experienced a rapid evolution over the past three decades. DSP is central to a wide range of applications: speech coding and decoding for telecommunications, array signal processing for radar and sonar, spectral analysis, multimedia processing, as well as in devices used in daily life and home entertainment.

Approximately 10 years ago DSP began to be applied in digital hearing aids due to the development of practical low voltage, low power silicon technology [1][2][3][4].

DSP Platform for DSP in Hearing Aids

DSP based amplifiers for hearing aids must be small, usually less than 20 mm^2 footprint and less than 3 mm thick. Operating voltage has to be between 1.0 and 1.3V. Current consumption has to be less than 1.0 mA, so that a standard battery can drive a hearing aid for 100 hours or more.

The first practical digital hearing aids were based on low power Application Specific Integrated Circuit (ASIC) DSP cores. Their designs were optimized for efficient execution of FFTs, iFFTs, FIR filtering, and applications of gain. Long lead times are required to develop the ASIC hardware. It is difficult to alter its functionality because ASIC systems have a hard-wired, or fixed, functionality. But they do allow parametric changes via a programming interface to optimize gain, frequency response and compression characteristics of the device to best suit an individual's hearing loss.

Software controlled, or open, DSP systems with low voltage operation and low power consumption have also been developed. These can run various digital signal processing strategies, implemented in the form of algorithms. The time required for development and algorithm upgrades is significantly shorter than with ASIC hardware. [5].

The current DSPs in hearing aids integrate algorithm controlled cores with dedicated ASIC blocks. These blocks handle the repetitive computations of specific signal processing tasks in the most efficient way thereby meeting the computational requirements within the power consumption constraints imposed by the hearing aid environment. The advantages of flexibility for design and algorithm upgrades, is maintained.

DSP Algorithms

Adaptive filtering, noise reduction, beam-forming, signal enhancement and hearing loss compensation can be implemented in the time domain, the frequency domain or both. Therefore, time-to-frequency and frequency-to-time transforms such as fast Fourier Transforms or filter banks are usually required to convert the signal between the time-domain and the frequency-domain.

A. Noise Reduction

The signal-to-noise ratio (SNR) is an important factor in hearing and speech perception. Whereas normal hearing persons have good sentence understanding with an SNR as low as –5 dB, hearing impaired persons may need +5 dB SNR to do as well.

Signal amplitude modulation has been widely used to differentiate between a desired or useful signal (speech or music) and noise in a hearing aid. Amplitude modulation of a speech or music signal in a frequency band is usually higher than amplitude modulation of stationary signals like pure tones or fan noise, office noise, traffic noise and multi-speaker babble.

Amplitude modulation does not on its own produce consistently reliable signal detection. Signal detection and noise reduction have been improved by using additional information such as temporal information and timing information about the signal and noise in combination with amplitude modulation [6]. Hearing aid NR algorithms now routinely reduce noise by up to 10 or 20 dB, depending on the type of noise. NR algorithms are known to improve signal quality and listener fatigue but efforts to improve speech understanding need to continue.

B. Hearing Loss Compensation

The most important task of a hearing aid is to provide audibility, namely to amplify useful input signals to a level that the user can hear. Multi-channel hearing instruments can produce gain requirements within 5 dB of target.

C. Beam-forming

Monaural beam-forming and binaural beam-former processing are the major forms of applications which exploit spatial information about an acoustic signal and noise. Fixed delay-and-sum methods produce a fixed directional pattern. When the delay and sum parameters are varied an adaptive beam-former that allows the directional pattern to steer itself to the location of the noise source in order to maximize the attenuation of the noise is produced.

D. Feedback Cancellation

Adaptive feedback detectors and active feedback cancellers are appearing in many new digital hearing aids. The terms "adaptive" and "active" usually imply that the feedback canceller responds to changes in the acoustic feedback path and may be on or off as required to ensure stable operation of the instrument. These systems adapt quickly to changes in the feedback path caused by distortion of the ear canal due to, for example, the motion of the jaw when eating or speaking. With active feedback cancellation, the maximum gain of the system will usually not be reduced as it is in a fixed or static feedback reduction system.

Intelligent Hearing Aids

An intelligent hearing aid should include at least three fundamental capabilities: auditory scene adaptation, adaptive signal enhancement and adaptive hearing compensation.

Auditory scene adaptation gives the hearing aid the capability to detect, recognize and adapt to the acoustic environment [7]. Adaptive signal enhancement gives hearing aids the ability to optimize signal quality and to reduce or eliminate the interference of noise as it exists in the real world. It includes applying adaptive filtering, artificial intelligence, array signal processing, neural networks and various signal processing technologies to achieve signal enhancement, noise reduction or even separation of the signal from noise. Intelligent signal detection, adaptive noise reduction, beam-forming or adaptive beam-forming, feedback cancellation, and overlapping signal separation are key technologies used to achieve signal enhancement. Auditory scene adaptation and adaptive signal enhancement can provide benefits for any listener. The same signal can be perceived differently depending on individual's hearing loss. Adaptive hearing compensation can give more benefits for people with hearing loss. When adaptive signal compensation is integrated with auditory scene adaptation and adaptive scene enhancement hearing perception can be optimized. Reliable auditory scene adaptation and adaptive signal enhancement will allow hearing compensation to be adapted to different sound sources and will allow the enhanced signal to be optimized for the characteristics of individual hearing loss.

Summary

More powerful and more efficient DSP platforms of small size and with low operating voltage and lower power consumption will be available in future for the development of intelligent digital hearing aids. Digital signal processing technologies and the associated algorithms will continue to improve in quality and performance. Intelligent digital hearing aids will make it possible to maximize hearing aid benefit by automatic adaptation to changes in the acoustic environment.

References

1. Popelka GR: Computer and hearing aids: A Prediction of the future. *The Hearing Journal* 1998; 51(11);52, 57-62.
2. Fabry D: Do we really need digital hearing aids? *The Hearing Journal* 1998; 51(11); 30,32-33.
3. Murray DJ and Hanson JV: Application of Digital Signal Processing to Hearing Aids: A Critical Survey, *Journal of the American Academy of Audiology* 1992; 3(2)145-152.
4. Schum DJ: Artifical Intelligence: The New Advanced Technology in Hearing Aids, July, 2004, Audiology Online http://www.audiologyonline.com/articles/arc_disp.asp?id=733).
5. Berg C, Canneyt M V: Hearing Instruments Become Core-centric, *EETIMEs.com: The Technology Site for Engineers and Technical Management,* posted: 6/5/98.
6. Luo H, Arndt H: Apparatus and Method for Adaptive Signal Characterization and Noise Reduction in Hearing Aids and Other Audio Devices; US Patent Application, Publication No. 2002/0191804 A1, Dec. 19, 2002.
7. Bregman AS: Auditory Scene Analysis, *MIT Press* 1990.