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Real-Time Automatic Tuning of Noise Suppression for Cochlear Implants and Hearing Aids

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Technology description

Overview

This noise suppression methodology is computationally efficient and can be readily implemented on conventional CI and HA processors with no additional hardware required.

More than 219,000 people around the world have received CIs and millions more rely on hearing aids for an improved quality of life. The performance of these devices in terms of speech intelligibility has improved considerably over the years, however their performance in noisy environments still remains a challenge, with speech understanding shown to greatly degrade as ambient noise levels increase.

To address this issue, basic hearing aids utilize a generic, non-optimized noise suppression algorithm that results in very poor device performance in most noisy situations. CIs and more sophisticated hearing aid devices in the market allow the user to manually switch between different noise suppression profiles whenever the ambient noise situation changes, but this approach has a number of major disadvantages, including:

Users must switch their device noise suppression programs (or modes) manually when the environment changes.

The available number of different modes is limited due to the small number of switch settings possible on these physically small devices.

Users with limited finger dexterity, such as the elderly, find it difficult and cumbersome to effectively actuate the small switches on conventional CI/HA devices.

Technology

To address the problems outlined above, researchers at the University of Texas at Dallas have developed the noise suppression methodology presented herein which automatically classifies the ambient noise environment and selects the optimum noise suppression parameters – all in real time and without any user intervention.

The system is currently designed to classify ambient noise into the following ten different commonly encountered noise categories:

Car Noise- noise from engine noise at low and high speeds as well as AC noise

Office Noise- typing, mouse clicking, and occasional copier/printer sound in the background

Apartment Noise- living room noise with TV on with occasional noise from dishes and AC noise

Street Noise- moving traffic and wind noise

Playground Noise- kids screaming, laughing in the background
Mall Noise- music played in stores, babble noise with reverberation
Restaurant Noise- babble noise mainly due to music and dishes
Train Noise- engine noise and the rhythmic noise made by wheels on railing
Flight Noise- engine noise together with air noise
Place of Worship Noise- people whisper, praying with occasional bell sound in the background
Because the system automatically selects and applies the optimum noise suppression parameters for a given environment, many more noise suppression profiles can be utilized than would otherwise be possible with conventional manually-switched devices, thus significantly improving the user’s overall speech intelligibility. It should be noted that if a user experiences a noise environment which does not fall into one of the above pre-defined categories, the system selects the noise class with the closest matching noise characteristics. Further, additional noise environments can be easily incorporated into the system if needed.
Great care was taken to ensure the present noise suppression algorithm is highly computationally efficient. This allows the system to operate in real-time, which is critical for the intended application. This also allows the algorithm to be implemented on conventional CI and HA processors, which possess limited processing power, without any additional hardware requirements. The adaptive noise suppression algorithm is written in the C programming language and can readily be compiled into any alternate language, as required by a given CI/HA processor. Further, the algorithm has been successfully demonstrated on a portable HP PDA research platform.
The following is a block diagram of the developed adaptive noise suppression algorithm, as optimized for use in a cochlear implant application. Please note that the illustrated process utilizes a recursive wavelet decomposition method, however this method can be easily replaced with the classical strategy of fast Fourier transform (FFT).
Perceptual Evaluation of Speech Quality (PESQ) is a worldwide industry standard test methodology for automated assessment of the speech quality as experienced by a user. The PESQ result range is 1-4, with 4 being best (pure speech with no noise). As can be seen in the PESQ results below, the present adaptive noise suppression technology provided significantly better performance as compared to both the no-noise suppression and the fixed-noise suppression systems. For example, for the playground environment, the PESQ improved from 2.3 with the fixed-noise suppression system to 2.6 with the adaptive system (a 13% improvement in speech quality).
Compile program code for given CI/HA processor platform
Optionally, add additional noise categories as desired

Advantages

Significantly Improves Speech Intelligibility	·Automatic classification of ambient noise and application of optimum filtering parameters. ·Accommodates a wide range of environment-specific noise scenarios.

Easy to Use	·System responds automatically – user doesn’t have to manually adjust switches.
Easily Implemented on Conventional CI/HA Platforms	·Algorithm is very computationally efficient. ·No additional hardware required.

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