An Improved Speech Processing Strategy for Cochlear Implants

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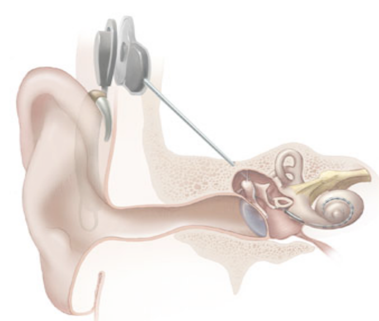
Plenty of people suffer from sensorineural hearing loss, in which the hearing loss is rooted to the vestibulocochlear nerve in the inner ear. People who suffer from sensorineural hearing loss typically receive a cochlear implant, which restores the hearing function through use of auditory neural prosthesis. This is done by electrically stimulating the auditory nerve and receiving the incoming pulse parameters which use “speech-by-speech” processors in the cochlear implant device. All in all, the cochlear implant reflects hearing in the peripheral nervous system. The cochlea in the ear decomposes and codes auditory information, and the cochlear implant uses a linear bandpass filterbank to do the same thing. A cochlear implant is unlike a hearing aid in that it doesn't amplify sound, but instead stimulates the still functioning auditory nerves in the cochlea, through the use of an electric field. The negative to these cochlear implants is that they often have high levels of noise, and are unable to replicate the quality of sound that the cochlea does naturally. The following study presents a way to reduce the noise levels and upgrade cochlear implants without extra surgery.

The model used was a dual resonance nonlinear model (DRNL), and is considered to be the best cochlear implant model because it uses a faster computational speed and eliminates feedback. The DRNL model consists of two pathways, a fixed linear filter and a nonlinear filter, which has a response variable that depends on the input signal level. This results in actively and nonlinear variables. Typically cochlear implants have problems detecting melodies, but this fine structure encoding attempts at correcting this, in addition to sound source localization and use of a small number of channels.

The acoustic simulation of the model was based on cochlear implant speech processing development. Using a one channel DRNL model, which consists of two pathways and two different center frequencies, and therefore a sum of two sinusoids are developed for each channel. Usually, cochlear implants are based on linear bandpass filters, and the waveform from the acoustic simulation is a result of the sum of multiple amplitude-modulated sine waves, with frequencies set as the center frequencies of the bandpass filter. In the case that was presented here, the acoustic simulation used two sinusoids per channel, rather than the one sine per channel. Ten people participated in the study, ranging from noisy to quiet surroundings.

The study showed that their processing system, based on the active nonlinear filter model, developed a more accurate speech comprehension. In conclusion, they found a way to improve the speech processor in cochlear implants through the use of nonlinear, active filter models. Conditions of high levels of noise were studied at different frequencies to obtain a better understanding on how cochlear implants may be pushed further in recognition of various sounds. Although no changes have been made yet, the study showed what was capable of these cochlear implants in noisy situations using only software upgrades, so that a change in hardware wouldn't be necessary.



<u>Reference:</u>

Kyung Hwan Kim, Sung Jin Choi, Jin Ho Kim, Doo Hee Kim. An Improved Speech Processing Strategy for Cochlear Implants Based on an Active Nonlinear Filterbank Model of the Biological Cochlea, *IEEE Engineering in Medicine and Biology Magazine,* Volume 56, Issue 3, March 2009.