ABSTRACT

For people with hearing loss, the function of Hearing Aids is not just for raising the volume. When used, this tool should also be able to make users hear conversations more clearly, without concerted efforts, especially in crowded circumstances, and prevent the inconvenience if the sound is too noisy. On digital hearing aid, a voice signal is processed to produce; SNR improvement, flexible gain addition, digital feedback reduction, etc.

This final project aims to simulate digital signal processing of a simple hearing aid in non-real time. With the input of a human voice files, signals are mixed with AWGN (Additive White Gaussian Noise) - a disorder of the system that cannot be avoided, and the background noise originating from the environment research model, then processing frequency shaping, noise reduction, and Amplitude compression are done.

Frequency shaping aims to select the cuttoff frequecies and band issolation then adjusted the hearing of people with hearing loss. At block of noise reduction, using FIR adaptive filter, with the long smoothing constant and the coefficient adaptation is determined by RACE (Real-Time Adaptive Correlation Enhancer) algorithm. Then, the signal is processed in blocks of amplitude compression that is used to overcome the discomfort of the patient.

This Reasearch shows the result that the RACE algorithm is good enough to be used as processing algorithm to reduce noise in the Hearing Aid. The Highest SNR 51,8522 is got from the biggest β value, it is 0.9.

Keywords: DSP Hearing Aid, Noise Reduction, Adaptive filters, RACE Algorithms, amplitude compression, speech signal.