ABSTRACT

Nowadays, hearing loss in both one or two ears happens on many people. Commonly, people with hearing loss solve it by using hearing aid. Function of hearing aid is to make a weak sound becomes audioble and to reconstruct the sound so communication will go properly. But there’s still a problem in using hearing aid, it’s a noisy sound that cannot be ignored by the users.

Noise is an unwanted accoustic signal in a system. This thesis has simulated digital signal processing in hearing aid. In this process, a clean signal mixed with noise and an environtment noise signal are taken as a sample. After taking samples, the next proccess are noise reduction, frequency shapping, and amplitude compression.

Frequency shapping is used to choose frequencies which later will be adjusted with hearing frequency of people with hearing impared. Noise reduction block uses FIR adaptive filter which its adaptive coeficient is taken from RLS algorithm. Reasearch has proven that there is a significant noise reduction. Output signal of the system is measured using two methods, subjective and objective. Subjective measurement parameter is MOS with degradation category rating and objective measurement parameter are PSNR and MSE.The optimum PSNR is 10,0955 which is achieved at filter orde 65 and forgetting factor value 0.99. The smallest MSE is achieved at 0.09782. Computation times that system achieve is 57,1989 second.

Keyword : *DSP Hearing Aid, Noise Reduction, RLS method, Amplitude Compression, Frequency Shaping*