ABSTRACT

Nowadays the function of hearing aid is not only to increase the volume. When hearing aid is being used, it should be able to allow user hear a conversation and a speech much clearer, especially in a noisy environtment so the user will not feel uncomfort by hearing a noisy speech. In digital hearing aid, an audio input produces; increased PSNR; hearing frequency adjustment and so on.

This study has simulated the digital signal processing in hearing aid by using a non-real time method. Input of for this systme is voice samples taken in research environment using two microphones. Primary microphone is used to record the sample of speech and noise then the secondary microphone is used to record the environment noise. The next steps are noise reduction, frequency shapping, and amplitude compression.

Output signal of the systme is measured objectively and subjectively. Parameter for the objective measurement are PSNR and MSE . Parameter for subjective measurement is MOS with degradation category rating method which the value of 4 is got from the most respondent . Optimum value of PSNR is 17.68 dB and optimum value of MSE is 0.0170 which both were achieved at filter orde 110.

Key word : DSP Hearing Aids, Noise Reduction, Wiener Filtering, Amplitude Compression, Speech Signal