ABSTRACT

The interference of background noise is the single greatest problem reported by hearing aid wearers. The background noise reduces the intelligibility of speech and that the greater the level of background noise the greater the reduction in intelligibility. Its possible to understand speech in a moderately noisy environment because speech is a highly redundant signal and thus even if part of the speech signal is masked by noise, other parts of the speech signal will convey sufficient information to make the speech intelligible, or at least sufficiently intelligible to allow for effective speech communication.

There is less redundancy in the speech signal for a person with hearing loss since part of the speech is either not audible or is severely distorted because of the hearing loss. Background noise that masks even a small portion of the remaining, impoverished speech signal will humiliate intelligibility significantly because there is less redundancy available to compensate for the masking effects of the noise. As a consequence, people with hearing loss have much greater difficulty than normally hearing people in understanding speech in noise.

Hearing aids allow for some degree of signal processing for reducing the effects of noise. It is vital that the effects of listening in background noise be taken into consideration not only when designing new hearing aids but also when giving out modern hearing aids. It is very important to perceive and understand speech in noise. Deafness is very tedious conical disability because of either sensorinural defect in which cells are dead because of age or some major diseases. Deafness can also be caused because of problem in born and air conduction. Always getting solution with cochlear implant is not giving success because of some disadvantages in the handling of the process. Widely used approach is use of digital hearing aids and process the speech for noise reduction as well as for individual frequency band enhancement.

The frequency spectra of everyday noises are seldom so different from that of speech and are sufficiently time invariant that a fixed filter can effectively not eliminate most of the noise without reducing speech intelligibility at the same time. It is possible to use
adaptive filtering to reduce noise levels without a significant reduction in intelligibility. The method is to obtain an estimate of the noise spectrum in some way and then to attenuate those frequency bands in which the noise exceeds the speech.

In the proposed work information of the noise can be extracted from the silences of the speech. As silence carries no portion of speech or information easily statistical characteristics of noise can be calculated and based on that filter can designed. Algorithm used for pauses detection is known as a voice activity detection algorithm and is based on zero crossing rate measurements and energy vector measurements.

LMS, NLMS and RLS are designed for the proposed task of the noise reduction. Different noises are taken for the reference which is most likely to be generated with the speech in practical scenario and it can be removed successfully. Prediction of the noise parameters is performing important task in the adaptation. Mentioned algorithms are compared for their characteristics like amount of noise reduced, estimation of filter weights, convergence rate of the filters, MSE and PSNR etc. Developed algorithm gives significant result in the noise reduction and can be observed in time domain also as well as subjective and objective tests are also giving significant results.

Most of time deaf patient are not able to recognized all the frequency equally. Frequency response of the ear of patient gives proper information about losses. Basically noised removed speech should be enhanced as per the audiogram structure. Cleaned speech is given for the enhancement in the individual frequency band in the wavelet domain. As per the requirement of the audiogram for some of the people individual frequency components are less sensible so only that bands are amplified in the loudness by the processing of the speech using multi resolution approach of discrete wavelet transform.

Novel approach is carried out for speech enhancement in the noise reduction by using adaptive filtering and frequency band enhancement in the wavelet domain in the present research.