Chapter 6

Conclusion and Future Scope

6.1 Conclusion

One of the key points verified in this research is that speech enhancement does result in improvement in the perceptual quality of the speech and it should be applied to noisy speech. However, the noise problem can never be completely solved, though it can be reduced to a tolerable level. For high SNR scenarios, the output quality can be quite good. For very low SNR situations, the output quality improves but it still suffers from either residual noise or distortion to speech or both. Speech enhancement aims at improving the perceptual quality and intelligibility of a noisy speech mainly through noise reduction. Speech enhancement may be applied to an industrial ambience, where lot of noise due to machine are mixed in speech signal and nearly inaudible, in this situation such speech enhancement machine can provide a proper noise free communication. The reported algorithms can be mainly classified as parametric and non-parametric. Parametric algorithms assume a model of the noise signal, whereas non-parametric approaches just need an estimation of the spectrum. The proposed algorithms in this thesis are considering both parametric and non-parametric behaviour.

Initially for basic adaptive filtering techniques LMS, NLMS is adopted and its performance is evaluated as SNR, MSE. Because of nonlinear and non-stationary behaviour of speech these traditional technique are unable to separate noise completely form speech, there will be different approach required which can be data driven. Hence EMD is adopted which decomposes such signals into zero mean oscillating components, referred as the intrinsic mode functions(IMF).IMFs give sharp and meaningful full identifications of the instantaneous frequencies.

Empirical mode decomposition for signal denoising is employed with the idea that noise has higher frequencies than the speech signal so that when the signal is decomposed with EMD initial IMFs gives the high frequency components of signal which can be easily removed, so the original speech signal does not lose its data even if noise is Gaussian or periodic since it
is decomposed in lower IMFs which remain untouched. In this process it very important to select proper number of IMFs for signal reconstruction so here also PSO algorithm is applied to select effective number of IMFs with an objective of highest SNR.

The results achieved from EMD+PSO based hybrid was not sufficient based on SNR and SQI performance evaluation parameter, for periodic industrial noise. Which further create a urge of blind source separation if industrial noise are there.

Since ICA is a very popular method for blind source separation, the research work was started with ICA and found that it effectively remove noise from the speech but it does not remove all the noise from the signal because in ICA condition for separation is that signals sources should be non-Gaussian in nature but noise always has some kind of gaussianity with it and that reduces the efficiency of ICA in industrial environment. To enhance the performance of system ICA separated signal is processed with DWT and found that SNR of signal improves but correct level of decomposition and perfect threshold value for shrinking is needed and that was found using particle swarm optimization algorithm. Results clearly state that the combination of ICA and PSO optimized DWT gives better result than the ICA.

System tested in 8 industrial noise cases depicts the superiority of proposed algorithm over conventional ones. The ‘lalala’ and ‘welcome’ speech signals have highest SNR and in rest of cases the SNR of proposed system is always greater than or equal to 100% from existing methods in domain of periodic noise. The rough averaging of results for periodic noise indicates that average SNR for ICA is 10 dB, ICA+DWT is 11 dB, EMD+PSO is 10.25 db while for ICA+DWT+PSO is 15 dB. In rest of cases and in periodic noise, the proposed technique has the highest SNR and Sound Quality Index. The difference in SQI has not significant and may not be satisfactory, yet for the range of input signals and noise environment, the optimization of PSO to filter the speech signals has best SNR compared to existing methods.

6.2 Future Scope

In future more categories of noises are prone to exist. With advancement in technology and people’s dependency on instruments is and will be serious concern for noise free environment. The noise reduction technology is yet to be upgraded for compatibility in infrastructure and living environment. Though the humans cannot differentiate types of noise
via naked ears, the noise reduction systems must be self-sufficient to encounter all possible
types of noise (audible or in-audible). The noise reduction techniques are being tested to cut
the background noise in telecommunication through adaptive methods. The approach and
results of this system is configured according to present scenario of environmental noise. For
signals with lower SNR, better algorithms such as ‘Digital Noise Reduction’ can play a vital
role in telecommunication. It is expected that more efficient algorithms shall monopolize the
technical aspects of communication. The computational complexity can be taken care in
future scope of work, the proposed algorithm hardware feasibility yet to recognize and can be
of further research interest.