CONCLUSIONS
CHAPTER 6

Chapter - 6. Conclusions

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6. CONCLUSIONS

6.1 CONCLUSIONS

In this research work, the author's intention is to recognize speech signals by considering three different feature extraction techniques like Linear predictive coding, Mel frequency cepstral coefficients, and Wavelet transforms and two Recognizing techniques like Dynamic time warping and Hidden Markov Model. Before recognizing speech signals, the speech signals are contaminated by noise to remove this two-traditional speech enhancement technique like Spectral Subtraction and Weiner Filter and four classical filters like LMS, NLMS, RLS, and Kalman filters used to enhance speech signals after removing noise present in the speech signals. Then speech features are extracted by using the proposed feature extraction techniques and Speech Recognition techniques for finding best Recognition accuracy for both stored templates and real-time speech to recognize Telugu digits.

From chapters three and four, when the input signal has no limits or no noise effects, the speech communication system performs greatly and is degraded when there is fairly large level of noise input signal. In such noisy cases, the system cannot meet the real requirements such as speech intelligibility, speech quality, or recognition rate. In this research work, among the all the filters such as SS, Weiner, LMS, NLMS, RLS, and Kalman algorithms which are used for speech
enhancement, simple and effective to implement is LMS algorithm but slower one.

Spectral subtraction process ought to be done carefully in order to avoid any talk distortion. Speech detection algorithm is needed to distinguish speech and noise frames. Computation complexity is more due to matrix inversion in weiner and also poor performance. Despite the fact that, with increased step size, the rate of convergence obtained in NLMS are not at acceptable level and even then, we have chosen the step size as 0.25 for LMS and 0.5 for the NLMS. Better convergence in LMS and NLMS with a step size of 0.5, but not up to the level of RLS and Kalman. The experimental results had shown that when compared to the other algorithms the Kalman as well as RLS algorithms provides better noise reduction at faster converging speed, improved speech quality and intelligibility respectively. As a result, kalman adaptive filter has more SNR as well efficient noise reduction than the other classical adaptive filters.

From Chapters five we know the effectiveness of LPC, MFCC, Discrete wavelet transform in feature extraction. The wavelet transform is a most dominant technique for speech processing than other techniques this is shown from results obtained. The features obtained by using the wavelet transform shows higher recognition. Wavelets are able to distinguish between different properties high frequency low amplitude spectral components and low frequency large amplitude spectral components.
During the simulation with LPC & DTW, MFCC & DTW, MFCC & HMM, Daubechies-8 mother wavelet for feature extraction. Here we have used level-4 and level-7 decompositions of db8 wavelets. The efficiency of speech recognition is increased with the level for recognition of isolated digits. In this research work we have consider a limited number of samples. The performance of the system can be improved by utilizing proposed noise reduction algorithms and training with large dataset. WAVELET & DTW, This combination of recognition system is better in real time recognition system because there is filter bank in wavelets this will avoid noise to enter the system this makes system to increase the recognition accuracy.

6.2 FUTURE SCOPE

The following future work is proposed.

- Investigation of the effectiveness of other features and feature extraction techniques for more robust Recognition.

- Investigation of other classification techniques.