**SPEECH RECOGNITION IN THE PRESENCE**

**OF WIDEBAND NOISE**

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Submitted in partial fulfillment of the requirements

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**CERTIFICATE**

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**ABSTRACT**

This thesis describes the implementation of speech recognition algorithm under noisy environment. Reducing the level of background noise is very important in many communication systems. For example, most communication system are used in environments where the communication system needs to operate in the presence of high levels of car noise or street noise. Noise reduction also improves the performance of the speech recognition algorithms increasingly employed in a variety of real environments.

 In this thesis we describe the implementation of its signal processing components, which provide a GUI-based environment to perform signal processing for speech recognition. This GUI-based configuration tool is presented in MATLAB. The tool described here deals with the Acoustic Front-end, which represent signal processing portions of a recognition system. More importantly, we present a thorough understanding and analysis for individual computational steps in the ASR algorithms.

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**ABBREVIATIONS**

ASR : Automatic Speech Recognition.

DTW : Dynamic time warping.

HMM : Hidden Markov Models.

IWR : Isolated word recognition.

CWR : Connected word recognition.

MFCC : Mel-frequency cepstral coefficients.

LPC : Linear predictive coding.

LVCSR : Large Vocabulary Continuous Speech Recognition.