VOICE RECOGNITION IN DUBBING ARRANGEMENT

This thesis is presented in partial fulfillment for the award of

The Bachelor of Electrical Engineering (Hons)

UNIVERSITY TECHNOLOGY MARA

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ACKNOWLEDGEMENTS

In the name of ALLAH, The Most Gracious who has given me the strength and ability to complete this project and thesis.

I would like to express my deepest gratitude to all the persons who have helped and shown their interest in completing this project, especially to my supervisor, Puan Rosnani Yahya for her guidance and patience in supervising and assisting my project. My gratitude also goes to En Ahmad Takiyuddin who had guide, support, advise and direct me throughout the development of the project.

Finally, to my family and all my friends who have put up with me through thick and thin during the completion of this project.

Thank you.

ABSTRACT

This paper highlights the voice recognition system using MATLAB. The main purpose is to identify the best suitable translator for English actors or actresses in English films or movies to other languages such as Malay, Thai or Philippines using the basic methods of digital signal processing.

Voices from different people are recorded and analyzed in the frequency domain. Their voices then will be compared with an actor's voice. Comparison was done using MATLAB and the closest voice to the actor was verified at the end of the process.

Among the topics discussed in this paper are the voice signal, which described how the speech waveform produced and the analog to digital signal processing. By transforming a voice signal into its frequency components, we can distinguish between speakers. Fast Fourier Transform described the mathematical transformation that the FFT computes.

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CHAPTER 1
INTRODUCTION

Signals encountered in real life are often in continuous time that is waveforms (or functions) on the real line. Signals processing method plays a central role in information technology and digital communication, in the efficient and optimal transmission, reception and extraction of information.

Speech defined as a sound signal used for language communication. Speech signal is non-stationary signals. In human speech communication, the voice generating mechanism provides a means for the talker to map each word into a distinct acoustic speech signal that can propagate to the listener.

Voice is the most method to communicate with one and another. It is one of the estimation analyses from speech signal. Voice signal containing information can be measured into two methods; one is through time domain and the other through the frequency domain. Voice signals have a different frequency variation and the frequency range for human voice is from 20Hz to 3.2kHz.

1.1 Voice Recognition

Although it is not strictly speech recognition, it uses some of the same techniques and is therefore allied with it. In voice recognition, the process of analysing a speech sample to identify the speaker, there is no attempt by the computer to understand what was said or even to convert the speech to a form that is understands to humans. The task is simply to compare the sample utterance to one spoken by the subject during a prior enrolment session to determine if, in fact, both utterances were spoken by the same