PROJECT REPORT

Sound Convertion using Fast Fourier Transform Algorithm

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INFORMATICS ENGINEERING DEPARTMENT

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APPROVAL AND RATIFICATION PAGE

PROJECT REPORT

Sound Convertion with Fast Fourier Transform Algorithm

by

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STATEMENT OF ORIGINALITY

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ABSTRACT

Voice Converter is a program where the user can change the octave of the input file to higher octave or lower octave. The file type that can be processed is wav file only. The wav file that will be used is 8bit wav file. This project is created with java language programming and use one dimension array, linked list and array list as data structure.

The one dimension array will be used to store the processed analog signal data from wav file and for processing the voice convertion of voice. The linked list and array list will be used to process the K-means Algorithm for clustering the signal data to determine what gender it is. For convert the input file from user , the project will using FFT(Fast Fourier Transform) algorithm.

Fast Fourier Transform is method to converts a signals from its original domain to a representation in the frequency domain. After doing various experiments with the program, the results show that the voice color characteristic that makes the different between female and male gender still cannot achieved. The only achieved experiments results just know the different concept of wav file data reading sample that if the sample rate (how much data sample readed each second) is 22050 with 20000 data sample is the same as 44100 sample rate with 40000 data sample, the FFT Algorithm that being used in this final project is still not the best method to change the voice because still cannot change the timbre(voice color that distinguishes between male and female) and has a lot of noise after processing, but the convertion the voice octave to higher or lower octave can be achieved.The noise of processed data of FFT Algorithm can be reduced using Low Pass Filtering Algorithm.

***Keywords** : Fast Fourier Transform, K-means, Low Pass Filtering.*

PREFACE

In order to gain practical Knowledge in the field of Computer Science, we are required to make report on Final Project “Voice Convertion using FFT Algorithm”. The Basic Objective behind doing this final project report is to fulfill faculty requirement. In this final project report I have included various concepts, effects and implication regarding “Voice Convertion using FFT Algorithm”.

In Chapter one, it was background, scope and the objective of this project. Chapter two will explain the data structure and the algorithm of this project. Chapter three will be planning anything that needed for this project and how to do the project. The chapter four and five are for analysis project, design, implementation of the concepts and also testing of the finished programm. Chapter six will be conclusion and further research of this project.

Doing this final project report helped us to enhance our knowledge regarding understanding the voice towards “Voice Convertion using FFT Algorithm” we doing undergo many experiments related with our topic concepts. Through this report we come to know about how to read the analog signal to digital signal of voice file, change the octave of voice file and reduce the noise of processed voice file.

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