VIRTUAL AUDIO PROCESSING LAB USING GUI
(GRAPHICAL USER INTERFACE)

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ABSTRACT

Recently, the wavelet transform has emerged as a cutting edge technology, within the field of audio compression. Wavelet compression technique provides a good compression scheme for audio processing, which is the current audio technique. In wavelet compression system, the wavelet transform selection and transform coefficient bit allocation procedure are design to take advantage of the masking effect in human hearing. They minimize the number of bits required to represent each frame of audio material at a fixed distortion level. This project studies the different effect of audio compression due to the different audio file, different wavelet families (Haar, Symlet and Daubechies) and different level of decomposition. An audio processing lab is implemented in Graphical User Interface (GUI) using MATLAB software. The audio compression coder is used to compress the signal. The results obtained were analyzed based on Signal to Noise Ratios (SNR) and Compression Ratios.

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CHAPTER 1

INTRODUCTION

1.1 PROJECT OVERVIEW

Audio processing covers many diverse fields, all involved in presenting sound to human listeners. Three areas are prominent which are high fidelity music reproductions (such as in audio compact discs), voice telecommunications and synthetic speech (where computers generate and recognize human voice patterns). While these applications have different goals and problems, they are linked by a common umpire, the human ear. Digital signal processing has produced revolutionary changes in these and other areas of audio processing [1].

When we speak of audio compression, we must distinguish between two different types which are lossless and lossy. Lossless compression retains all the information in a given signal, means a decoder can perfectly reconstruct a compressed signal. In contrast, lossy compression eliminates information from the original signal. As a result, a reconstructed signal may differ from the original. With audio signals, the differences between the original and reconstructed signals only matter if they are detectable by the human ear [2] [3].