AUDIO COMPRESSION USING FAST FOURIER TRANSFORM

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In the name of Allah S.W.T
Most Gracious Most Merciful

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ABSTRACT

This project presents an implementation of audio compression using the Fast Fourier Transform (FFT). The fundamental concept is to find innovative ways to reduce the data rate of the audio signal to as low as possible while still keeping the signal intelligible. In this project Fast Fourier Transform and Inverse Fast Fourier Transform are used for the compression and decompression of a speech signal.

This audio compression scheme is simulated using MATLAB 6.5. Simulations are performed for different Fast Fourier Transform sizes and different number of components chosen. Two different methods were used that are by retaining the first n-components and by retaining dominant n-components. The Signal-To-Noise-Ratios are computed for all the simulations and used to study the behaviours of the compression scheme using Fast Fourier Transform. The noise introduced in the signal (for various cases) is studied both by listening to the recovered signal and by the calculated Signal-To-Noise-Ratios.

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