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EE 382 V - Term Project Proposal

Audio File Compression / Decompression

Modern audio compression techniques have made major strides in the last 20 years. The primary driving force was the quantity of data in a native audio recording, and the limited bandwidth available for transmission. Furthermore, the compression of the audio data typically meant loosing something along the way. The quality of the encoding determines how easy it is to detect the differences between the original and encoded audio clip. In general, when an audio signal is compressed, it is broken up into different parts of the audio spectrum and samples are taken at various intervals to create a digital fingerprint. These samples are processed using a variety of techniques to remove the "unnecessary" data, to include inaudible frequencies. The resulting data is then compressed using various techniques to get the smallest file possible. Later, when the audio file needs to be reconstructed, it follows the reverse process, but due to the pruning and sample rate, the resulting file has lost some its information.

For our term project, we will be investigating some of these trade-offs and how they affect the quality of a compressed and decompressed audio file. We will also be evaluating how various adjustments affect the compression rate and performance of the pipeline given specific audio samples. Because the mp3 standard is very precise and would take an exceptional effort to program an encoder which can both create and read a file which has been properly encoded, we will limit our efforts to some of the major pieces of the pipeline. We will take a raw audio clip, pass it through the encoding process, then send it back through the same decoding pipeline and compare the quality of the resulting file versus the original. We will also compare the performance of each variation for compression rate as well as wall-time of processing in both directions.

The major stages of the encoding pipeline which will be investigated are the Filterbank where the audio spectrum is broken up into sub-bands. The MDCT (acronym for … Cosine …) which breaks each sub-band up into a set number of "lines". Those lines would normally get fed into a signal processing and distortion control system, but this portion will be omitted for this project. Finally the data goes through the compression algorithms to get into their final form. The Huffman encoding algorithm will be used in this study for compression.