Digital Signal Compression Project

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44.1 KHz Stereo Audio

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1 Introduction

The aim of the project is to attempt compression of audio signals digitally and evaluate the various techniques used in mordern digital compression techniques.
Phase 1 consisted of a study of Huffman encoding and entropy. We evaluated the efficiency of the Huffman encoding, in relation with the entropy. The effect of encoding differences instead of the values directly on the average length of the encoding was also studied and tabulated. Phase 2 comprised of implementing a predictive encoding algorithm, specifically the Linear Predictive Coding algorithm.

2 Sample Space

The sample space of sounds used consisted of animals sounds like cats, dogs, horse etc, musical instruments guitars, flutes etc, and vehicles like cars, vans etc.
All the sounds were of the specified 44.1 KHz frequency and of the .wav format, so that they could be read by the wavread function in MATLAB, and stereo i.e. they had two different sets of sounds for the left and right year separately.

3 Phase 1

3.1 Huffman encoding

The aim of this phase was to evaluate the Huffman encoding algorithm. The majority of the encoding was done on Python, which read a .txt file of values.
All of the sound files in the sample space were all stitched together into on large .wav file. This file was then read into MATLAB with the built-in wavread function. These values were written into the the .txt file to be read into the Python code.
The Python code uses these values and calculates the Huffman encoding for the following cases

1. Single Channel Left
   Considering only the left channel values
2. Single Channel Right
   Considering on the right channel values
3. Single Channel Both
   Considering both the channels with the Left channel values concatenated by the right channel values
4. Difference encoding Left
   Considering the difference between the current and previous samples of the Left channel
5. Difference encoding Right
   Considering the difference between the current and previous samples of the Right channel
6. Difference encoding Both
   Considering the difference between the current and previous samples of both channels with the Left channel values concatenated by the Right channel values
7. Encoding of difference between Left and right
   Considering the difference between of the Left and Right samples at the same instant
8. Difference Encoding of difference between Left and Right
   Considering the difference between the current and previous values of the difference between the Left channel and the Right channel values

3.2 Observations

3.2.1 Single Channel

Channel	Entropy	Average Length
Left	13.287078	13.316172
Right	13.347654	13.376184
Both	13.322456	13.351348

3.2.2 Single Channel - Difference Encoding

Channel	Entropy	Average Length
Left	9.771333	9.798181
Right	9.837859	9.864783
Both	11.396513	11.423576

3.2.3 Double Channel

Type Encoding	Entropy	Average Length
Normal Values	10.833883	10.869549
Difference Values	8.457565	8.497198

3.3 Conclusions

The following can be concluded based on the results tabulated above.

- Encoding the difference between successive samples yields better results than just encoding the values
- Encoding the difference between the left and right samples gives better average length than when we just concatenate the and treat them as different entities
- Putting the above two points together and encoding the difference between the succesive samples of the difference between the left and right channel samples yields the best average length and entropy

LPC Analysis on Audio Files

4 Linear Predictive Coding

Linear Predictive Coding is a technique used for compression of the audio signals. LPC coding generally gives high compression ratios as the compression depends on the number of bits used for quantization of the error and the number of bits for quantization for the reflection coefficients.
Usually, higher bit values for the quantization for the reflection coefficients are sent and lesser number of bits for the quantizaiton for the computed error.

5 Implementation

The sound file in .wav is read into MATLAB and then the LPC coefficients are extracted for a window of length 20ms. The successive LPC coefficients are extracted for non overlapping segments of the window.
Based on the extracted coefficients of the single channel of the wav file, the error was computed using the original and the quantised output from the predictor.
After obtaining the error, it was also quantised to levels based on the number of bits selected for the quantization of the error coefficients.
Post this, the code was run for different window length to check the effect. This was done on single channel where the previous bits of the same channel are used to generate the prediction of the next sample of the input signal.
Later analysis was made using the same LPC technique using p/2 coefficients of first channel and p/2 coefficients of the next stereo channel, (where p is the order of the LPC filter) to predict the next coefficient of the first channel from the LPC coefficient output. This was done to find out whether the correlation between the samples assist in improving the output at the decoder. Since the values of the left and the right channel will be close to each other may be less coefficients are required to generate the output.

5.1 Comparison of Compression Ratios

No. of quantization bits for excitation values	1 channel	2 channel
4	9.24	0.508131
8	6.82	0.507461
12	5.72	0.507001
16	2.045	0.506639

Table 1: Comparison of compression gain for various number of bits used for quantization of the excitation coefficients of the LPC codec

5.2 Variation of Compression with order of LPC predictor

Order of Predictor - p	1 channel	2 channel
4	0.635818	4.282102
8	0.636249	1.491426
12	0.636121	0.882646
16	0.636088	0.634654
20	0.636088	0.524820

Table 2: Comparison of Compression ratio for various values of the order of the LPC filter

Order of Predictor - p	1 channel	2 channel
4	1.42	1.62
8	1.79	1.634
12	1.87	1.633
16	1.07	1.6331
20	1.17	1.6332

Table 3: Stereo Audio Wav file of Dog Comparison of bits

5.3 Comparison of Signal to Noise Ratio for various values of bit resolution for error quantization bits

No. of quantization bits for error value	1 channel SNR(dB)	2 channel SNR(dB)
4	-17.4345	-18.1879
8	-3.4361	-3.1173
12	-3.0688	-3.1107
16	-3.0994	-3.0848
20	-3.0987	-3.0847
24	-3.0987	-3.0846

5.4 Comparison of Signal to Noise Ratio for various values of filter order of LPC filter

Filter order	1 channel SNR(dB)	2 channel SNR(dB)
4	-1.8927	-1.8428
8	-2.9023	-2.8310
12	-3.2364	-3.1870
16	-3.0994	-3.0848
20	-3.0565	-3.0511
24	-2.8068	-2.7900