Ryerson University
Department of Electrical
& Computer Engineering
Faculty of Engineering & Architectural Science

Course Title:	Ele 725
Course Number:	Ele 725
Semester/Year (e.g.F2016)	W 2020

Instructor:	Dr. Naimul Khan

Assignment/Lab Number:	1
Assignment/Lab Title:	Sampling and Quantization

Submission Date:	07/02/2020
Due Date:	07/02/2020

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Reset Form

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ELE 725 Lab 1 Report

Abstract

Signals data are large which creates the need to compress the signal size reaches a certain point as transmission though a medium such as the internet has a certain bandwidth limit. Paper will explore audio processing though the usage of MATLAB. This paper will explain how to use MATLAB to read audio file and split it into its properties, down sampling an audio signal, and quantization of a signal though uniform quantization and Mulaw Quantization.

Introduction

In audio sampling, a balance between decompression and signal integrity must be achieved for an effective audio compression. As the audio signal is compressed more and more which means less samples are taken, the reconstruction of the signal becomes less accurate as there is less data to reconstruct with and the signal becomes distorted. To be able to keep compression high while keeping quality, the human ear is considered as the media is meant for humans. This allows designers to ignore certain high frequency ie more than 20Khz for more precision on lower frequencies.

Theory

For a signal, the Nyquist-Shannon sampling theorem explains the highest frequency that can be accurately represented for a specific sampling frequency.

Sampling frequency >= 2*maximum signal frequency or

Wb < Ws/2

If the Shannon theorem is not meet, the higher frequency of the signal start to add noise into the lower frequency components.

Methodology

Part 1

The audio information was obtained though the MATLAB function audio info.

Part 2

To down sample, we call the DownSample function I created which down samples the data and reconstructs it. The function reads the file and then down samples either with the function down sample or decimate depending on whether to use a filter or not and writes it to file. It reconstructs the signal by the interp command and writes it to file.

Part 3

For uniform quantization, I get the maximum and minimum of the 2-d matrix in the case of multiple channels. I use the max and min to calculate the step size which is stored in q. In which step the value is stored in is calculated and stored in the temp matrix. The temp matrix is multiplied by step size to get value and error is calculated for the graph. Immse is used to get mean square value of temp and Xn. Temp is written to file.

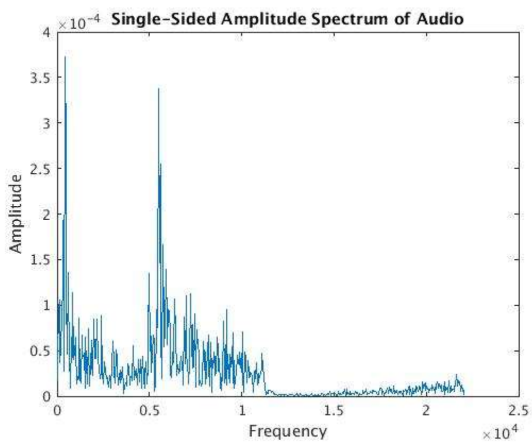
For mulaw, the audio file is transformed by the mu law formula. It is then uniform quantized which follows the same step as above. It is finally inversed by the inverse mu law formula and written to file.

Results

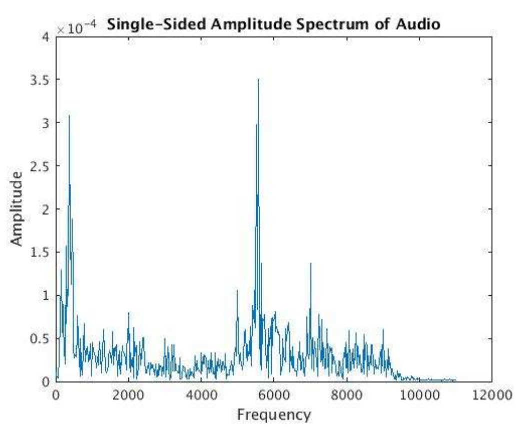
To calculate the file size

Total samples = 116736
116736*16 bits per sample= 1867776 bits
186776 bit / 8 = 233472 bytes
233472 *2 channels= 466 944 bytes /1024 = 456 kb

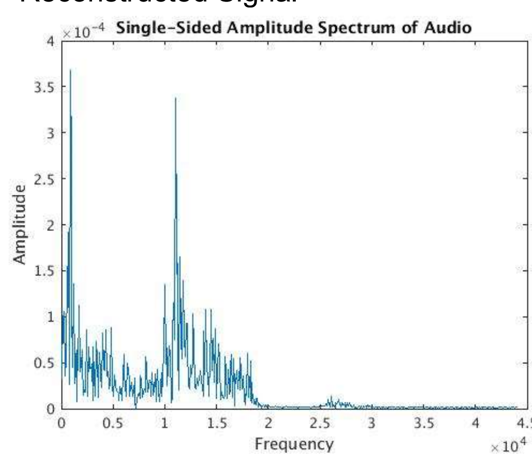
Original Signal



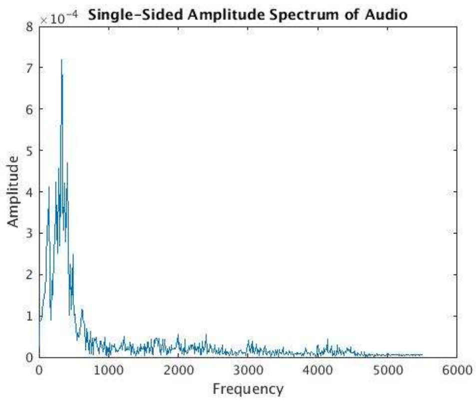
Down sampling by factor of 2



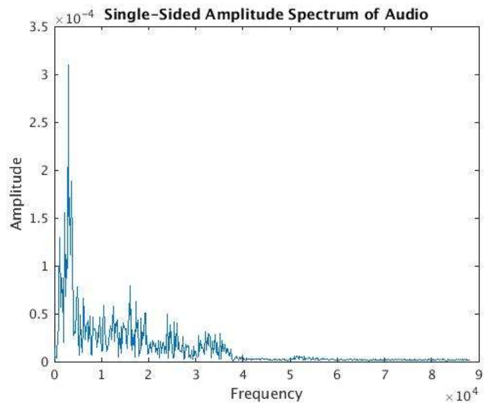
Reconstructed Signal



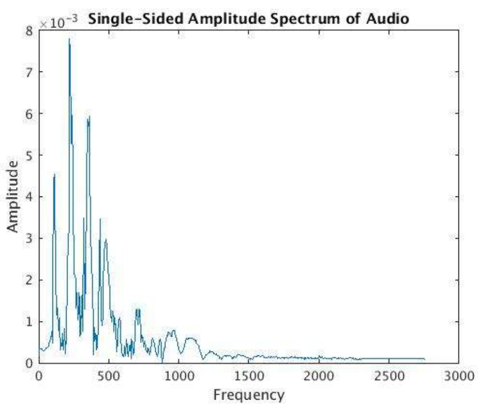
Down sampling by factor of 4



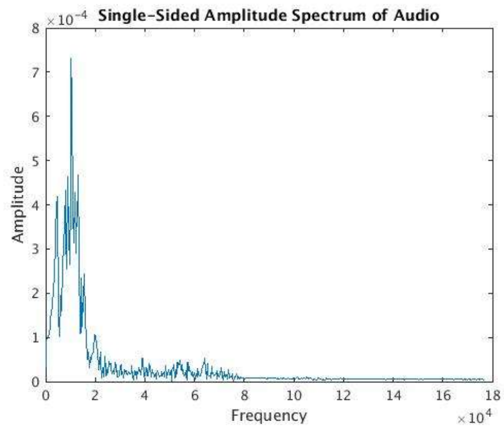
Reconstructed Signal



Down Sampling by factor of 8



Reconstructed Signal

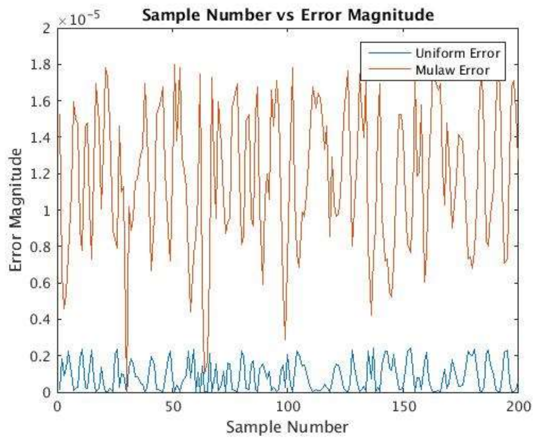


MSE 2:
ans = 8.6470e-04
ans = 0.0016

MSE 4:
ans =7.7832e-05
ans = 0.0016

MSE 8:
ans = 4.0305e-07
ans = 0.0016

Sampling Error Magnitude vs Sample Number



Discussion:

The audio file size told by the operating system is the same as one calculated. In the graph, as the down sampling factor increases the difference between the reconstructed signal and the original signal. The reason is that as the down sampling factor increases, more and more frequency exceeds the nyquist frequency which cause more aliasing which makes the signal more distorted. For uniform quantization, as the bits increases the error decreases. The reason is because as the number of bits increases then the range each bit represents decreases, which increases accuracy and decreases error. The mu law error was the same for the three different bit representation because they mapped the bit to similar locations in the low frequency so judging by this fact, the signal just did not have any high frequency component. The quantizer that yields smaller error for small signal values is the mulaw quantizer as it puts an emphasis on low frequency components so small frequency values will have more precision while the higher frequency values have big error. For larger signal values, it is better to use the uniform quantized as the higher frequency values and the low frequency values need to both be considered so we can not place an emphasis on a range we want.

Conclusion:

In conclusion, the finds of the lab were according to theory. All the theory parts were tested and affirmed.

Reference

Course Lecture Notes