Transform Coding

Audio Coding

For this assignment my main resources were the slides from lecture 5, the description for the assignment, and the DCT Wikipedia page. (https://en.wikipedia.org/wiki/Discrete_cosine_transform)

For part 1 of the audio I coded DCT for a 1-dimensional array (vector) Represented

as

$x_k = q_k \cdot y$

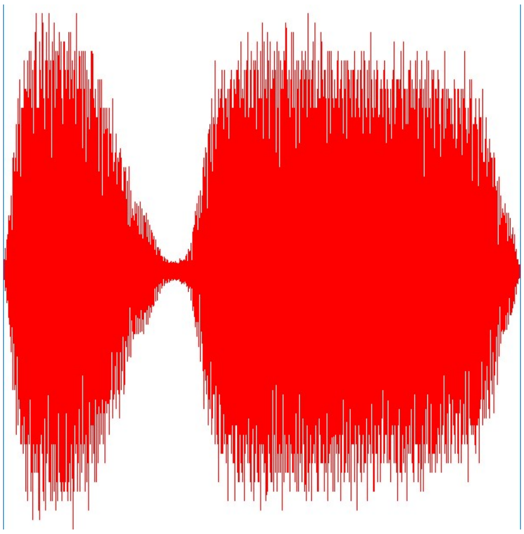
Which is done in my code in function *DCT* and *dotProduct* is a function that is given.

Inverse DCT as it appears in the slides:

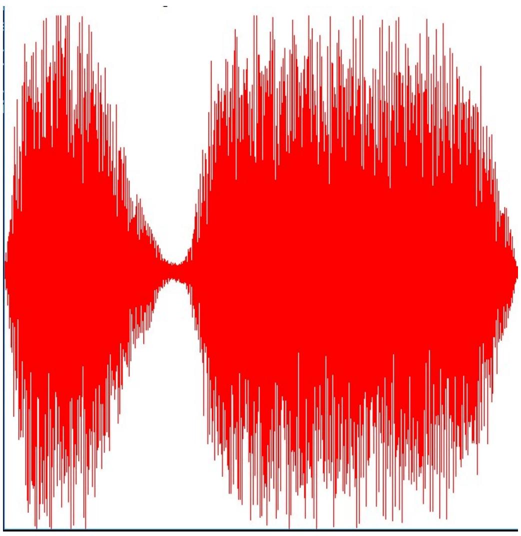
$y = \Sigma x_k q_k$

and is represented in the *InverseDCT* function

Where I zeroed out the higher placed vectors representing the higher pitches, in the function compressWAVSignal.



Before compression



After compression

Image Coding

In the function *CompressImage* I break the image into 8 by 8 blocks which are then sent to *CompressBlock* to actually be compressed using DCT and DCT inverse

For the image block, as described by the assignment I created a temporary C array, where I store the summations from using our DCT calculation. Then moving it into the output of pointer of B, where I use Inverse DCT to get the decompressed values back.

No Compression

M=12

M = 7

M = 3