Lab 3 Report

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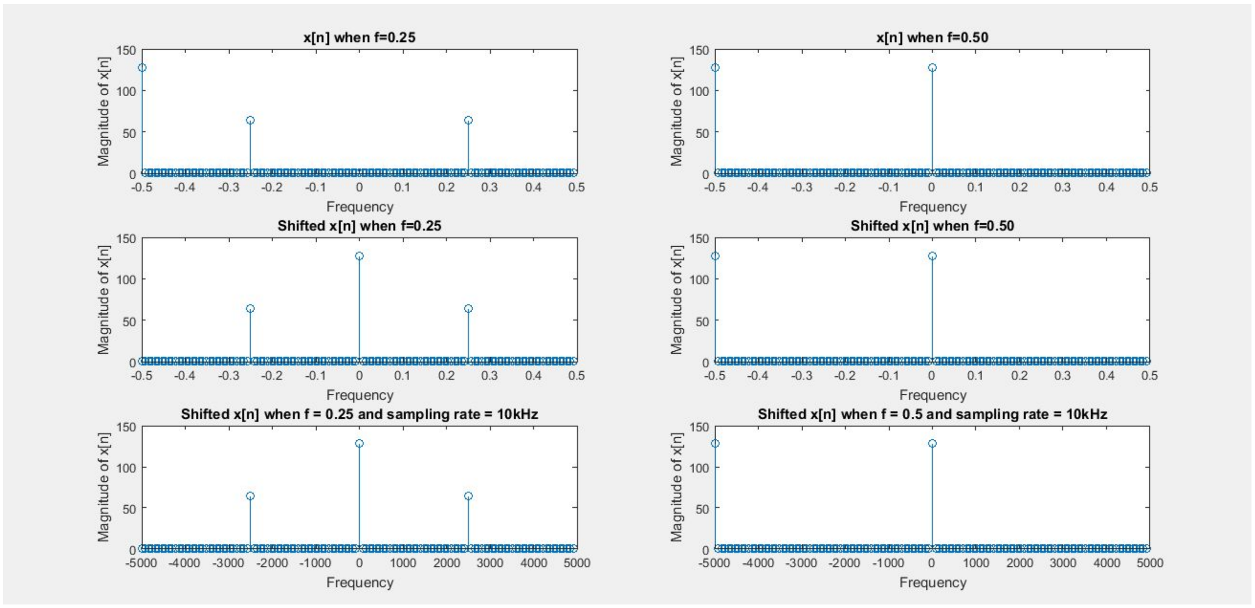
Assignment 1:

In this assignment, we learn about the function fft and fftshift. We were given a formula: x[n] =1+ cos(2π fn); 0 ≤ n ≤127. For our case, fn corresponds to two value, f1= 0.25 and f2: 0.5 . In our lab we plotted 3 graphs for each fn. The first graph plotted is the unshifted DFT of the formula above. Here, we just used the fft function and take the absolute of the output function. We did the same thing for fn=0.5. We plotted this against a normalized frequency from -.5 to .5 Hz. For the shifted DFT section, we use the fftshift function and take the absolute value of the output. We did the same process as the previous one in which we plotted against a normalized frequency from -.5 to .5 Hz. For the third one, we did it the same as the shifted DFT section. Instead of using a frequency of -0.5 to 0.5,we used a sampling rate of 10 kHz which makes the bounds from -5000 to 5000 Hz. The picture below shows the output when assignment 1 is run.



The frequency peak location makes sense as when f = 0.25, the peak is at -0.25 and 0.25. When we use fftshift function, we centered the frequency at w = 0. When we use a sampling rate of 10 kHz, the frequency peaks is 0, -2500 and 2500.

The frequency peak location makes sense as when f = 0.5, the peak is at -0.5. When we use fftshift function, we centered the frequency at w = 0. When we use a sampling rate of 10 kHz, the frequency peaks is 0 and -5000.

Assignment 2:

In this assignment ,we use learn about filters and the plot the following functions

a) $x_1[n]= \mathrm{sinc}(f_1(n-32))$

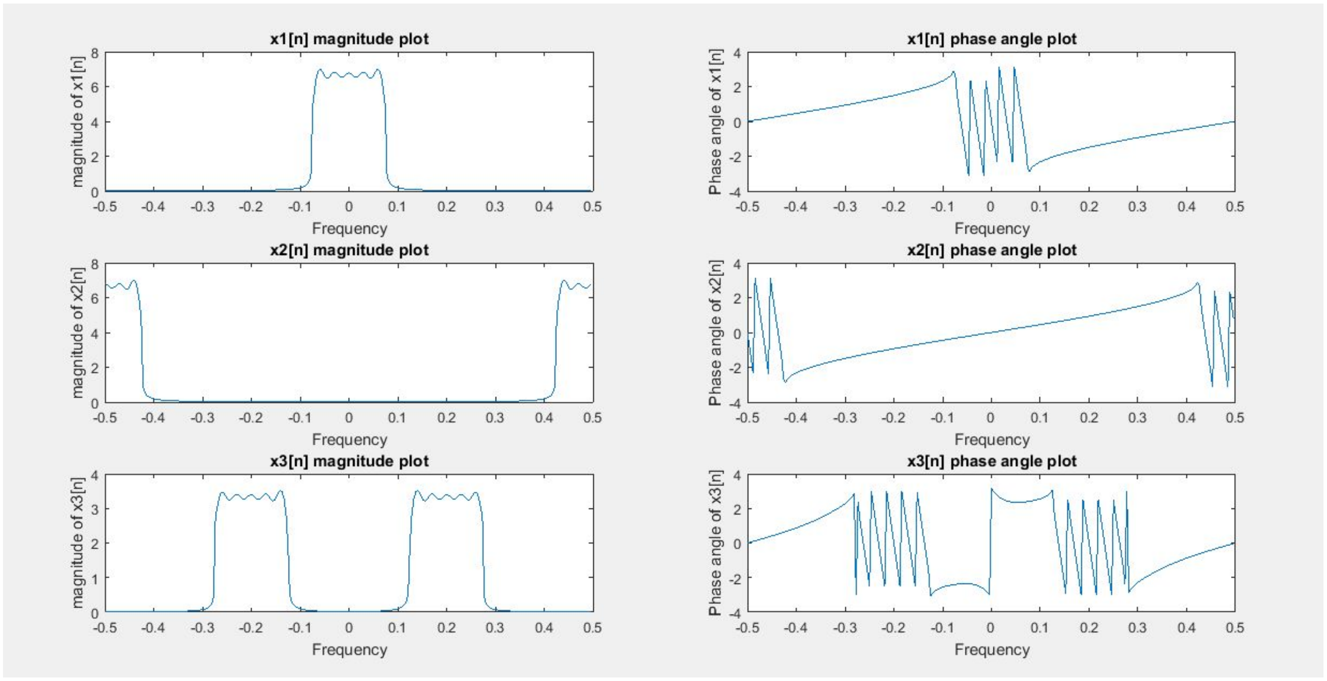
b) $x_2[n]= \mathrm{sinc}(f_1(n-32))(-1)^n$.

c) $x_3[n]= \mathrm{sinc}(f_1(n-32))\cos(2\pi f_2 n)$ where $f_2 = 0.2$

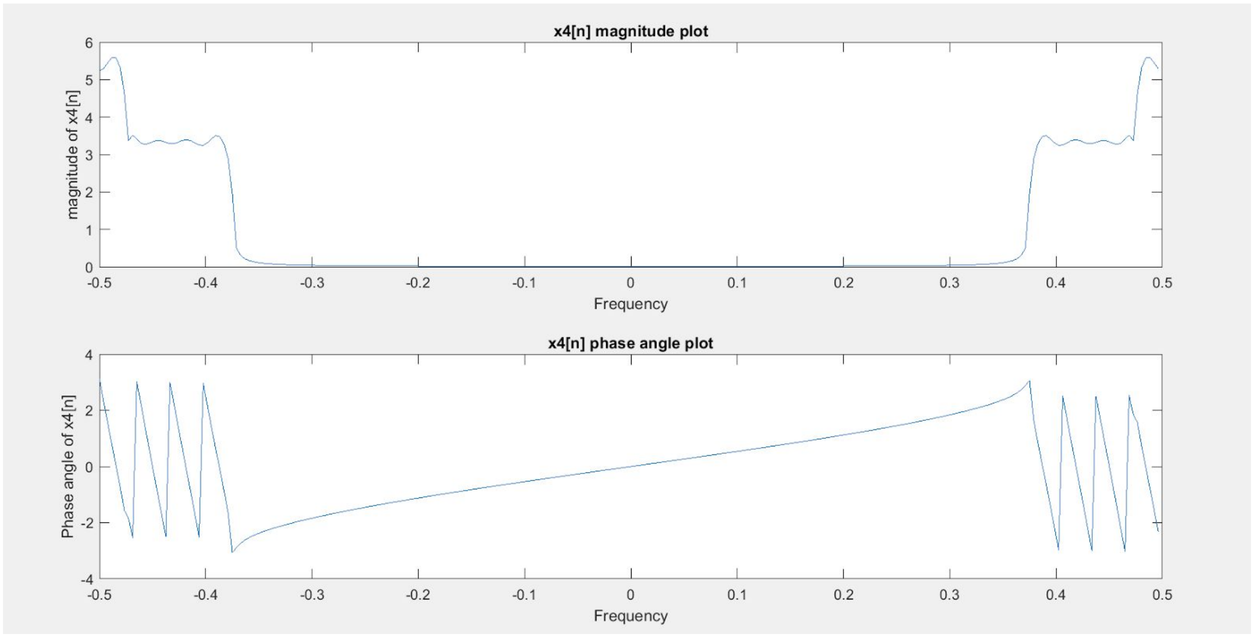
d) $x_4[n]= \mathrm{sinc}(f_1(n-32))\cos(2\pi f_3 n)$ where $f_3 = 0.45$

For each of the function above, we used the function of fftshift and then take the magnitude(using absolute value) and the angle(angle function). We plotted the magnitude and the angle against the normalized frequency.(-0.5 to 0.5)

When assignment 2 is run, the following figures are shown.

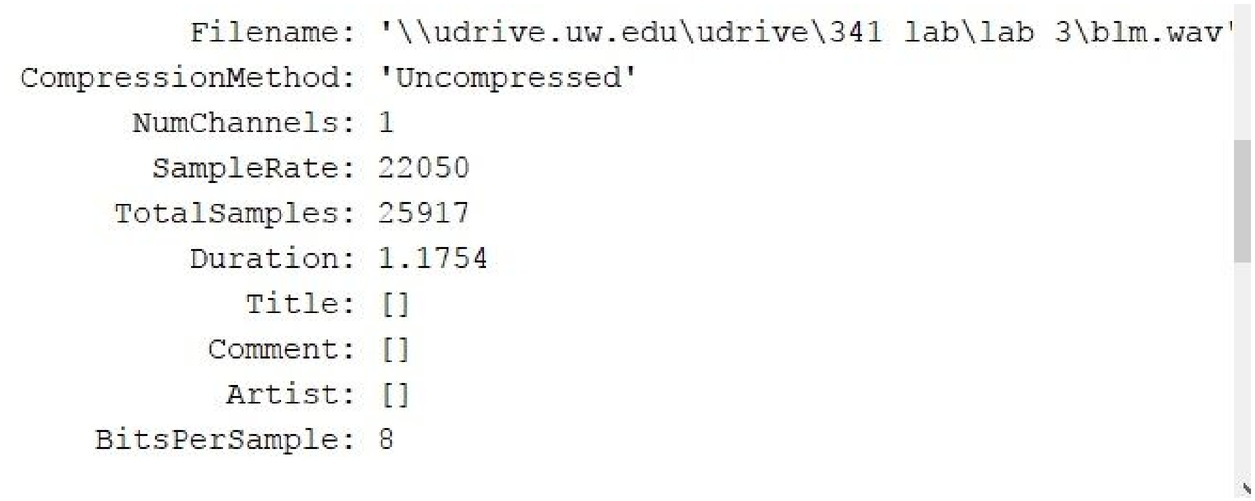


From this, we can see x1 is low pass filter, x2 is a high pass filter and x3 is a bandpass filter. Part d does not have a flat frequency response because it suffers from aliasing and therefore there is some parts which have a higher magnitude. The magnitude and the angle of part D is shown below.



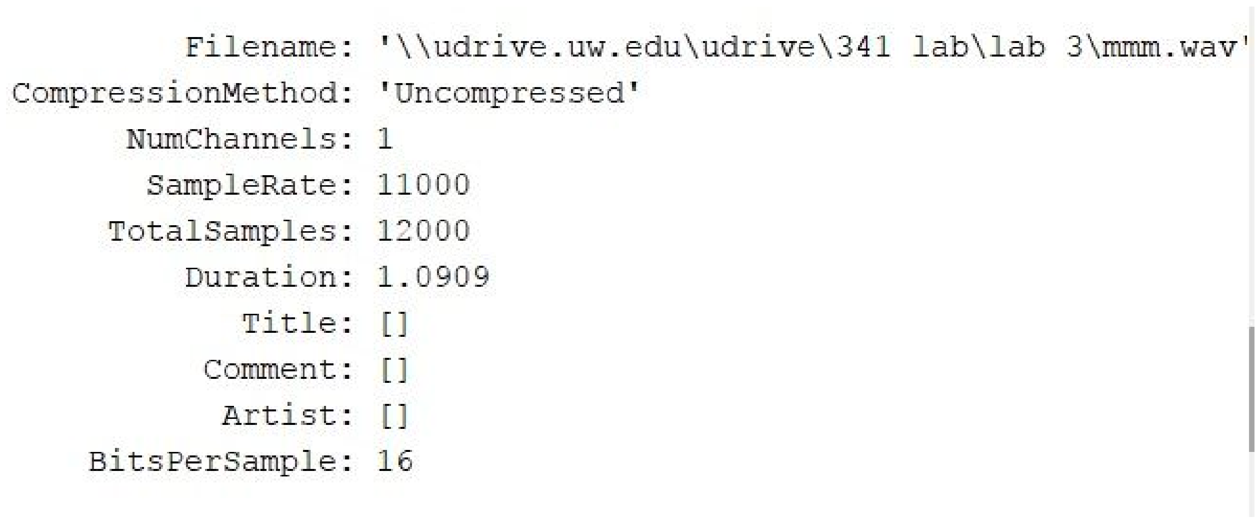
Assignment 3

For this part, we were using mmm.wav and blm.wav. In the beginning of this part, we did create our time vector, load the sound via audioread() command, play the sound via soundsc() command, and extract audio info using audioinfo() command. This is info for BLM:



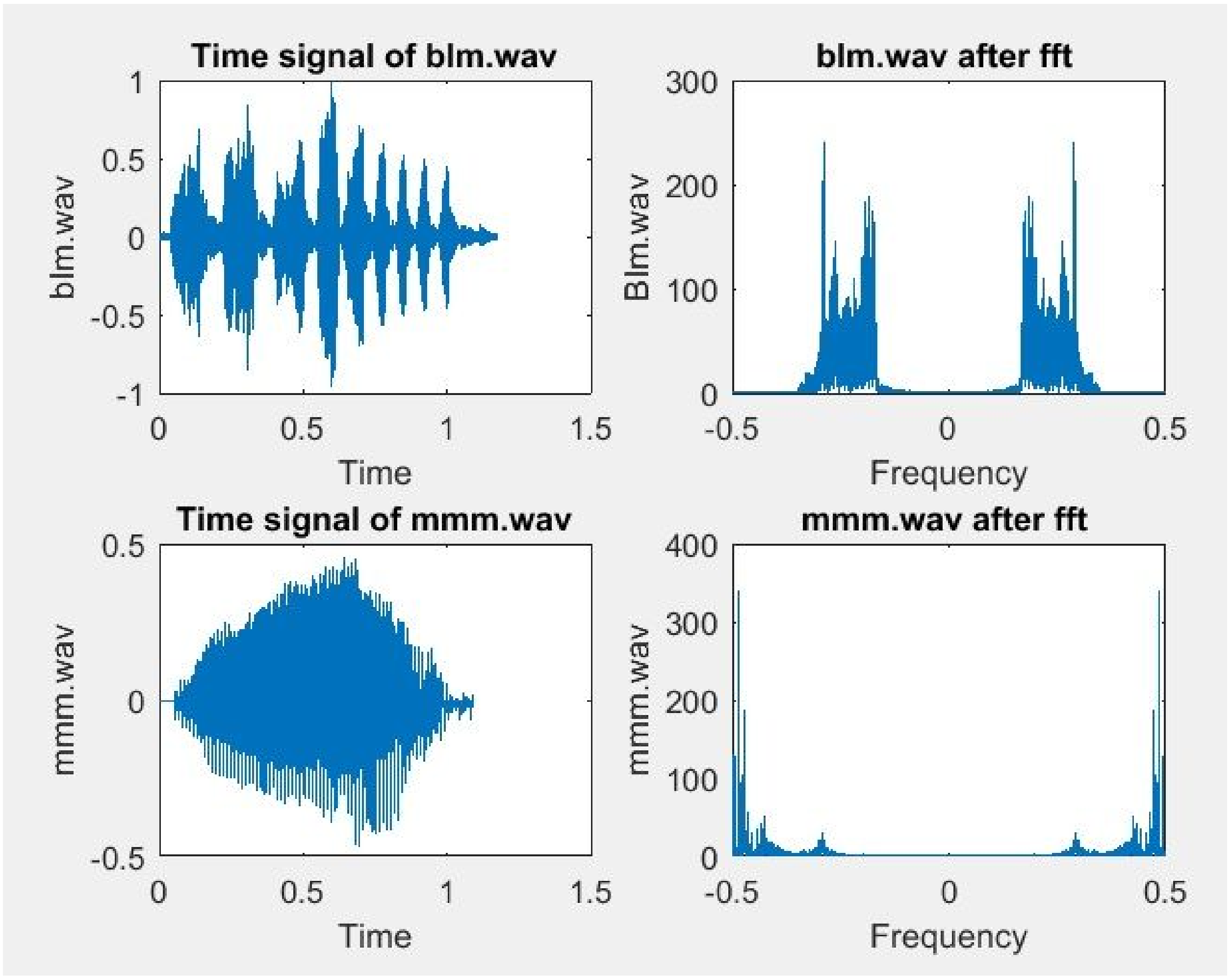
```
           Filename: '\\udrive.uw.edu\udrive\341 lab\lab 3\blm.wav'
  CompressionMethod: 'Uncompressed'
        NumChannels: 1
         SampleRate: 22050
       TotalSamples: 25917
           Duration: 1.1754
              Title: []
            Comment: []
             Artist: []
      BitsPerSample: 8
```

And this is info for mmm:



```
           Filename: '\\udrive.uw.edu\udrive\341 lab\lab 3\mmm.wav'
  CompressionMethod: 'Uncompressed'
        NumChannels: 1
         SampleRate: 11000
       TotalSamples: 12000
           Duration: 1.0909
              Title: []
            Comment: []
             Artist: []
      BitsPerSample: 16
```

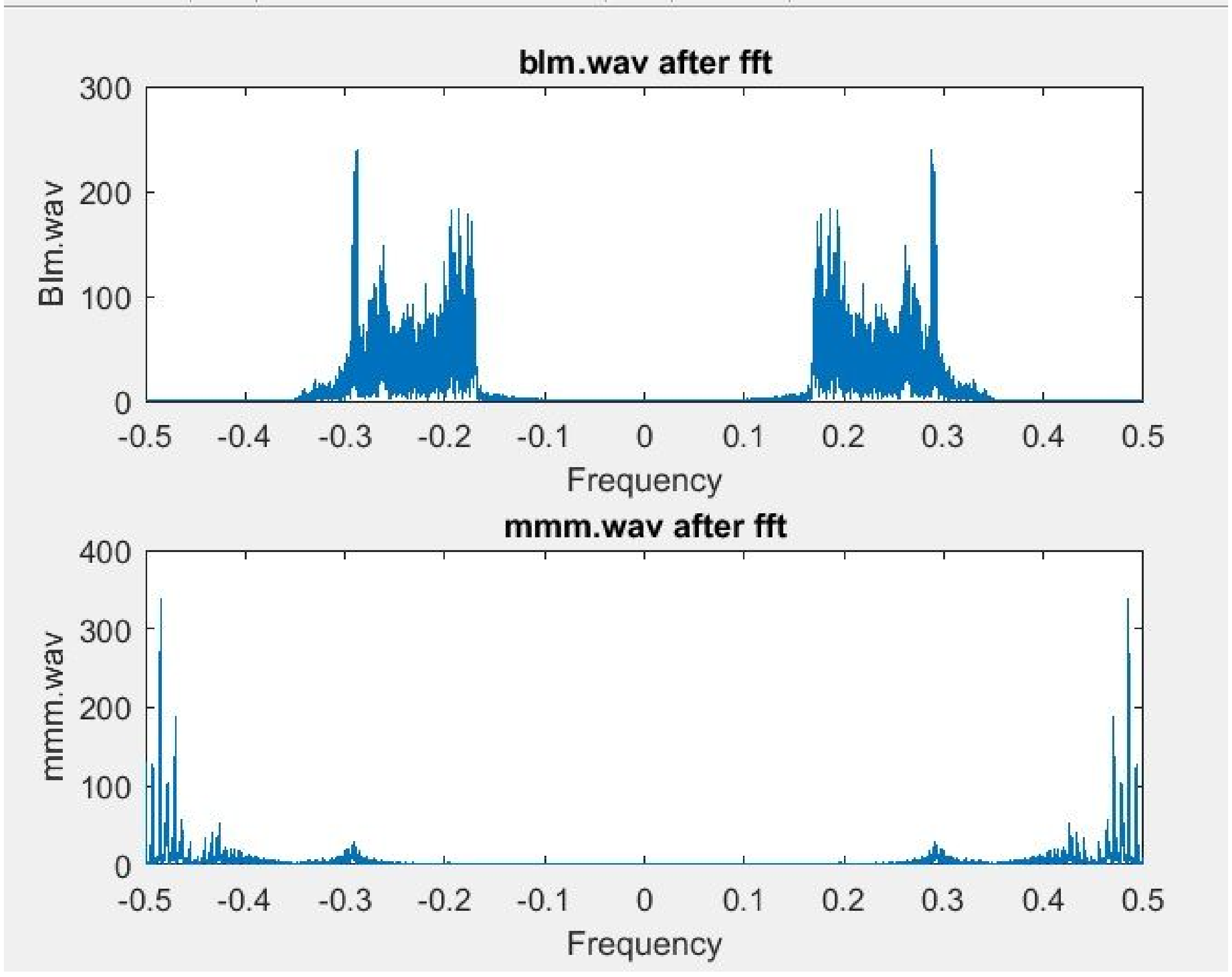
We also successfully fft our signals:



And yes, the frequency of the content matches our expectation

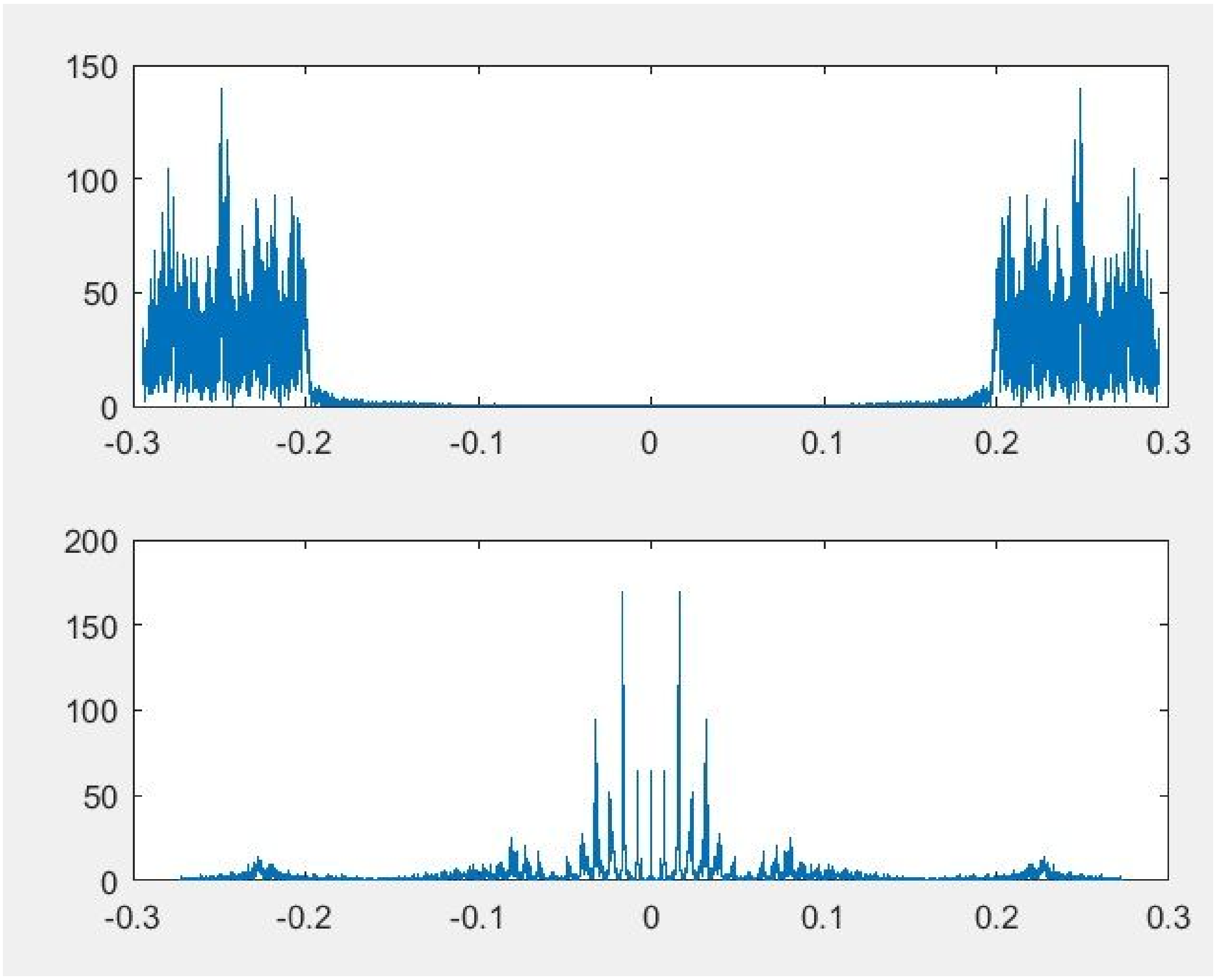
Part A)

We successfully multiplied our signal with (-1)^n and fft it to frequency domain



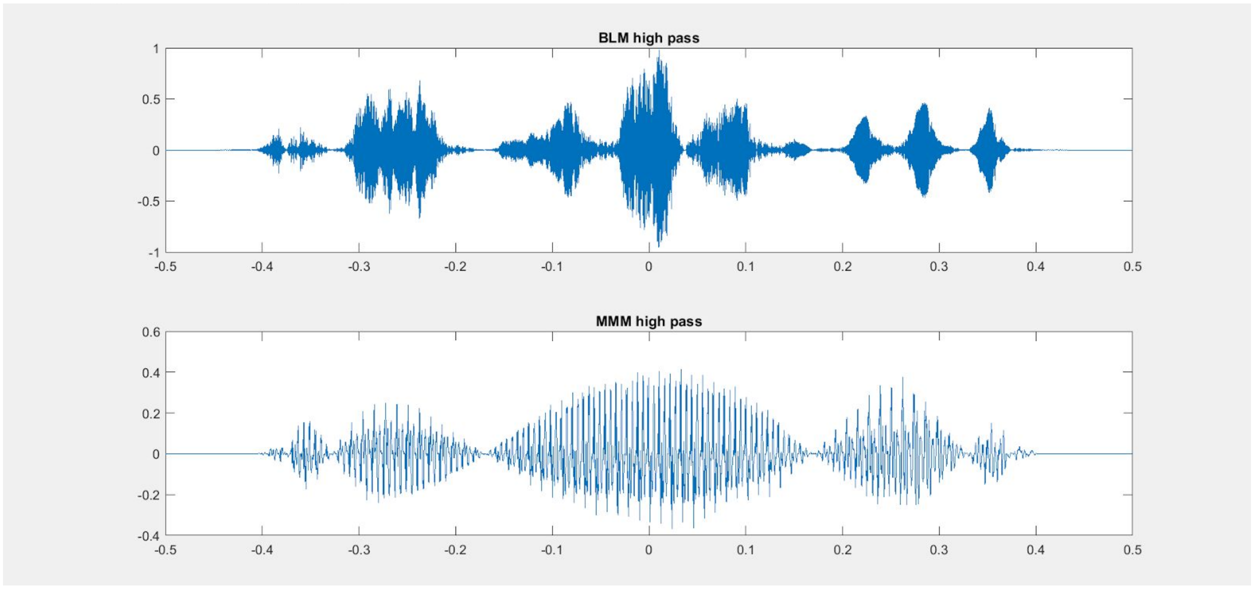
Part B)

We successfully sampling our signal with downsample() command. The main difference between shifting and scaling is that in scaling, the signal is either minimized or maximized where shifting only shift the signal to left or right. The first figure is for blm.wav and the second one is for mmm.wav.



Part C)

We applied the filter using butter() command with cutoff frequency of 0.25 and filter() command to filter our signals. The result is exactly as we expected. The final result really zero out frequency terms and the filtered result is different than the original one. The filtered sound somewhat less than the original because a lot of energy are eliminated in the filtering process.


BLM high pass
1
0.5
0
-0.5
-1
-0.5
-0.4
-0.3
-0.2
-0.1
0
0.1
0.2
0.3
0.4
0.5
MMM high pass
0.6
0.4
0.2
0
-0.2
-0.4
-0.5
-0.4
-0.3
-0.2
-0.1
0
0.1
0.2
0.3
0.4
0.5