ESE 482

Extra Credit

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Professor Sutton

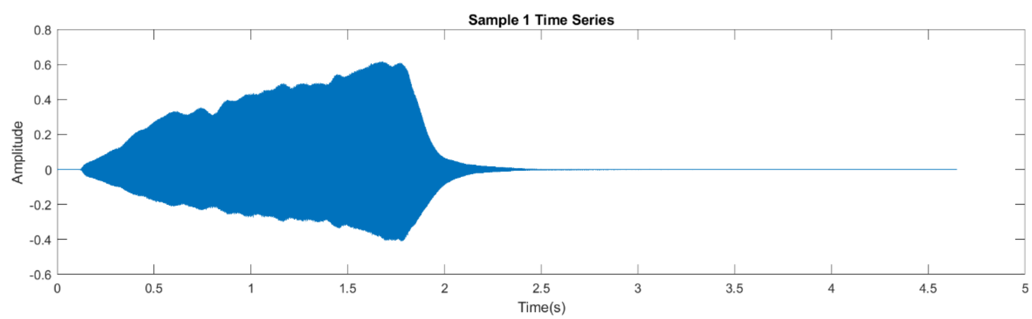
1. Using the code:

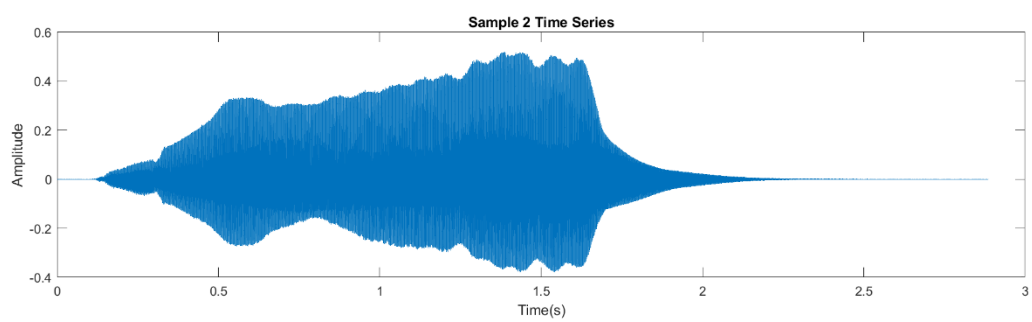
```
[y_1, Fs_1] = audioread("sample_1.aif");
[y_2, Fs_2] = audioread("sample_2.aif");
```

Sampling frequency is 44100 in both files. With a resolution of 0.5 Hz, multiply sampling frequency by 2, to result in:

$M = 88200.$

2. The time domain representation of a single channel in each sample is plotted:



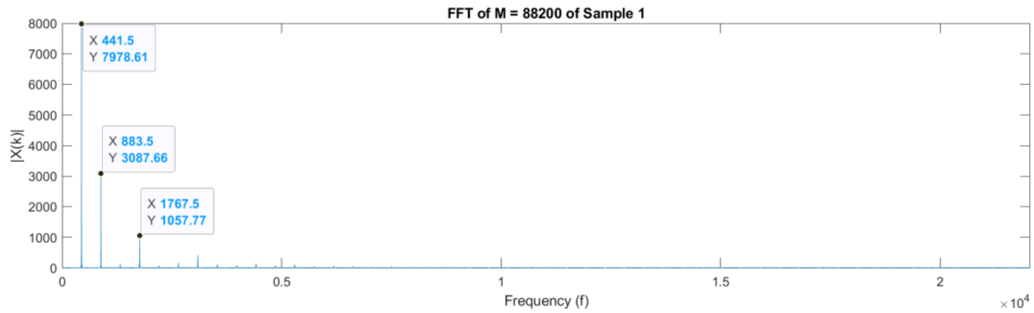


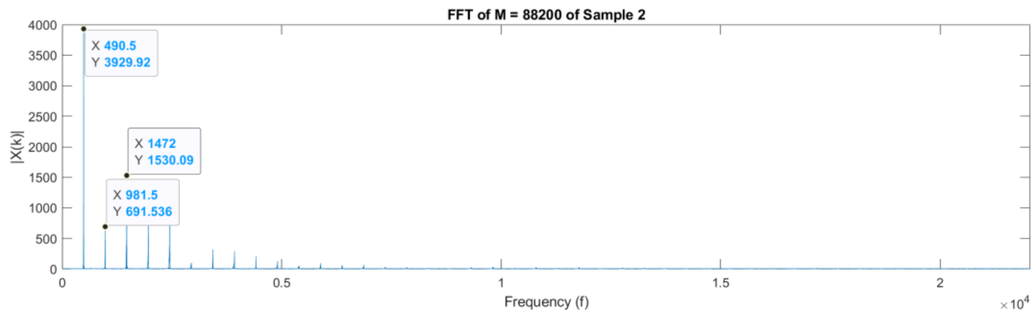
Sections were chosen using the code:

```
section_1 = y_1(5000:M+5000, :);
section_2 = y_2(5000:M+5000, :);
```

This cut off the initial silence and kept the bulk of the signal with the needed signal length.

3. Using the fft function for the sections chosen in part 2, the spectrum for a single channel was plotted (from a range of $0 - \frac{F_s}{2}$).





4. Since we know that the frequency resolution is 0.5 Hz, which means that there is 2 bins for each integer increment of the frequency, multiplying the frequency by 2 provides the value k for the first 3 peaks.

For sample 1:

$k = 883, f = 441.5$

$k = 1767, f = 883.5$

$k = 3535, f = 1767.5$

For sample 2:

$k = 981, f = 490.5$

$k = 1963, f = 981.5$

$k = 2944, f = 1472$

Note: The 4^{th} peak of sample 2's spectrum was on $f = 1963$.

5. For sample 1: A || For sample 2: B

These notes are not perfectly in tune. If they were sample 1 would be 440 Hz, and sample 2 would be 493.9 Hz.