Tune-Up Tuesday for October 5, 2021

% Create a MATLAB simulation for fall 2021 midterm problem 1.2
% The problem involves analysis of the output of a squaring system $y(t) = x^2(t)$.
% Let input $x(t) = \cos(2\pi f_1 t) + \cos(2\pi f_2 t)$ where $f_1 = 110$ Hz and $f_2 = 220$ Hz.

% (a) Using a sampling rate of 8000 Hz and time from 0 to 3 seconds, write
% the code to generate a sampled version of the signal $x(t)$.

```
fs = 8000;
Ts = 1/ fs;
tmax = 3;
t = 0 : Ts : tmax;
f1 = 110;
f2 = 220;
x = cos(2*pi*f1*t) + cos(2*pi*f2*t);
```

% (b) Play sampled version of $x(t)$. Describe what you hear.
% Answer: The signal x(t) is composed of A note in the second octave (110 Hz)
% and A note in the third octave (220 Hz).
*% **On a laptop**. On many laptop speakers, the 110 Hz tone may not be audible*
% due to limitations in playing back low audible frequencies. On a laptop, the
% playback sounded like a single note with hum in the background.
*% **Audio system with a sub-woofer**. A sub-woofer plays low audible frequencies*
% down to 20 Hz. The sub-woofer is often a separate large speaker (due to the
% longer acoustic wavelengths λ for low frequencies, i.e. $\lambda = c / f$) in an audio
% system. On an audio system with a sub-woofer, the playback sounded like a
% beat frequency with hum in the background. Both notes were audible.

```
soundsc(x, fs);
pause(tmax+1);
```

% (c) Plot the spectrum of the sampled version of $x(t)$. *Principal frequencies*
% are 110 Hz and 220 Hz.
% **Using the spectrogram**. *See the second page for the plot.*

```
figure;
spectrogram(x, 512, 256, 512, fs, 'yaxis');
```

% **Using the fast Fourier transform approach from mini-project #1**.

```
fourierSeriesCoeffs = fft(x);
N = length(x);
freqResolution = fs / N;
ff = (-fs/2) : freqResolution : (fs/2)-freqResolution;
figure;
plot(ff, abs(fftshift(fourierSeriesCoeffs)));
xlabel('f');
xlim( [-1000, 1000] );
ylim( [-10, 15000] );
```

% (d) Using a sampling rate of 8000 Hz and time from 0 to 3 seconds,
% write the code to generate the sampled version of the signal $y(t)$.

```
y = x.^2;
```

% (e) Play sampled version of $y(t)$. Describe what you hear.
% Answer: The signal y(t) is composed of 'A' note in the second octave (110 Hz)

% *‘A’ note in the third octave (220 Hz), ‘E’ note in the fourth octave (330 Hz),*
% *and ‘A’ note in the fourth octave (440 Hz). When played back, the signal*
% *y(t) has a higher pitch than x(t) but it was difficult to distinguish more than*
% *two notes. The 0 Hz term is not audible. Please see the answer in part (b).*

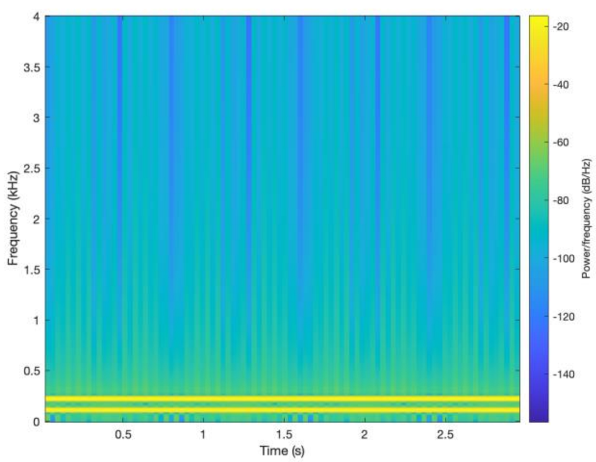
```
soundsc(y, fs);
pause(tmax+1);
```

% (f) Plot the spectrum of the sampled version of x(t). *Principal frequencies*
% *are 0, 110, 220, 330, and 440 Hz. Principal frequencies at 0, 330, 440 Hz aren’t*
% *in x(t) and are called intermodulation distortion caused by the squaring system.*
% **Using the spectrogram**. *See below for the plot*.

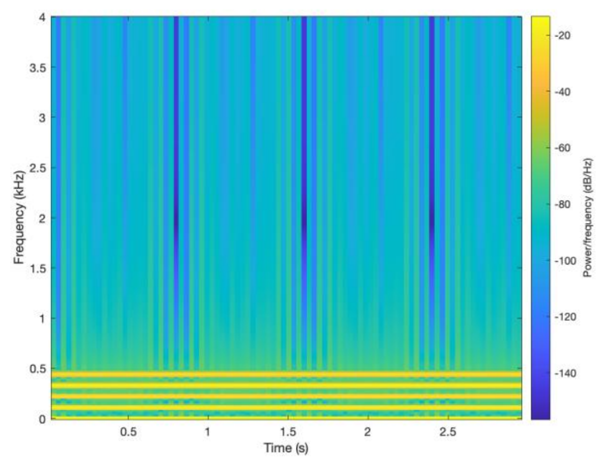
```
figure;
spectrogram(y, 512, 256, 512, fs, 'yaxis');
```

% **Using the fast Fourier transform approach from mini-project #1**. *See below*.

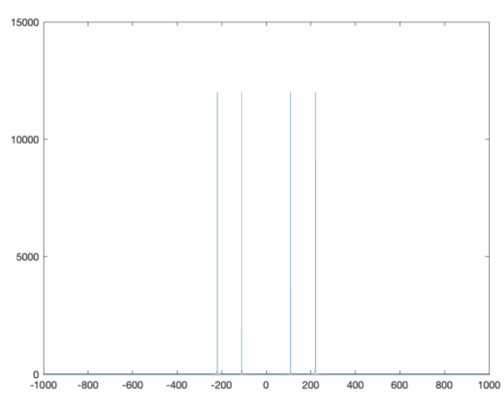
```
fourierSeriesCoeffs = fft(y);
N = length(y);
freqResolution = fs / N;
ff = (-fs/2) : freqResolution : (fs/2)-freqResolution;
figure;
plot(ff, abs(fftshift(fourierSeriesCoeffs)));
xlabel('f');
xlim( [-1000, 1000] );
```



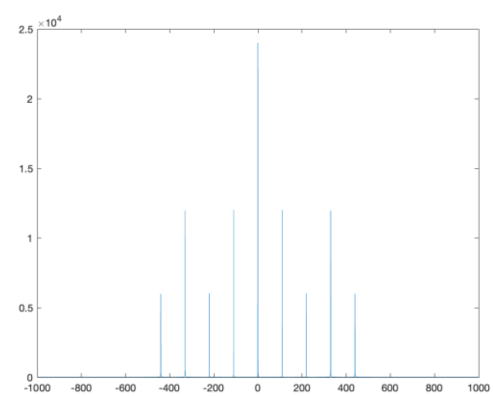
Spectrogram for sampled $x(t)$



Spectrogram for sampled $y(t)$



Spectrum for sampled $x(t)$



Spectrum for sampled $y(t)$