ECE 220-203

Lab 5

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Objective

The objective of this lab is to increase our knowledge of CTFT and how to use it in MATLAB.

MATLAB Code

See Appendix A.

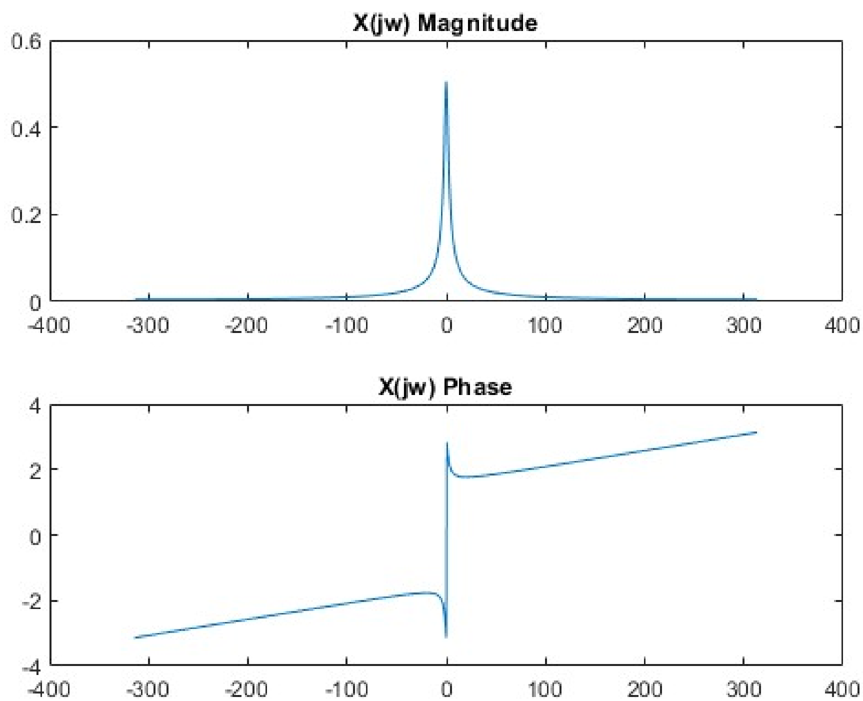
Results

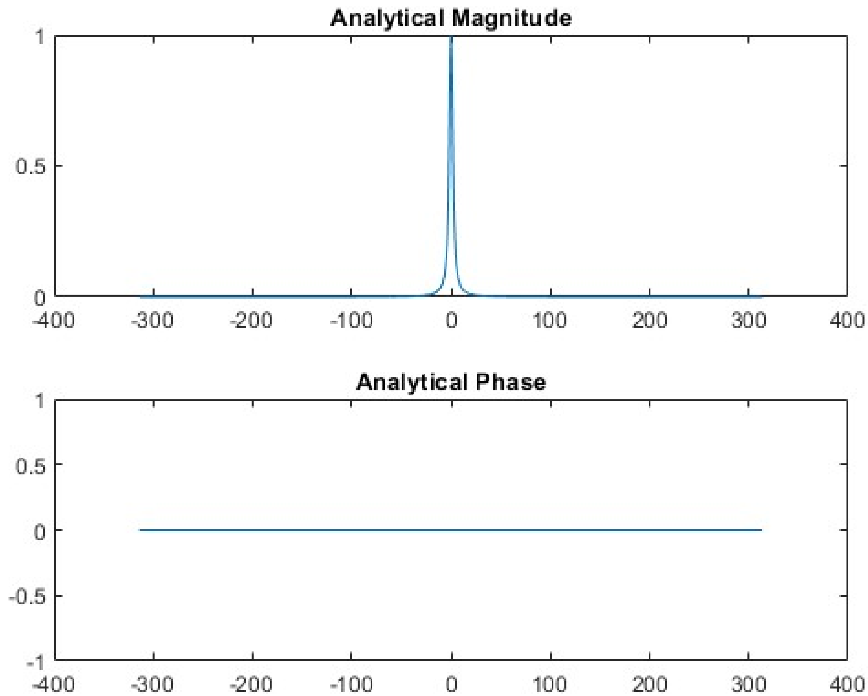
Part I1:

Taken from textbook's list of Fourier transforms;

$$e^{-2|t|} \overset{FT}{\Rightarrow} \frac{4}{4+\omega^2}$$

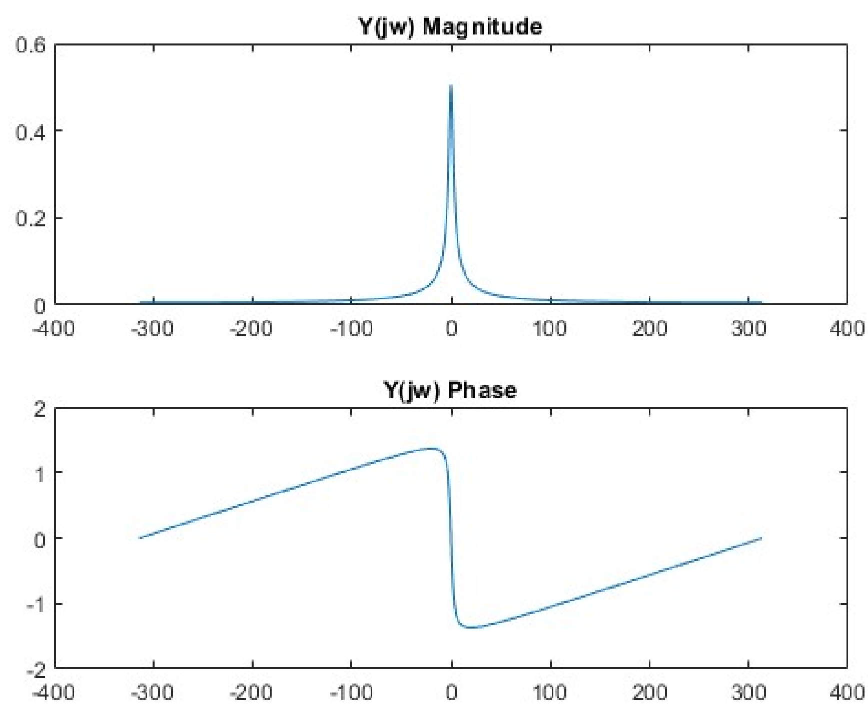
Part I6:





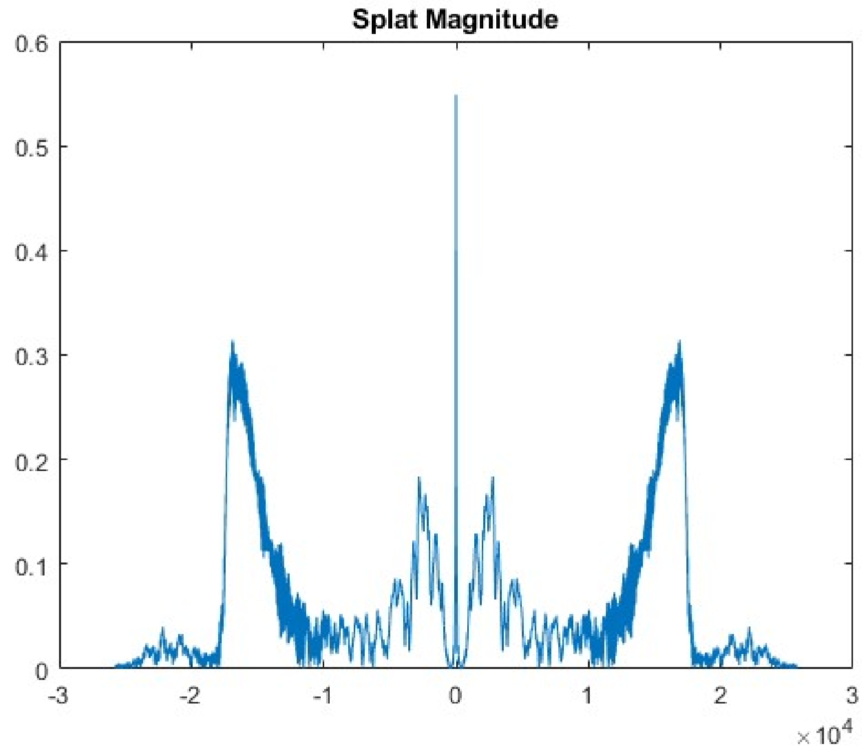
My approximation of the CTFT matches the amplitude, but not the phase.

Part I7:



The magnitude is the same as X(jω), but the phase is reversed. I could have predicted this result because the function was just time shifted.

Part II1:



Part II2:

The signal $y1$ sounded like the original signal, y, but it was reversed.

Part II3:

For $y_2(t)$, the magnitude of a real input, $y(t)$, will always be real. Therefore, $y_2(t)$ will be real.

Euler's formula says $e^{j\pi} = -1$, and $y_3(t) = e^{j\angle y(t)}$. The angle of $y(t)$, $\angle y(t)$, will be an ω value, which in of itself is a multiple of π. Therefore, $y_3(t)$ will be real

Part II6:

The magnitude is more crucial in representing an audio signal since the signal with magnitude, while distorted, gave more information about the signal than the signal with just the amplitude, which just played a single note.

Part III1:

System I

$$y' + 3y = 3x$$

$$H(s) = \frac{Y(s)}{X(s)} = \frac{3}{s+3}$$

$$|H(j\omega)| = \frac{\omega}{\sqrt{\omega^2 + 9}}$$

$$\angle H(j\omega) = -\tan^{-1}\left(\frac{\omega}{3}\right)$$

System 2

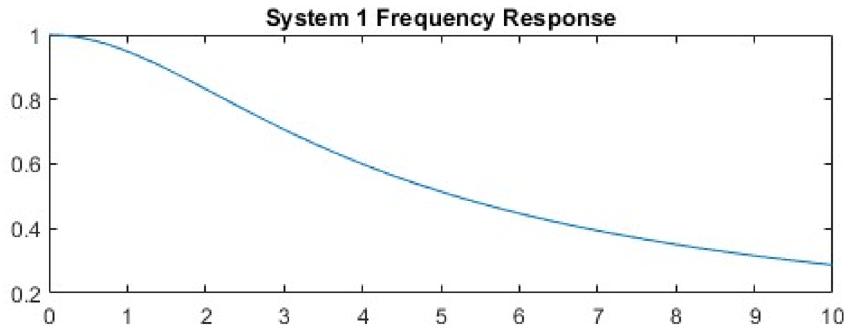
$$y' + \tfrac{1}{3}y = \tfrac{1}{3}$$

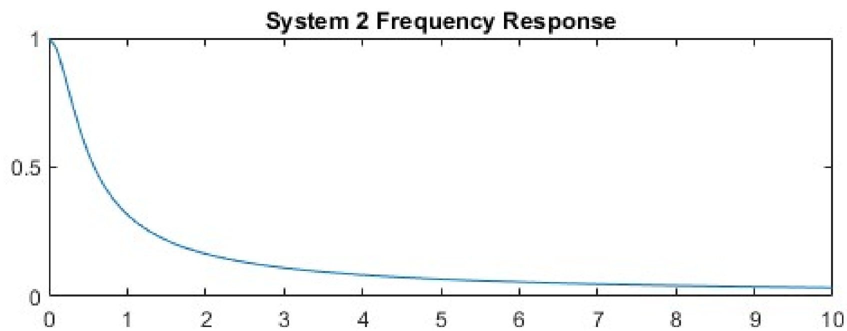
$$H(s) = \frac{Y(s)}{X(s)} = \frac{1/3}{s+1/3}$$

$$|H(j\omega)| = \frac{\omega}{\sqrt{\omega^2 + 1/9}}$$

$$\angle H(j\omega) = -\tan^{-1}\left(\frac{\omega}{1/3}\right)$$

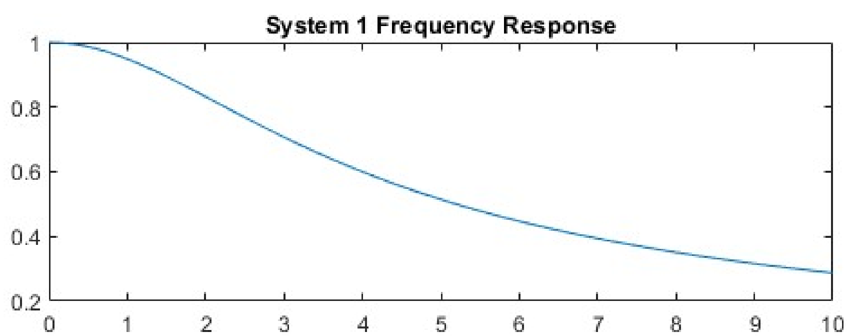
Part III2:

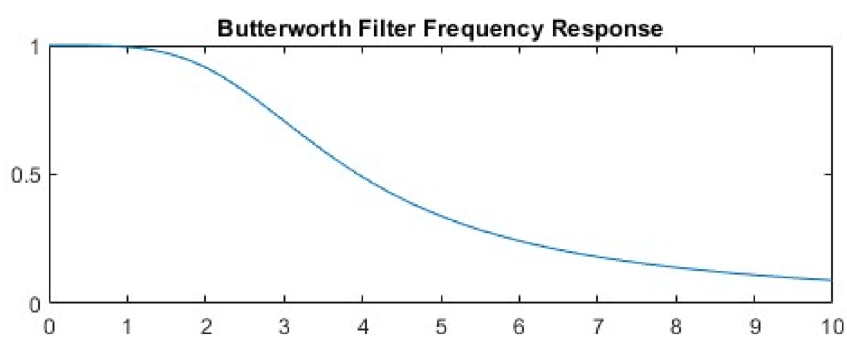




The magnitudes agree with my analytical expressions.

Part III3:





Part III4:

The second-order Butterworth filter more closely approximates an ideal lowpass filter was a $\omega_c = 3$. The phase of the Butterworth filter is much steeper with a slope of about -1, while System 1's phase is much shallower at around $-\frac{1}{2}$. An ideal lowpass filter's phase is a closer to -1, so the Butterworth filter is again a better representation of a lowpass filter.

Conclusion

There were many issues encountered while completing this lab, all of them in Part I. I'm positive that my analytical work of finding the FT of $e^{-2|t|}$ was correct, since it came directly from the book, but I didn't know how to get the phase of it since there was no way the equation would output imaginary numbers. Also, I couldn't figure out how to time shift $Y(j\omega)$ by 5, so $Y(j\omega)$ is the unshifted signal and X is a different signal from what I think I should be getting. I tried many different methods to work out this problem, but I couldn't find any real solution. I had no issues in Part II and it was interesting to see how the inverse Fourier transform could be used on signals to extract their magnitude and phases'. Part III was also straightforward since most of the code was given and the math was something we've already did in lecture. Many of my classes are all focusing on filters right now coincidentally, so extra practice on understanding them is always welcome. Besides Part I, I believe my results to be accurate and found no errors when executing my MATLAB script.

Appendix A

```
%%
%Part 1
figure(1)
tau = 0.01;
T = 10;
t = 0:tau:T-tau;
N = T/tau;
y = length(N);
for n = 1:N
    y(n) = exp(-2*abs(t(n)));
end
Y = fftshift(tau*fft(y));

w = -(pi/tau)+(0:N-1)*(2*pi/(N*tau));

X = Y .* exp(-1i*w*5);

subplot(211), plot(w, abs(X)), title('X(jw) Magnitude');
subplot(212), plot(w, angle(X)), title('X(jw) Phase');

figure(2)
Z = length(N);
for n = 1:N
    Z(n) = 4/(4+w(n)^2);
end
subplot(211), plot(w, abs(Z)), title('Analytical Magnitude');
subplot(212), plot(w, angle(Z)), title('Analytical Phase');

figure(3)
subplot(211), plot(w, abs(Y)), title('Y(jw) Magnitude');
subplot(212), plot(w, angle(Y)), title('Y(jw) Phase');
%%
%Part 2
figure(4)
tau = 0.01;
load splat;
y = y(1:8192);

N = 8192;
fs = 8192;
%soundsc(y, fs);

w = (-pi:2*pi/N:pi-pi/N)*fs;
Y = fftshift(tau*fft(y));
plot(w, abs(Y)), title('Splat Magnitude');

y = ifft(fftshift(Y));
y = real(y);
%soundsc(y, fs);

y1 = conj(Y);
y1 = ifft(fftshift(y1));
y1 = real(y1);
%soundsc(y1, fs);
```

```
y2 = abs(Y);
y2 = ifft(fftshift(y2));
%soundsc(y2, fs);

y3 = angle(Y);
y3 = ifft(fftshift(y3));
y3 = real(y3);
%soundsc(y3, fs);
%%
%Part 3
figure(5)
w = linspace(0,10);

a = [1 3];
b = [3];
s1 = freqs(b, a, w);

a = [1 1/3];
b = [1/3];
s2 = freqs(b, a, w);

subplot(211), plot(w, abs(s1)), title('System 1 Frequency Response');
subplot(212), plot(w, abs(s2)), title('System 2 Frequency Response');

figure(6)
wc = 3;
[b2, a2] = butter(2,wc,'s');

s3 = freqs(b2, a2, w);
subplot(211), plot(w, abs(s1)), title('System 1 Frequency Response');
subplot(212), plot(w, abs(s3)), title('Butterworth Filter Frequency
Response');
```