Signals and Systems Computations using Matlab

ECE 146 Lab 1

October 14, 2019

Student: Ivan Arevalo
Perm Number: 5613567
Email: ifa@ucsb.com

Department of Electrical and Computer Engineering, UCSB

1 Introduction

This lab is designed to become familiar with MATLAB and practice using functions and syntax necessary reinforce signal and system concepts. Furthermore, we will get experience developing functions to approximate continuous time operations using a discrete time framework.

2 Exercises

2.1 Exercise 1: Functions and Plots

2.1.1 Part (a)

```
function [signalx_out_vector] = signalx(input_time_vector)
% Problem 1a
%   Write a Matlab function signalx that evaluates the following signal at an arbitrary
    set of
%   points
signal_time = input_time_vector;
length_signal = length(signal_time);
signalx_out_vector = zeros(1, length_signal);
index_m3_m1 = (signal_time >= -3) & (signal_time <= -1);
index_m1_p4 = (signal_time >= -1) & (signal_time <= 4);
signalx_out_vector(index_m3_m1) = 2 * exp(signal_time(index_m3_m1) +2);
signalx_out_vector(index_m1_p4) = 2 *
    exp(-1*signal_time(index_m1_p4)).*cos(2*pi*signal_time(index_m1_p4));
end
```

2.1.2 Part (b)

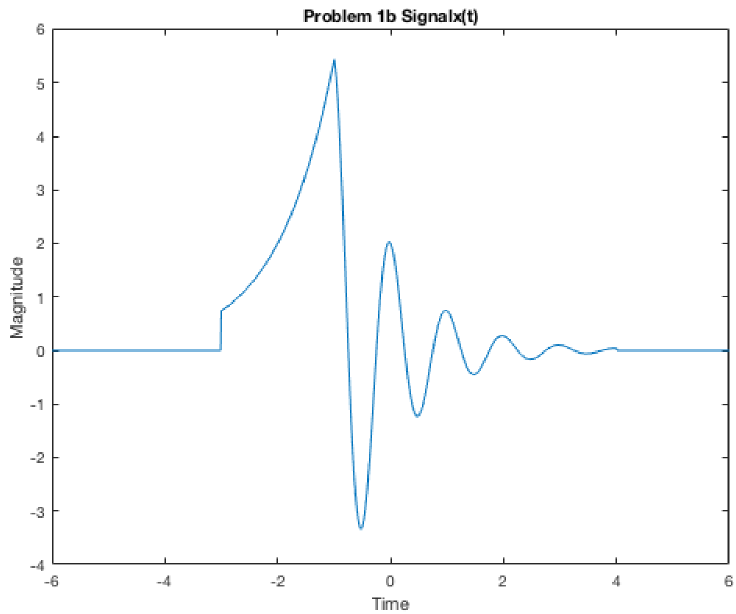

Figure 1: Signalx(t)

2.1.3 Part (c)

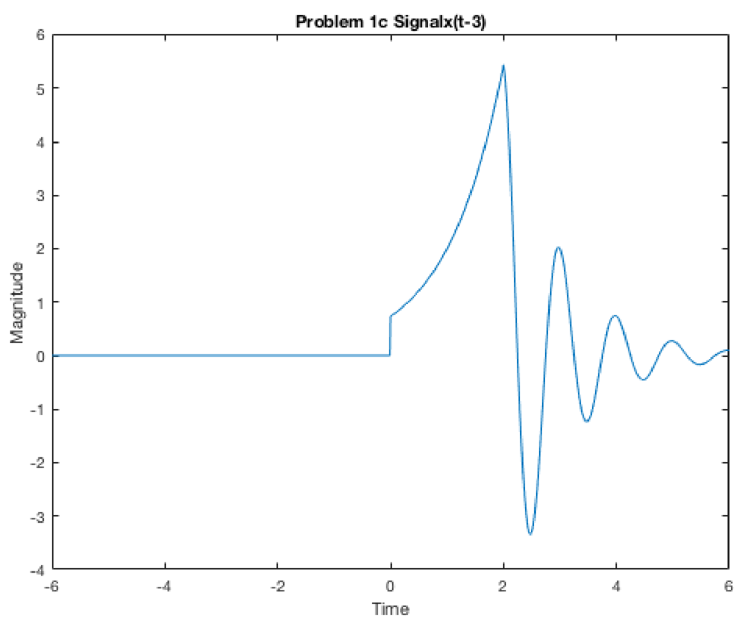

Figure 2: Signalx(t-3)

2.1.4 Part (d)

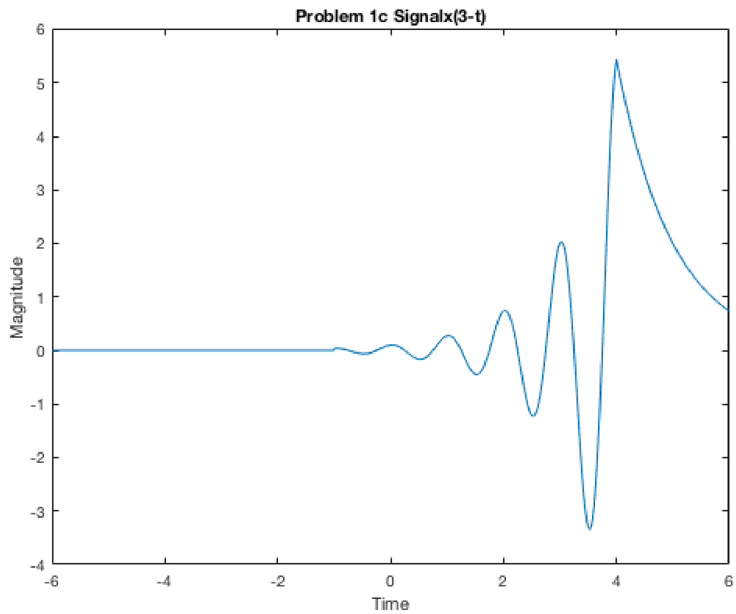

Figure 3: Signalx(3-t)

2.1.5 Part (e)

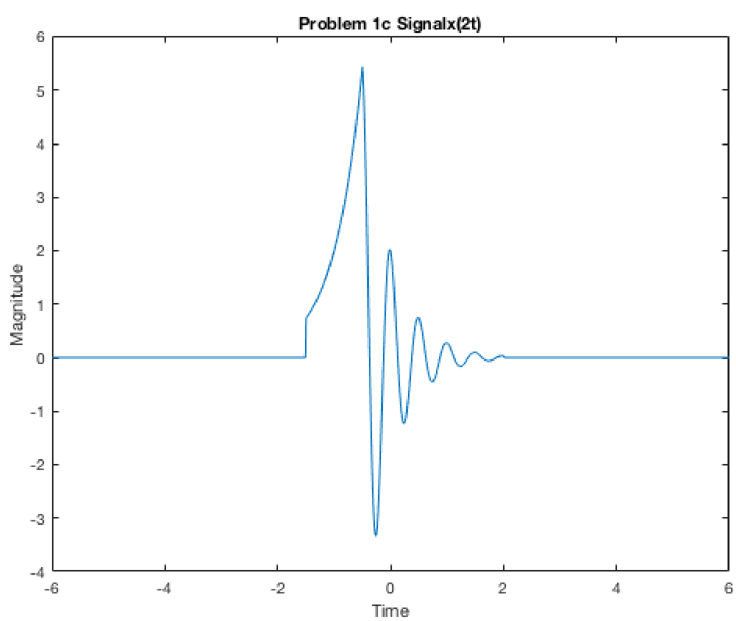

Figure 4: Signalx(2t)

2.2 Exercise 2: Convolution

2.2.1 Part (a)

```
function [convolved_signal,convolved_signal_time] = contconv2(signal_1, signal_2,
    time_1, time_2, dt)
%Problem 2a
%   Write a Matlab function contconv that computes an approximation to continuous-time
%   convolution.

% Find the convolved signal time vector
number_of_points = length(signal_1) + length(signal_2) -1;
start_conv_sig_time = time_1 + time_2;
% convolved_signal_time = zeros(1, number_of_points);
convolved_signal_time = start_conv_sig_time:dt:start_conv_sig_time + ((number_of_points
    -1) * dt);

% Make a matrix with size length(signal_1) * number_of_points)
% and increasingly add row vector signal 2 starting in the diagonal of the matrix.
padded_matrix = zeros(length(signal_1), number_of_points);

for i=1:length(signal_1)
    padded_matrix(i, i:i + length(signal_2) - 1) = signal_2';
end
convolved_signal = dt*(signal_1' * padded_matrix)';
end
```

2.2.2 Part (b)

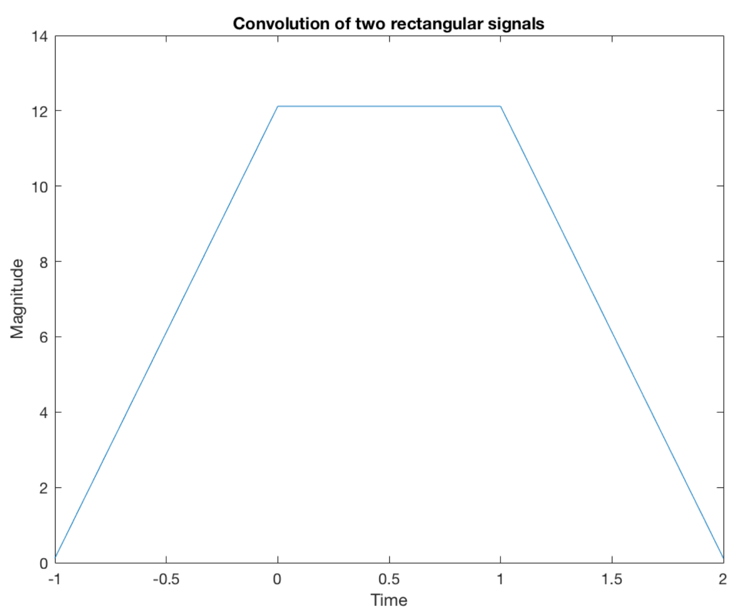

Figure 5: Testing convolution function on two rectangular signals

Hand calculations agree with graph in both time and scale.

2.3 Exercise 3: Matched Filter

2.3.1 Part (a)

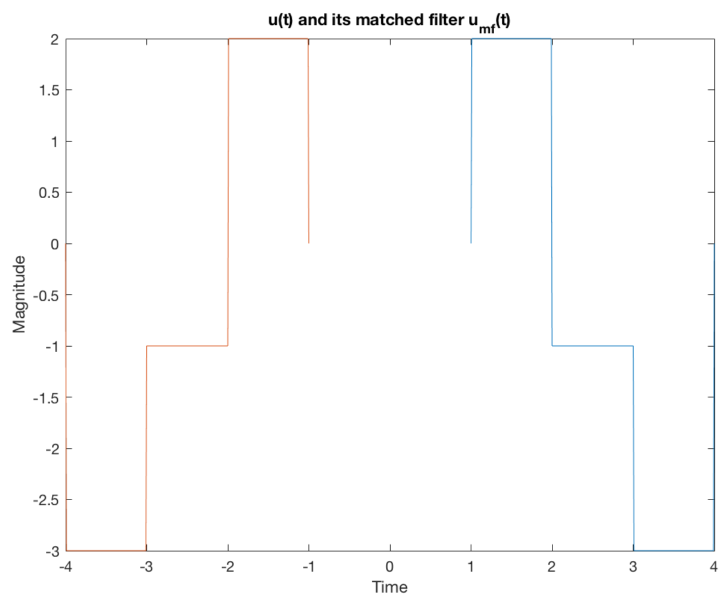

Figure 6: u(t) and its matched filter u$_{MF}$(t)

2.3.2 Part (b)

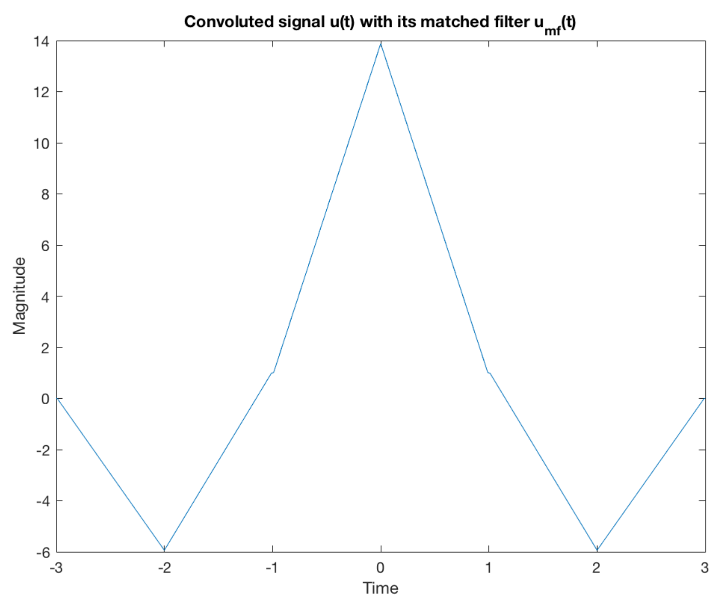

Figure 7: u(t) * u$_{MF}$(t)

The peak of the signal is located at time = 0 seconds. This is because the maximum overlap between a signal and its matched filter occurs at time = 0.

2.3.3 Part (c)

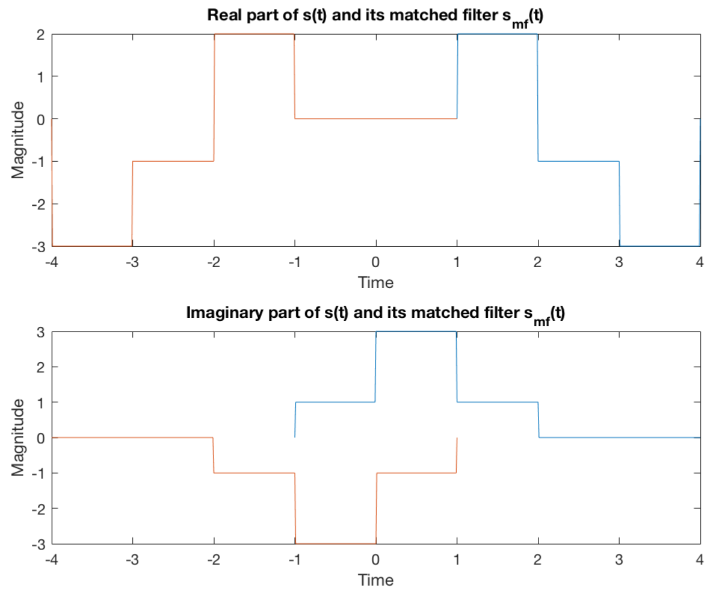

Figure 8: Real and Imaginary parts of s(t) and s_{MF}(t)

2.3.4 Part (d)

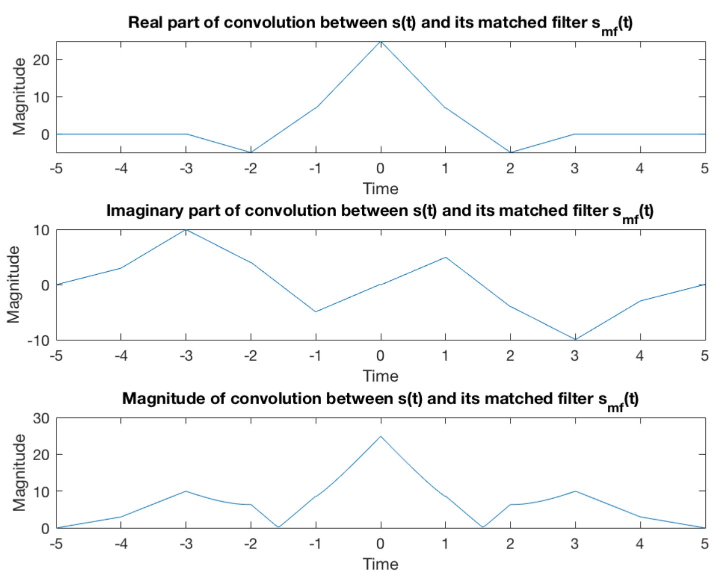

Figure 9: Real, Imaginary, and Magnitude plots of s(t) * s_{MF}(t)

There is a peak present in both the real and magnitude plots at t = 0 seconds. In the imaginary plot there is positive / negative peak respectively at t = -3 seconds and t = 3 seconds.

2.3.5 Part (e)

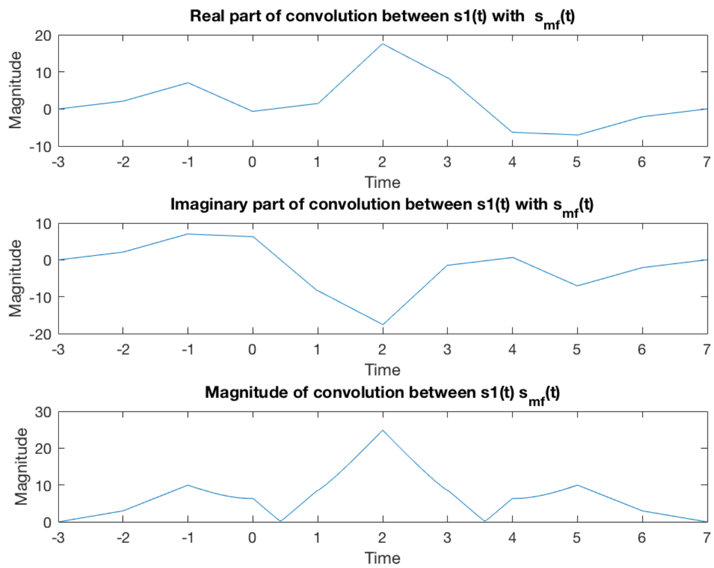

Figure 10: Convolution of $s_1(t)=s(t-t_0)e^{j\theta}$ $s_{MF}(t)$, for $t_0=2$ and $\theta=\pi/4$

There is a peak present at t = 2 seconds in all graphs.

2.3.6 Part (f)

Based on the fact that the peak in the real part of the convolution is at 2 seconds, and understanding that the peak occurs at the point of maximum overlap, we can estimate that s(t) was time shifted by two seconds. Furthermore, since the matched filter of s(t) is defined as $s^*(-t)$, the convolution at the point of maximum overlap is equal to $\int s(t)s*(t)\,\mathrm{d}x$. which is a purely real value. Given that Figure 10 has a value of 20 in the real part and -20 in the imaginary part, we can conclude that the angle by which s(t) is being modulated $\pi/4$.

2.4 Exercise 4: Fourier Transform

2.4.1 Part (a) and (b)

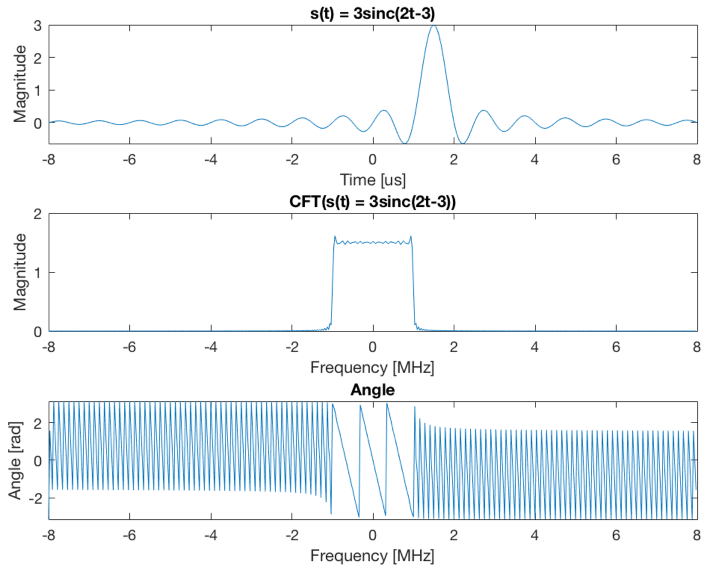

Figure 11: Magnitude and Phase of Fourier Transform of $s(t) = 3sinc(2t - 3)$

Figure 11 shows a plot of $s(t) = 3sinc(2t-3)$, and magnitude and phase of its Fourier Transform. These graphs match the expected shapes for each. s(t) was shifted by 3 and compressed by two which is accurately shown in the top graph. The expected Fourier Transform of a *sinc* function is a rectangular wave which is well approximated in the the middle graph. Finally, the phase portrayed in the bottom graph is only significant within the frequencies which have a magnitude that doesn't approach 0. If we think about a phasor in the complex plane, as the magnitude of the phasor approaches zero, the angle of the phasor is unstable. Therefore the only valuable data in this graph is within the frequencies -1 KHz and 1KHz.

2.5 Exercise 5: Matched Filter in Frequency Domain

2.5.1 Part (a)

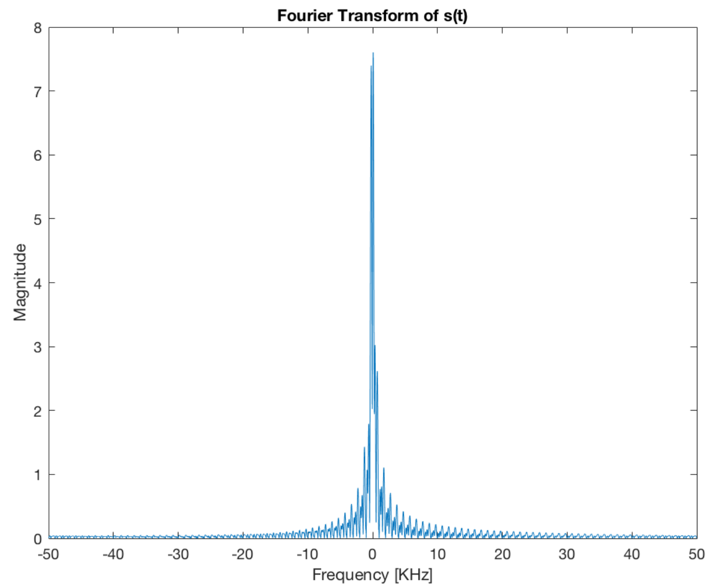

Figure 12: Magnitude of Fourier Transform of $s(t)$ from Figure 8

2.5.2 Part (b)

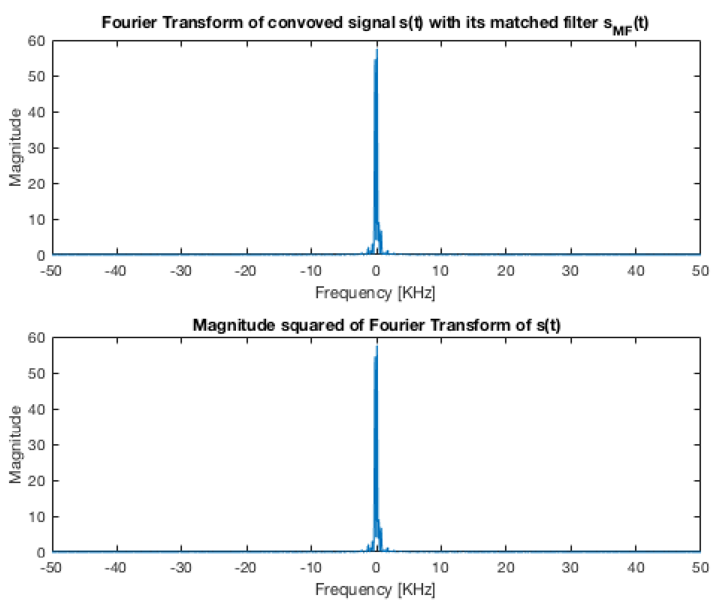

Figure 13: Magnitude of Fourier Transform $s(t) * s^{*}(t)$ from Figure 9 compared to Magnitude of Fourier Transform of $|s(t)|^2$

2.5.3 Part (c)

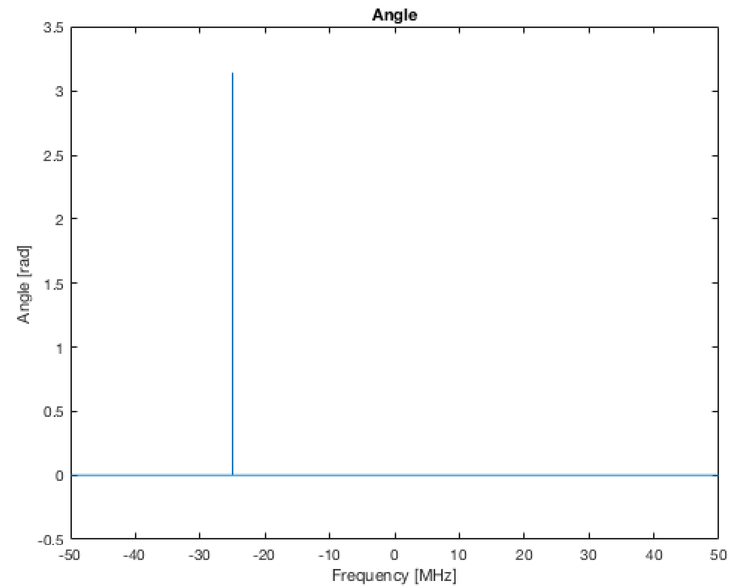

Figure 14: Phase of Fourier Transform of $s(t) * s^*(t)$ from top of Figure 13

Given that the convolution of a signal with its corresponding matched filter is purely real, the phase of the Fourier transform is 0. There is also a spike at -25 KHz which goes to *pi* which also corresponds to a purely real component.

3 Conclusion

In conclusion, this Lab provided exercises which allowed us to practice emulating various continuous time operations in MATLAB. This will become and essential tool to develop in order to compute, visualize, and apply the concepts learned in communication theory. Furthermore, we got some intuition and understanding regarding *sinc* functions and matched filters, more specifically, we demonstrated that the Fourier Transform of the *sinc* function is a rectangular wave and that the Fourier Transform of convolving a signal with its matched filter is equal to squaring the magnitude of the Fourier Transform of the original signal.