ECE 591: Software Defined Radio

Lab 1: Amplitude Modulation

We certify that this work is original and not a product of anyone's work but our own.

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Abstract

Becoming familiar with software is crucial to the development of a student's ability to understanding the system they will be working on. This lab would be the first in which students were exposed to the Simulink add-on within MATLAB as well as the RTL-SDR and Pluto SDR and would act as a precursor to the next labs for the remainder of the semester. During the course of the laboratory, students will learn how to use MATLAB and Simulink and will apply their knowledge of Modulation techniques to implement a DSB-TC transmitter and receiver. Students will utilize both the RTL-SDR Dongle as the receiver, as well as the Pluto SDR as the transmitter.

1 Introduction

This laboratory consisted of a four part in which, MATLAB, Simulink, an RTL-SDR, and a Pluto SDR were used. For this lab, students were first tasked with understanding given MATLAB code that generated an AM DSB signal, as well as its corresponding FFT and time domain graph, from a sinusoidal carrier and message signal. Next, students were asked to explore the effects of increasing amplitude sensitivity in an AM-DSB design using both MATLAB as well as Simulink to simulate the transmitted signals. For the third part, an RTL-SDR receiver was used in conjunction with Simulink's ”Spectrum Analysis with SDR Radio” example to examine and measure the signal characteristics. Lastly, both the Pluto SDR transmitter and the RTL-SDR receiver were to be utilized to send and receive respectively an AM DSB signal. Both the transmitter as well as the receiver were to be designed in Simulink.

2 Methods and Procedures

2.1 Part 1: Signal Representation in Time Domain and Frequency Domain

To fulfill the requirements for part 1 detailed in the lab handout which can be found in the ECE 591 mycourses page as well as in the References section of this report, the below code was to be analyzed and copied verbatim into a MATLAB script. Some questions were posed where the given code had to be reworked. In specific, two other sinusoids namely “sig2” and “sig3” were created that would contain the specified frequencies as shown below in **Figure 1**. With these two new signals a new sinusoid was created named ”merged” that would be the sum of all three signals each with differing frequencies. The last bit of change required from the original code provided was that when performing the fft function the argument was altered from “sig” to the new “merged” signal. Along with this, when plotting the time domain of the merged signal, the second argument of the “plot” function was changed to the “merged” signal.

```
%% Part 2.1: Signal Representation in Time Domain and Frequency Domain
clear; clc; close all;

fc = 50;
Ts = 0.001;
t = 0:Ts:1;
n = length(t);
fs = 1/Ts;
sig = cos(2*pi*fc.*t);
sig2= cos(2*pi*200.*t);
sig3= cos(2*pi*600.*t);
merged=sig+sig2+sig3;
Y=fft(merged);
Y=fftshift(Y);
fshift = (-n/2:n/2-1)*(fs/n);
powershift = abs(Y).^2/n;
figure(1);
plot(t, merged);
xlabel('Time');
ylabel('Amplitude');
title('Oscilloscope');
figure(2);
plot(fshift, powershift);
xlabel('Frequency');
ylabel('Power');
title('Spectrum Analyzer');
```

Figure 1: Part 1 MATLAB Code

2.2 Part 2: Display and Analyze DSB AM Modulated Wave

Part 2 of the laboratory consisted of using MATLAB scripts and Simulink in order to observe the effect the amplitude sensitivity factor has on an AM DSB-TC transmitter. The developed MATLAB code to build an AM transmitter is shown below in **Figure 2**. The basic structure of this code relies upon the provided code from Part 1 of this laboratory. To construct the AM DSB-TC modulator, line 56 of the code block shown below was added which is the general formula of an AM DSB-TC modulated wave which was obtained from the lab handout.

```
42  %% Part 2.2 Display and Anaylze DSB AM Modulate Wave
43  clear; clc; close all;
44  
45  % the modulator
46  fc = 0.4;
47  fm=0.05;
48  Ts = 0.1;
49  t = 0:Ts:200;
50  n = length(t);
51  fs = 1/Ts;
52  message= cos(2*pi*fm.*t);
53  Ac=1;
54  carrier=cos(2*pi*fc.*t);
55  ka=10; %amplitude sensitivity
56  messageAM=Ac*(1+ka.*message).*carrier;
57  Y=fft(messageAM);
58  Y=fftshift(Y);
59  fshift = (-n/2:n/2-1)*(fs/n);
60  powershift = abs(Y).^2/n;
61  figure(1);
62  plot(t, messageAM);
63  xlabel('Time');
64  ylabel('Amplitude');
65  title('Oscilloscope');
66  figure(2);
67  plot(fshift, powershift);
68  xlabel('Frequency');
69  ylabel('Power');
```

Figure 2: Part 2 MATLAB Code

To accomplish the same task in Simulink, the below block diagram shown in **Figure 3** was created. This follows exactly what an AM DSB-TC transmitter block diagram looks like. Note that the summer block used indicates that the carrier is always present in the transmitted signal, even though the message signal may not have important information in it.

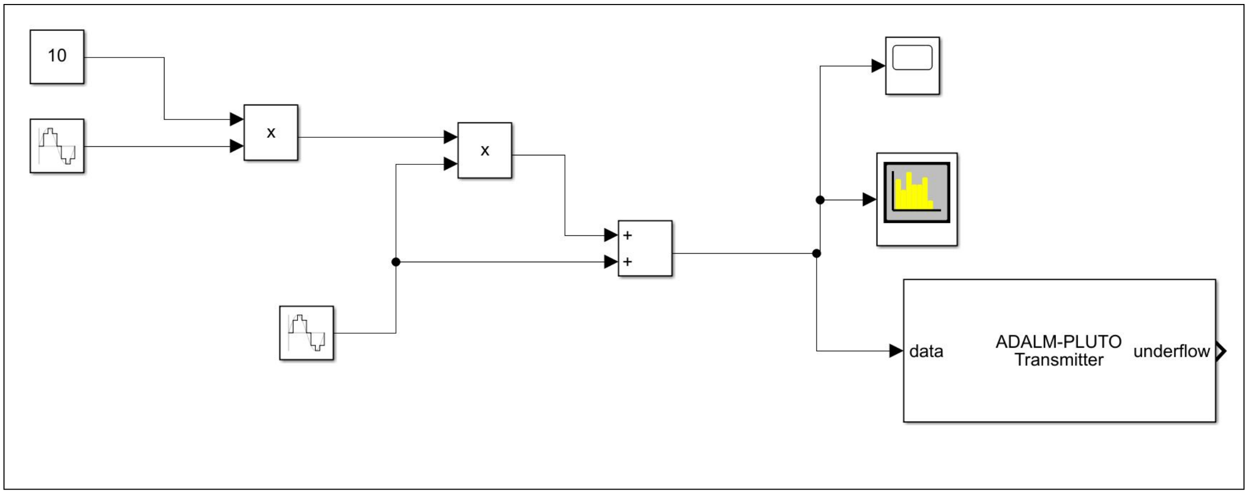

Figure 3: AM DSB-SC Simulink Block Diagram

2.3 Part 3: Spectrum Analysis with RTL-SDR Radio

Part 3 of the laboratory invovled running the Simulink "Spectrum Analysis with RTL-SDR Radio" example and measuring the various signal characteristics that it produces. The Simulink model used in this example can be seen below in **Figure 4**. The basic structure of this model is that it allows the user to define some center frequency that the RTL-SDR is looking for, and once received the signal is brought into the model where it decomposes its various spectral qualities.

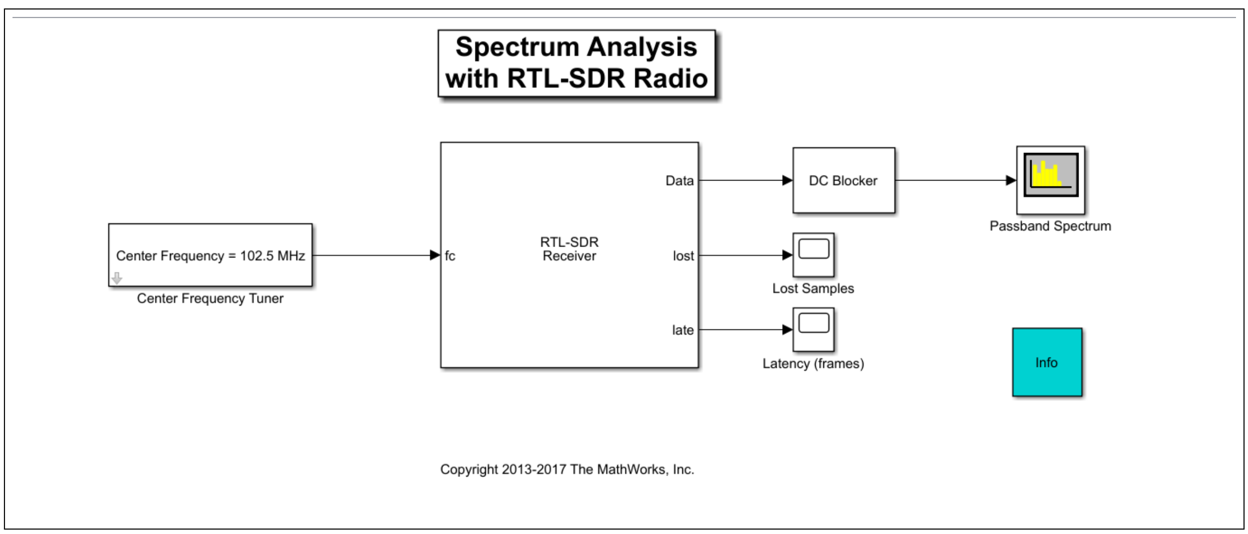

Figure 4: Spectrum Analysis Simulink Model

2.4 Part 4: Build Your Own AM Radio

Part 4 of the lab was dedicated to building an AM radio using Simulink/MATLAB. For this part, the group decided to build the radio in Simulink. The radio consisted of two different parts, the transmitter side and the receiver side. In order to transmit an AM signal, the Adalm Pluto Radio was used, while the RTL-SDR was used to receive and demodulate the AM signal.

2.4.1 AM Radio Transmission with Adalm Pluto Radio

In **Figure 5**, one can see the block diagram for the AM Radio Transmitter. The block diagram, from left to right, consists of:

1. Place the block of the signal that will be transmitted, which in this case is a .wav audio file.

2. Connect the signal to a gain block that has a value of 50.

3. Use an "FIR Rate Conversion" block to up-sample the waveform to 120kHz. The FIR block has a interpolation factor of 5, a decimation factor of 2. Its FIR filter coefficients are firpm(50, [0 15e3 24e3 240e3/2]/(240e3/2), [1 1 0 0], [1 1], 20).

4. Use another FIR Rate Conversion block to up-sample the waveform again up to 200kHz. The FIR block has a interpolation factor of 5, a decimation factor of 2. Its FIR filter coefficients are firpm(100, [0 15e3 30e3 600e3/2]/(600e3/2), [1 1 0 0], [1 1], 20).

5. Add 0.05 to the up-sampled waveform using an "ADD" block.

6. Use the "Real-Imag to Complex" block by connecting the waveform to the 'Re' input port. This is to convert the waveform data to complex.

7. Finally connect the output of the "Real-Imag to Complex" block to the ADALM-PLUTO Transmitter block.

8. Set the ADALM-PLUTO Transmitter's center frequency to the frequency one wishes to transmit, in this case it was 86.3e6

9. Set the Baseband sample rate to 240e3.

10. Lastly, have the transmitter's gain be zero and channel mapping be 1.

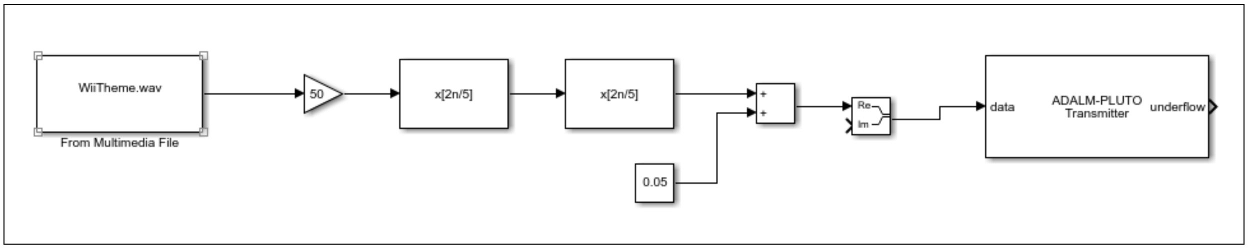

Figure 5: AM Radio Transmission Simulink Block Diagram

2.4.2 AM Radio Reception with RTL-SDR

In **Figure 6**, one can see the block diagram for the AM Radio Receiver. The block diagram, from left to right, consists of:

1. Place a constant block with a value of 86.3e6.
2. Place another constant block with a value of -40e3.
3. Add the constants together.
4. Place a third constant block with a value of 30.
5. Place the RTL-SDR receiver block. The block takes two inputs, the center frequency ’fc’ and a gain.
6. Connection the summation of the first two constants to the center frequency of the receiver block.
7. Connect the last constant block, with a value of 30 to the gain input port of the RTL-SDR receiver block.
8. The receiver block has one output port, ’data’. Connect the data to a bandpass filter block.
9. Figure 7 shows the bandpass filter block parameters used in this design.
10. Connect the output of the bandpass filter to Abs block, which will output the absolute value.
11. Now the received signal needs to be downsampled, this is done by using an FIR decimation block with a decimation factor of 5 and FIR coefficients ’firpm(100, [0 15e3 20e3 (240e3/2)]/(240e3/2), [1 1 0 0], [1 1], 20)’.
12. Finally, connect the output of the decimation block to a audio device writer block, which will play the received signal through the speakers of the computer.

13. Spectrum analyzers or oscilloscopes can be used before the bandpass filter to view the received signal prior to demodulating it and after the down-sample to see the demodulated signal.

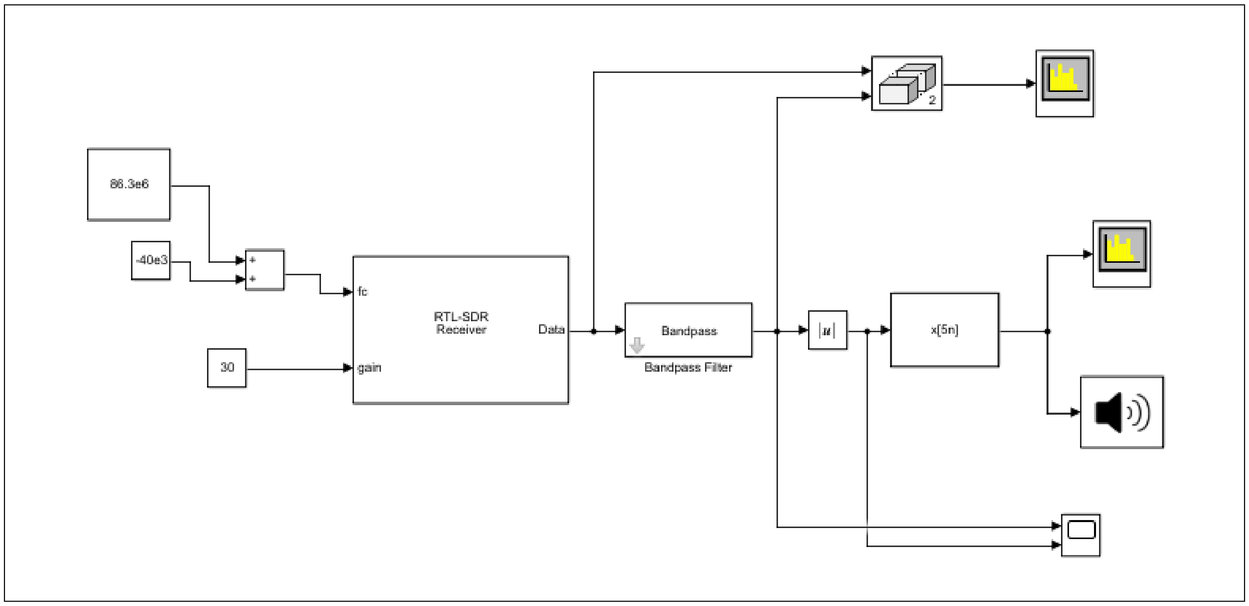

Figure 6: AM Radio Reception Simulink Block Diagram

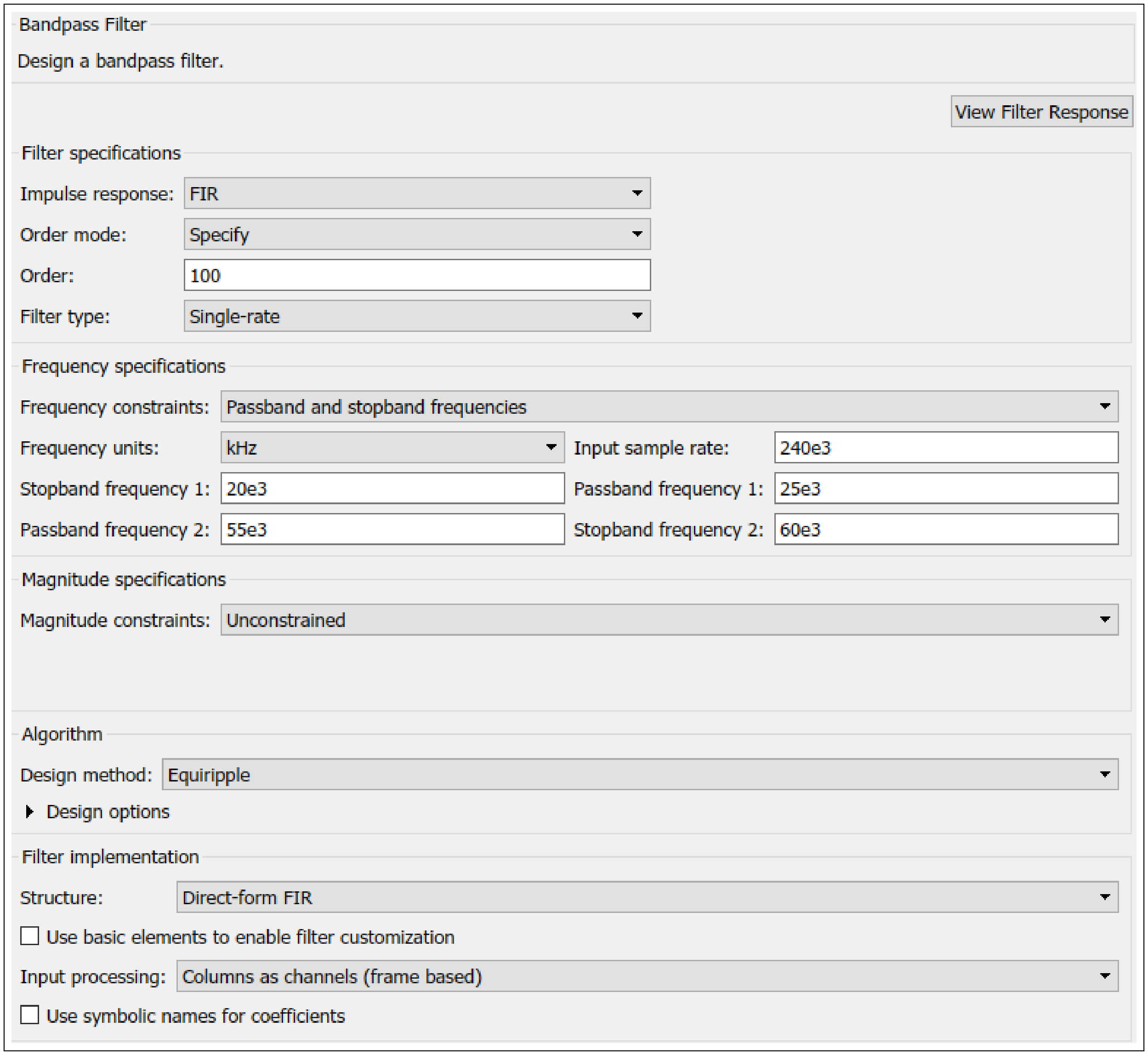

Figure 7: Bandpass Filter Parameters

3 Laboratory Experimental Results

3.1 Part 1: Signal Representation in Time Domain and Frequency Domain

In order to verify that the changes made to the code provided in part 1 worked, the results were to be analyzed. Below in **Figure 8** both the FFT as well as the time domain waveform can be seen. Looking at the time waveform, it can be verified that a sinusoid is present with a frequency of 50 Hz. In the time span of 0.06 second, 3 cycles are present meaning that in 1 second 50 peaks would be present. Moving attention to the FFT of the same signal, we can see two strong impulses at

50Hz and -50Hz which matches intuition of the frequency domain of a single sinusoid containing a frequency of 50 Hz.

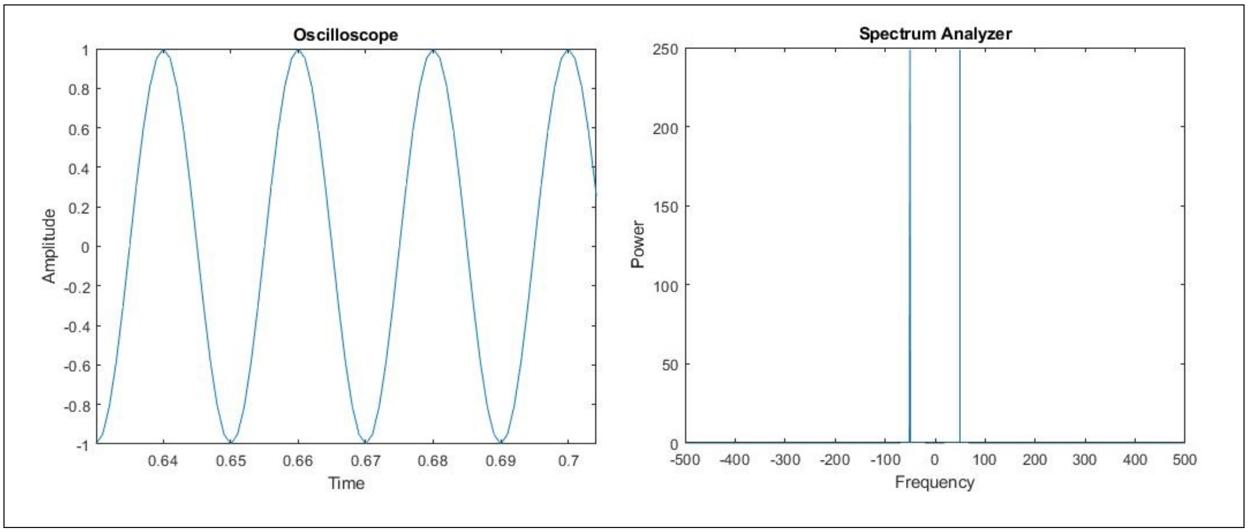

Figure 8: 50Hz Waveform and FFT

Now, adding a second sinusoid of frequency 40Hz to the previous 50Hz signal yields the below results in **Figure 9**. Looking at the time domain graph, it is shown that on top of the 50Hz signal is riding a slower 40Hz cosine wave which is to be expected from the sum of two sinusoids of differing frequencies. The FFT is similar to that shown above in **Figure 8**, yet there are another pair of impulses at 40 and -40Hz because of the presence of the pure 40Hz cosine wave.

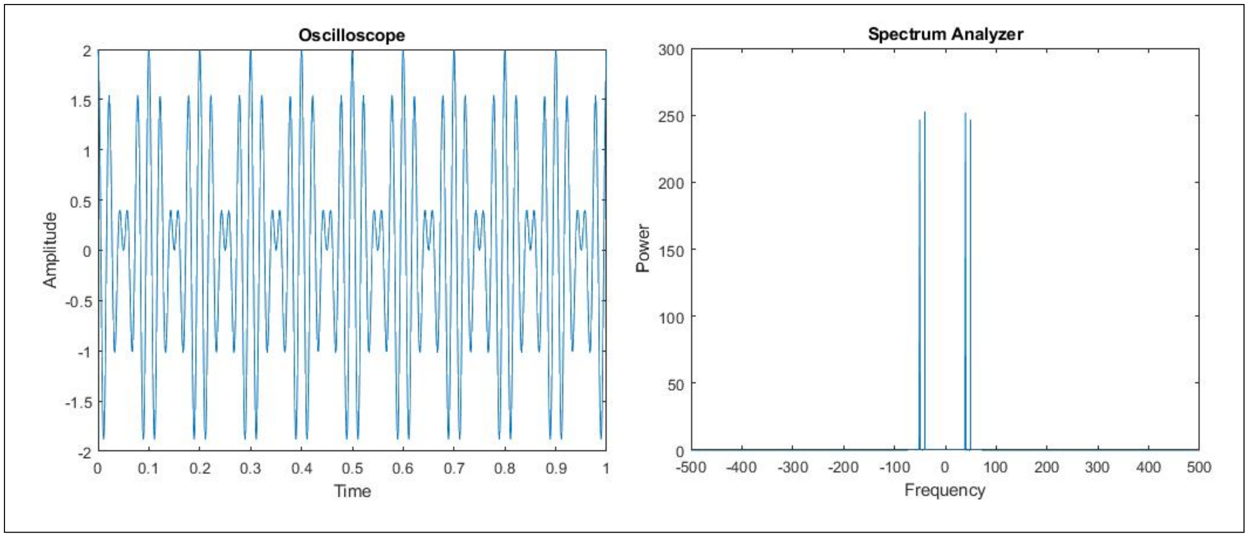

Figure 9: 50 and 40Hz Waveform and FFT

Adding a third cosine at a frequency of 200Hz and changing the frequency of the second signal to 100Hz grants no new exciting findings as the time waveform and FFT now have a higher frequency component at 200Hz. This can be verified by analyzing **Figure 10** below.

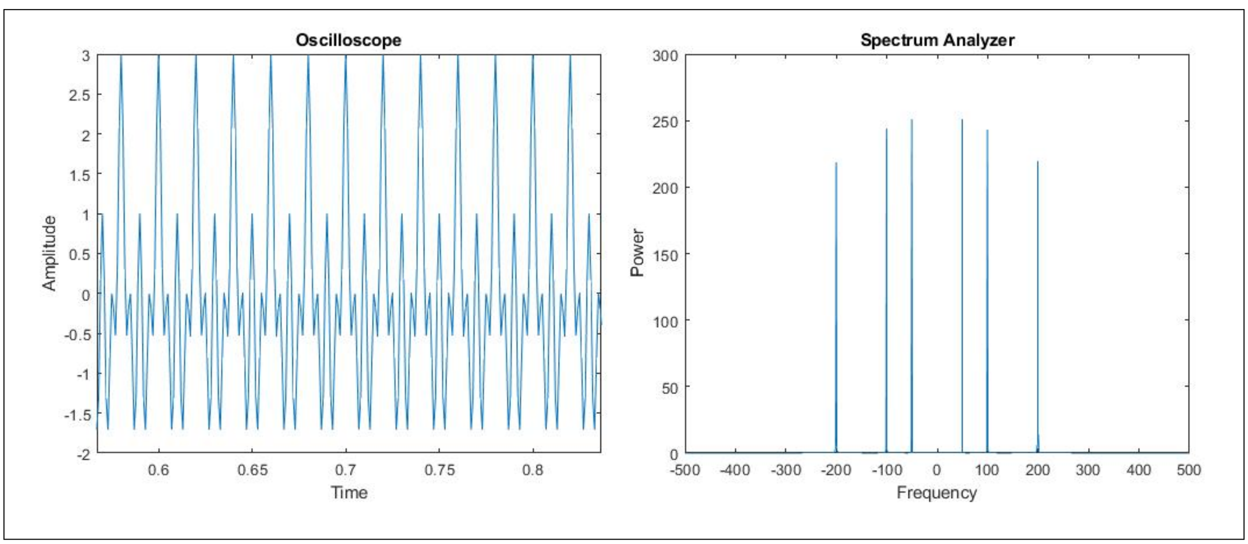

Figure 10: 50, 100, and 200Hz Waveform and FFT

Lastly, the 200Hz cosine was changed to have a frequency of 600Hz and the following results were generated below in **Figure 11**. What is interesting is that on the frequency domain side of the figure, an impulse appears at 400Hz instead of the 600Hz that was specified in the signal. The reasoning for this anomaly is frequency aliasing. Referring back to **Figure 1**, we can see that the original sampling frequency labeled “fs” is 1000Hz. According to the Nyquist theorem, in order for no frequency aliasing to occur, the sampling frequency must be at least twice the highest frequency component of the signal of interest. Since the highest frequency in the “merged” signal is 600Hz and twice that is 1200Hz, our sampling frequency has to be at least 1200Hz. In the case of the code that generated **Figure 10**, the sampling frequency being 1000Hz was not enough.

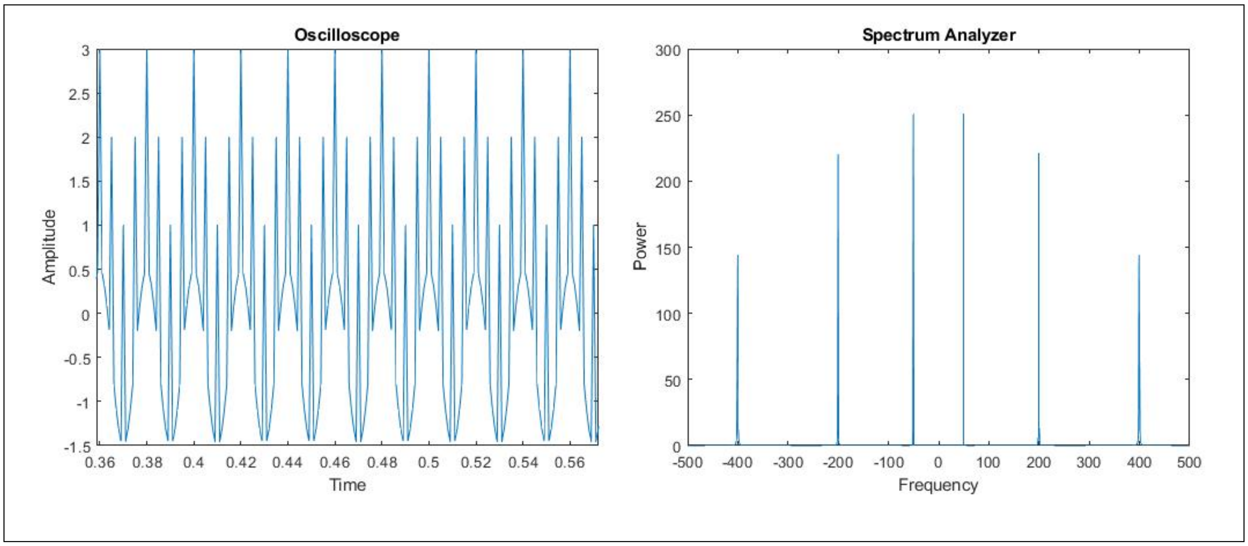

Figure 11: 50, 100, and 600Hz Waveform and FFT

3.2 Part 2: Display and Analyze DSB AM Modulated Wave

The results of Part 2 came in the form of waveforms and FFT’s of three different amplitude sensitivity factors 0.5, 1, and 10. Below in **Figures 12 and 13** these graphs can be seen.

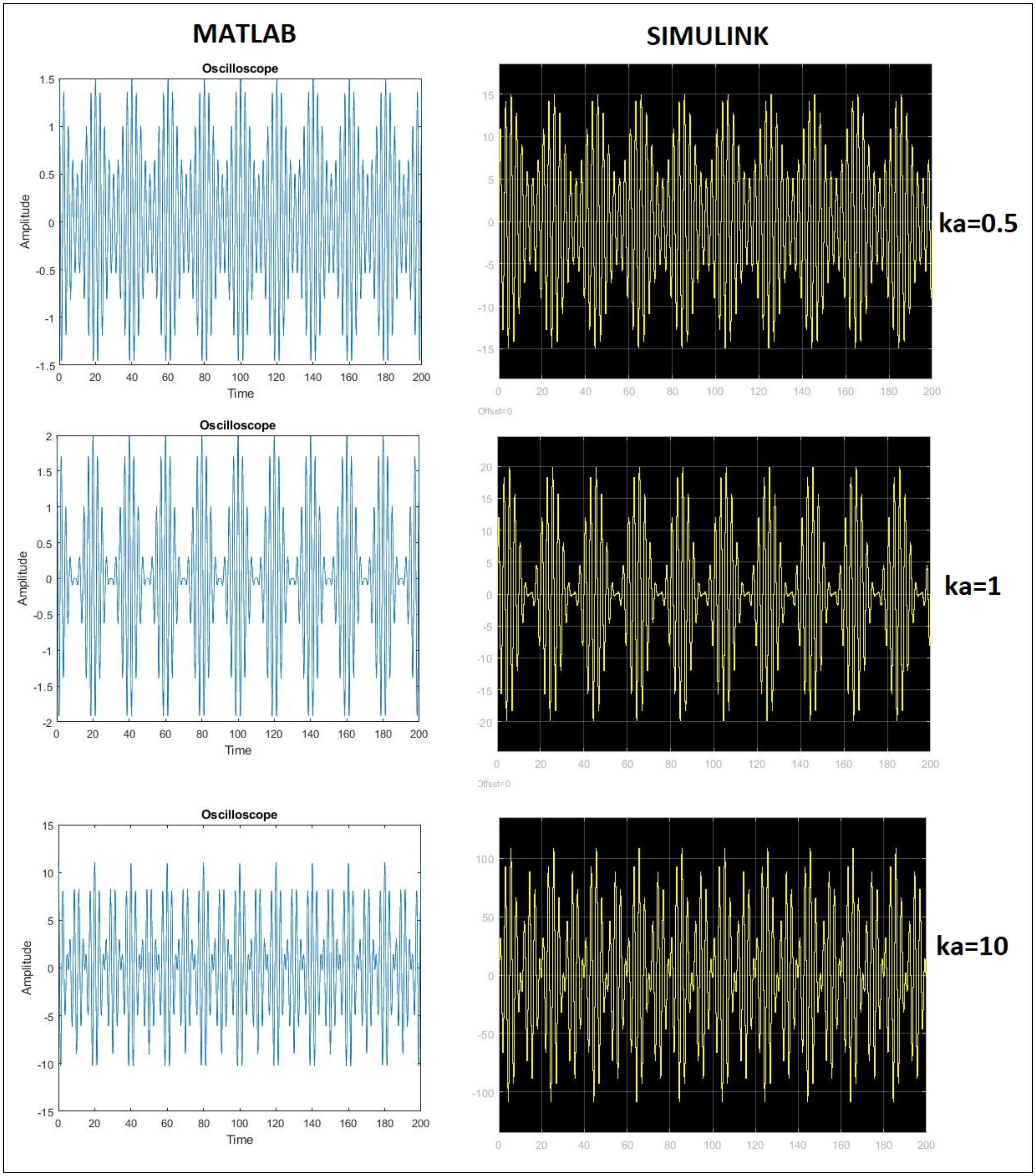

Figure 12: MATLAB vs Simulink Waveforms

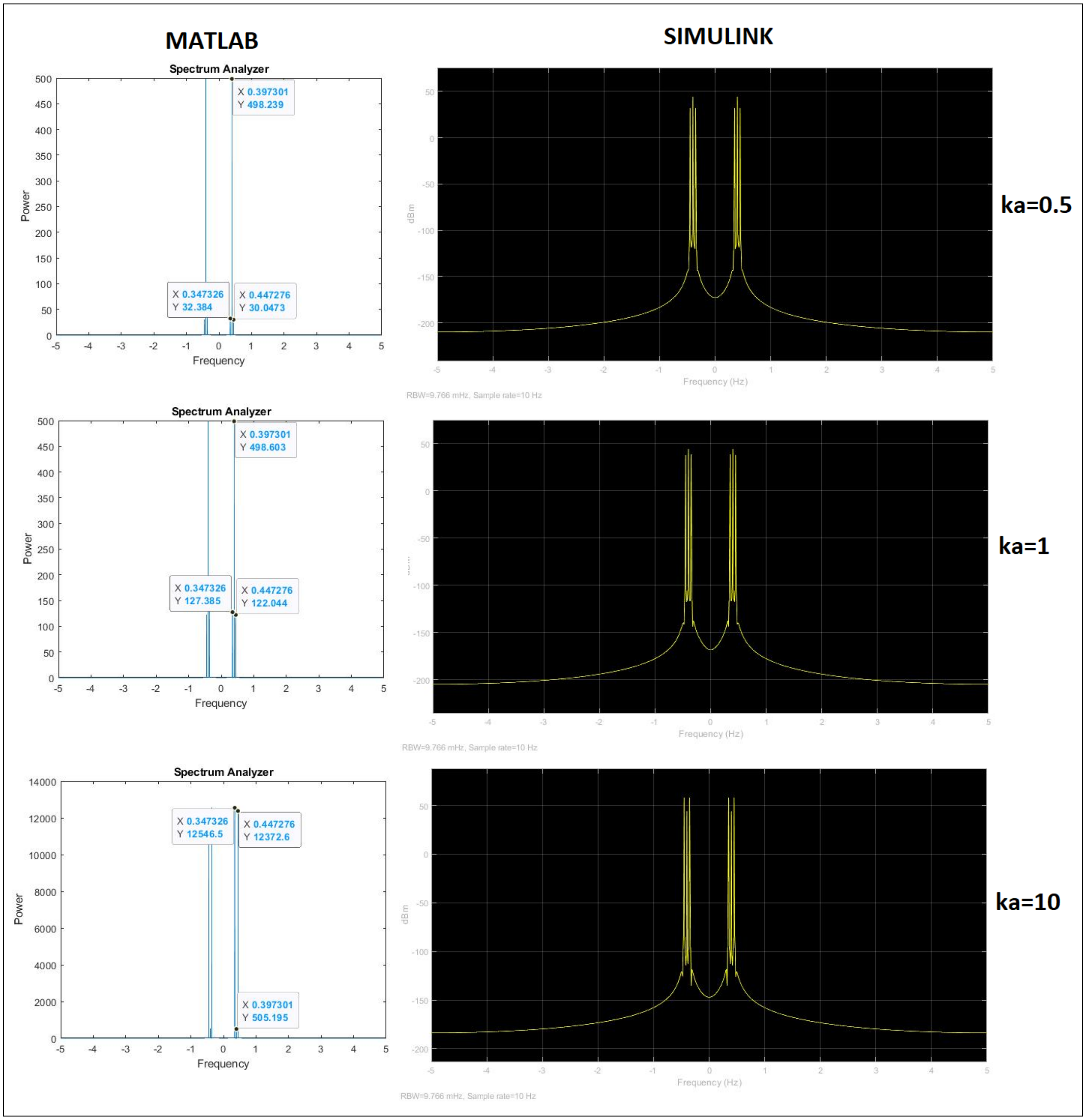

Figure 13: MATLAB vs Simulink FFTs

Analyzing the graphs above leads to interesting discoveries. First, when the amplitude sensitivity factor ka=0.5, we can clearly see a lower frequency sinusoidal signal riding on top of a sine wave that has high frequency comparatively. Even more apparent, when ka=1, the troughs of the low frequency sine wave are just about hitting the x-axis, yet the overall shape of the low frequency message signal is still present. Increasing the sensitivity factor even more to ka=10, the original low frequency sinusoidal message signal is in-distinguishable and thus not recoverable.From these waveforms it can be concluded that for an AM DSB-TC transmitter, a good rule of thumb is to keep the magnitude of the amplitude sensitivity factor at or below 1. In this way, the overall shape of the message signal can be recovered by the receiver.

Now looking at the FFT's in **Figure 13**, it is present that the transmitter does shift the low frequency message sinusoidal from 0.05Hz up to the carrier frequency of 0.4 leaving an impulse at 0.05+0.4=0.45Hz for all ka values. The carrier frequency is also present in all of these FFT's located at 0.4Hz. What might not be intuitive at first is where the lower frequency impulse at 0.35 comes from.That impulse is a direct consequence of the negative side of the message signal, that being -0.05Hz, being shifted up by the carrier frequency, 0.4Hz, leaving that impulse at -0.05+0.4=0.35. Both the MATLAB script as well as Simulink show these results.

3.3 Part 3: Spectrum Analysis with RTL-SDR Radio

The results of part 3 came in the form of various signal characteristics such as loss samples, passband spectrum and latency at different center frequencies. The Simulink example defaults to a center frequency of 102.5 MHz so the first attempt was ran with this parameter. The resulting characteristics can be seen in **Figure 14** below.

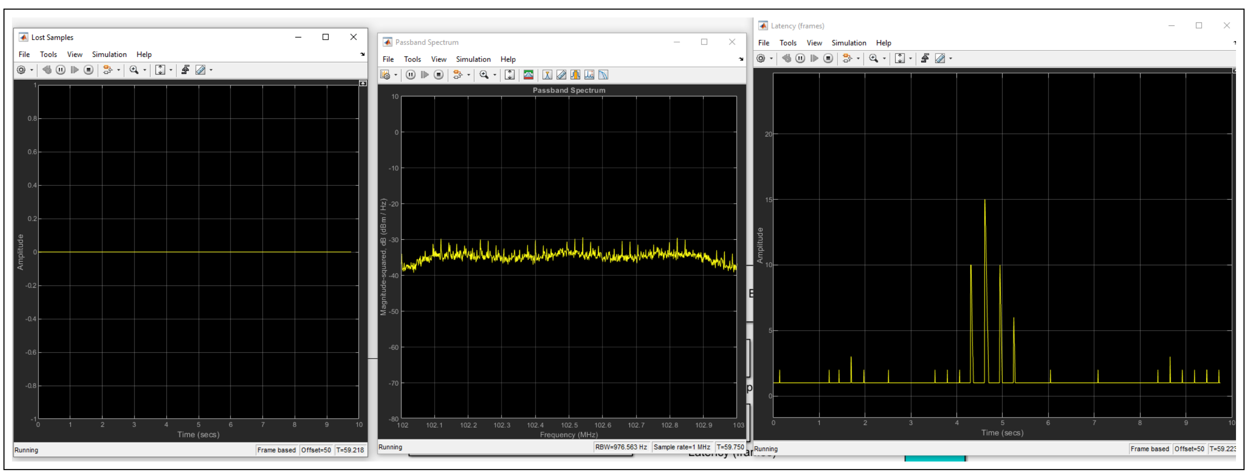

Figure 14: 102.5MHz Characteristics

To gain a better understanding of what was really being shown in this spectral analysis, this time another run through of the example was done but with the center frequency tuned to 92.3 MHz, a well known and frequently tuned in radio station that is active 24 hours everyday. This frequency yielded the characteristics viewed in **Figure 15** below.

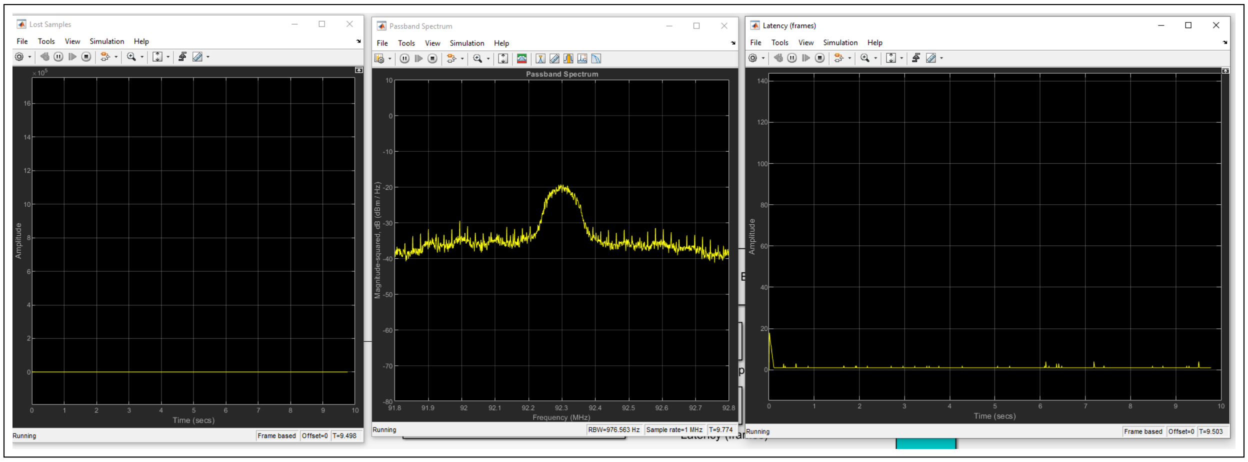

Figure 15: 92.3MHz Characteristics

3.4 Part 4: Build Your Own AM Radio

The working AM Radio was demoed in class. Below, one can see the spectrum of the transmitted signal in **Figure 16**, along with the spectrum of the received signal in **Figure 17**. Notice how wide the spectrum is for the transmitted signal compared to the spectrum of the received signal. The received signal having a smaller spectrum shows that the signal was correctly down converted to baseband allowing one to hear the transmitted audio.

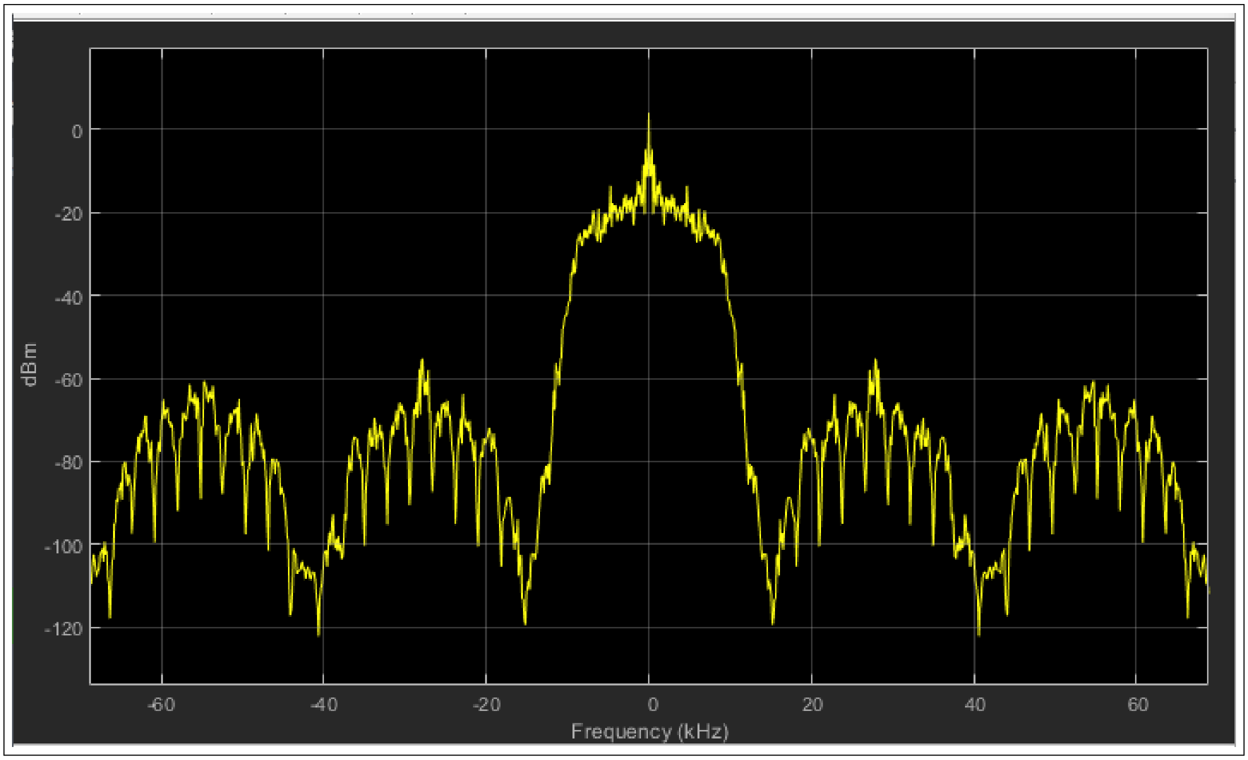

Figure 16: Spectrum of Transmitted Signal

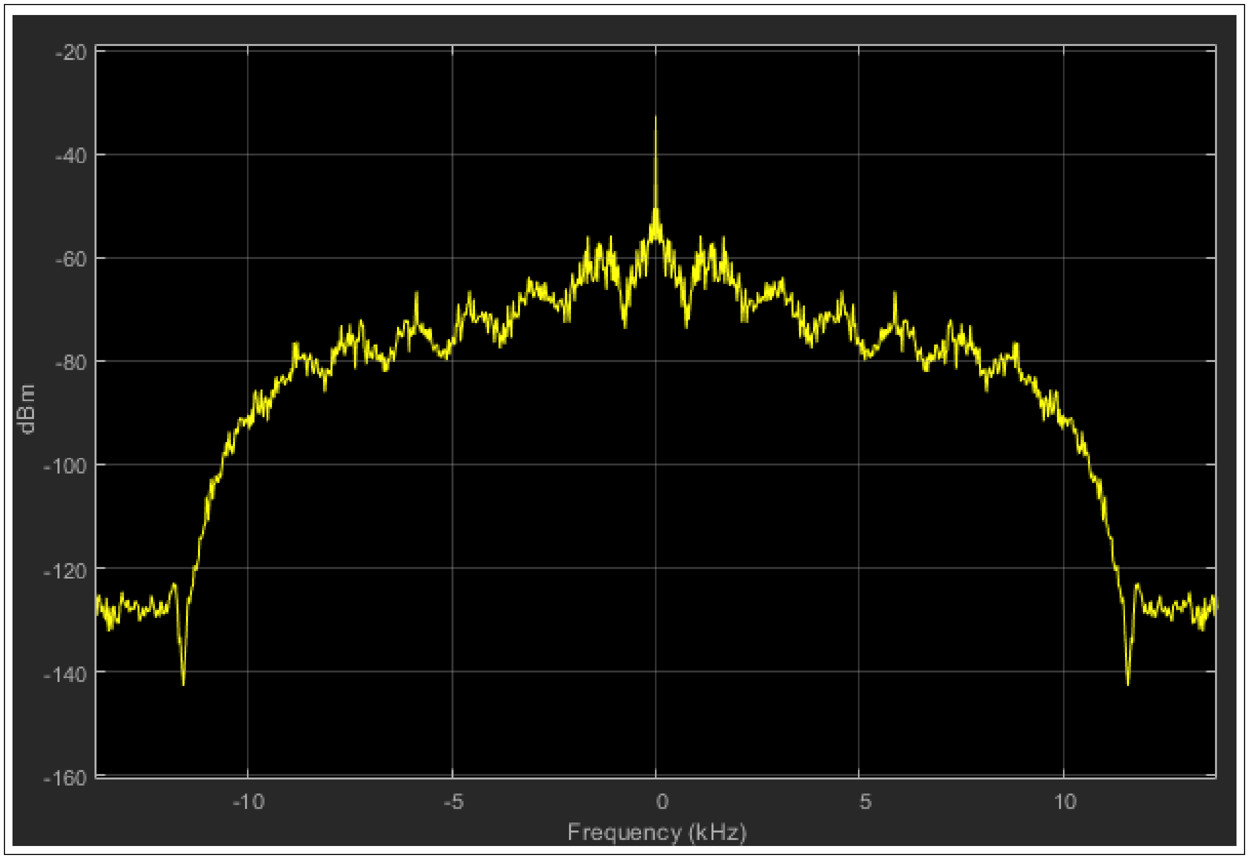

Figure 17: Spectrum of Received Signal

4 Discussion

4.1 Part 1: Signal Representation in Time Domain and Frequency Domain

Taking a look at the waveforms generated in **Figure 8**, we can see that when analyzing a single signal with a frequency component, in this case that being 50 Hz, through the spectrum analyzer we can see what components make up the incoming signal. This also holds true when modulating two signals with different frequency components which can be seen in **Figure 9**. The generated waveform was made by modulating two signals, one with a frequency of 50 Hz and the other with 40 Hz, which was then analyzed by the spectrum analyzer code. It can clearly be seen in this Figure that there are two frequency components present in the signal, one at 50 Hz and one at 40 Hz, which is exactly what we would have expected. A similar result occurs when looking at **Figure 10** which yields similar results as **Figure 9**, except now three frequency contents can be clear viewed when looking at the spectral analysis of the modulated signal.

Applying this same analysis to **Figure 11**, three frequency components would be expected, one at 50, 100 and 600 Hz. However when looking at the spectral analysis of this modulated signal, we only see components at 50, 100 and 400 Hz. The reason for this difference between the 400 and 600 Hz component is simply due to the frequency at which the signal is being sampled and the Nyquist Theorem. This theorem states that when sampling, the rate should be at least two times higher than our highest frequency component. Since 600 Hz is our highest component in this case, we would need a sampling rate of at least 1200 Hz which is 200 Hz more than the set sampling frequency value of 1000 Hz. This would mean that the highest component that we could measure accurately would be 500 Hz and anything greater than that would be affected by aliasing, distorting the expected 600 Hz into the observed 400 Hz.

4.2 Part 2: Display and Analyze DSB AM Modulated Wave

Peering at **Figure 12**, the generated DSB AM modulated wave can be observed at various ka values. Delving into these results, a clear relationship between the frequency sensitivity, ka, value and the quality of the signal strikes out. As ka becomes increasingly closer towards a value of 1, the signal's quality becomes much clear, and as the value continue to grow past a value of 1, in this case to 10, the quality of the signal becomes less than ideal. This ultimately suggests that the frequency sensitivity value of a DSB AM modulated wave should never be greater than a value 1 in order to maintain signal integrity.

After taking the Fourier Transform of the previous signals, viewed in **Figure 13**, we can see that the transmitter shifts the frequency of the message single from .05 Hz up the the carrier frequency of .4 Hz. This causes an impulse of the magnitude of the sum of the two frequencies, .45 Hz for each ka value, with the carrier remaining at .4 Hz. An impulse of .35 Hz can also be observed due to summation of the two signals, but on the negative side of the frequency spectrum, that is -.05 Hz is brought up to the carrier, resulting in .35 Hz.

4.3 Part 3: Spectrum Analysis with RTL-SDR Radio

Examine the content in **Figure 14**, we can see that at 102.5 MHz, there is nothing but noise present which is indicated by the middle graphic. it can also be seen that there are impulses of latency that tend to occur through the reception which can either be due to the physical hardware used or simply due to the quality of the transmitted signal. Since the RTL-SDR is a low power antenna, it should be kept in mind that the location of the transmitted signals may be too far to travels to the RTL, causing these spikes of latency.

The analysis becomes more interesting when taking a look at an active station like at 92.3 MHz, seen in **Figure 15** which comes from Providence, Rhode Island. An immediate difference can be seen simply by looking at the spectral analysis located in the middle of the figure. A clear hump is visible that wasn't present in the previous iteration. Looking at the accompanying latency and loss samples for the figure, it appears that there is much less latency occurring than with the reception at 102.5 MHz. Interestingly, there appears to be little to no loss in samples present for either station, which may not be accurate due to the hardware used for experimenting. Overall, it is clear that various characteristics of signals can change depending on the content being transmitted, as well as the physical location of the signal.

4.4 Part 4: Build Your Own AM Radio

When developing the AM broadcasting and receiving station, there are important steps that must be taken in order to successfully implement each functionality. Take for instance the transmission block diagram for Part 4 where the signal must be up-converted up to the target frequency of around 200 KHz which can then be brought into the Pluto-Radio block for transmission. Without this up conversion, the transmitted signal wouldn't propagate at a "unique" frequency, allowing for the potential to data or audio that is present in the signal to overlap with another without it. This same principle applies for the down-conversion used on the receiving end of the station as if the receiver is looking at for that "unique" frequency, it would be able to bring in back down to the target message's frequency and read the data present on the signal with no interference from outside signals riding at a similar frequency.

Viewing the graphs in Figures 16 and 17, we can see from the spectrum of our received and transmitted signals that the block diagram works as designed. This is evident by how wide/narrow the bands are for the received and transmitted signals, showing the concept of down and up converting the respectively signal works as theoretically intended.

5 Conclusion

In conclusion, the team was able to successfully explore AM modulation as an introduction to software defined radio. The students used MATLAB/Simulink in order to generate and analyze DSB AM signals in the time and frequency domains. Signal analysis was also done using an RTL-SDR to mainly analyze the characteristics of signals. The RTL-SDR was also used as a receiver in the AM radio designed in simulink for part 4 of the lab. The transmission for the AM radio was provided by the Pluto Radio. Designing this radio gave the students an understanding of when and where to up sample or down sample in order to correctly transmit or receive.

6 Recommendations

1. As mentioned, in order to successfully create an AM radio one needs to know when and where to up sample or down sample and by how much.
2. The received signal was also somewhat noisy and did have the highest volume, therefore one thing that can be done to fix the volume is increase the gain on the RTL-SDR receiver.

7 Laboratory reflection

Overall the laboratory went as well as it possibly could have. The real challenge for this lab was figuring out ways to coordinate with the group due to the limited hardware. If one member had the Pluto radio, then the team had to rely on that member to accomplish any transmission of signals for the lab. This was the only roadblock that was experienced while attempting this lab. As this is only the first lab of the semester, some hiccups are to be expected but as time progress and the group becomes situated, this shouldn't be much of a problem.

8 References

Lab 1 Handout:

UMass | Dartmouth
COLLEGE OF ENGINEERING

Electrical and Computer
Engineering Department

ECE 403-04/591-02 Software Defined Radio

Lab 1. Amplitude Modulation

Objective: Be familiar with amplitude modulation; be able to display and analyze AM modulated wave in MATLAB and Simulink; be able to transmit and receive over-the-air AM signals.

1 Introduction

Modulation is defined as the process by which some characteristic of a carrier wave is varied in accordance with an information-bearing signal. The carrier is needed to facilitate the transportation of the modulated signal across a band-pass channel from the transmitter to the receiver. A commonly used carrier is a sinusoidal wave, the source of which is physically independent of the source of the information-bearing signal. When the information-bearing signal is of an analog kind, we speak of continuous-wave modulation, a term that stresses continuity of the modulated wave as a function of time.

A sinusoidal carrier wave is $c(t) = A_c \cos(2\pi f_c t)$, where A_c is the carrier amplitude and f_c is the carrier frequency, assume the phase of the carrier wave is zero. AM is formally defined as a process in which the amplitude of the carrier wave $c(t)$ is varied about a mean value, linearly with the message signal $m(t)$: $s(t) = A_c[1 + k_a m(t)] \cos(2\pi f_c t)$, where k_a is a constant called the amplitude sensitivity of the modulator responsible for the generation of the modulated signal $s(t)$. Typically, the carrier amplitude A_c and the message signal $m(t)$ are measured in volts, in which case the amplitude sensitivity k_a is measured in volt^{-1}.

In AM, the amplitude of the modulated wave $s(t)$ is $A_c\,|1 + k_z m(t)|$. Thus, the envelop of $s(t)$ has essentially the same shape as the message signal $m(t)$ provided that two conditions are satisfied:

1. The amplitude of $k_a m(t)$ is always less than unity, $|k_a m(t)| < 1$ for all t. This condition ensures that the function $1 + k_a m(t)$ is always positive. Then, the envelop of the AM wave is $A_c[1 + k_a m(t)]$. When the amplitude sensitivity k_a of the modulator is large enough to make $|k_a m(t)| > 1$ for any t, the carrier wave becomes *over modulated*, resulting in carrier phase reversals whenever the factor $1 + k_a m(t)$ crosses zero. The modulated wave then exhibits envelop distortion.

By avoiding overmodulation, a one-to-one relationship is maintained between the envelope of the AM wave and the modulating wave for all values of time. The absolute maximum value of $k_a m(t)$ multiplied by 100 is referred to as the percentage modulation.

2. The carrier frequency f_c is much greater than the highest frequency component W of the message signal $m(t)$, which is $f_c >> W$, where W is the message bandwidth. If this condition is not satisfied, an envelope cannot be visualized (and therefore detected).

If these two conditions are satisfied, demodulation of the AM wave is achieved by using an envelop detector, which is defined as a device whose output traces the envelope of the AM wave acting as the input signal.

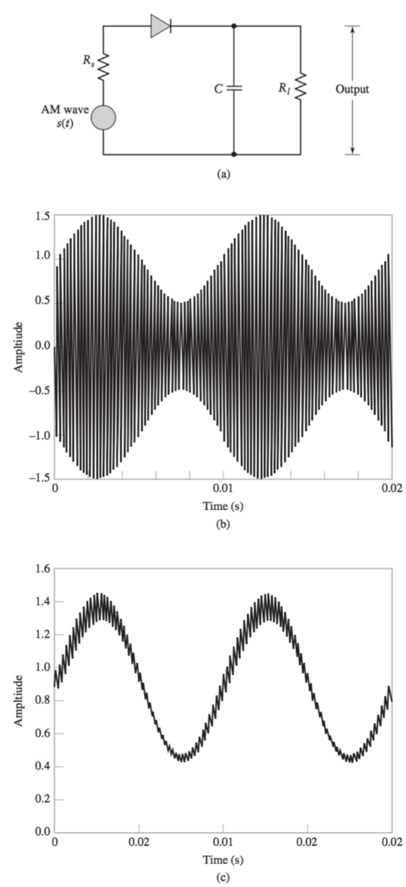

Figure 1: Envelop Detector. (a) Circuit diagram; (b) AM Wave Input; (c) Envelope Detector Output.

Double Sideband-Suppressed Carrier Modulation (DSB-SC) modulation consists of the product of the message signal $m(t)$ and the carrier wave $c(t)$:

$$s(t) = c(t)m(t) = A_c \cos(2\pi f_c t)m(t) \tag{1}$$

Accordingly, the device used to generate the DSB-SC modulated wave is referred to as a *product modulator*. From the Eq. 1, DSB-SC modulation is reduced to zero whenever the message signal $m(t)$ is switched off. The modulated signal $s(t)$ undergoes a phase reversal whenever the message signal $m(t)$ crosses zero. The envelope of a DSB-SC modulated signal is therefore different from the message signal, which means that the simple demodulation using an envelope detection is not a viable option for DSB-SC modulation. Therefore, the recovery of the message signal $m(t)$ can be accomplished by first multiplying s(t) with a locally generated sinusoidal wave and then low-pass filtering the product. It is assumed that the local oscillator signal is exactly coherent or synchronized, in both frequency and phase, with the carrier wave c(t) used in the product modulator to generate s(t). This method of demodulation is know as coherent detection or synchronous demodulation, illustrated in Fig. 2.

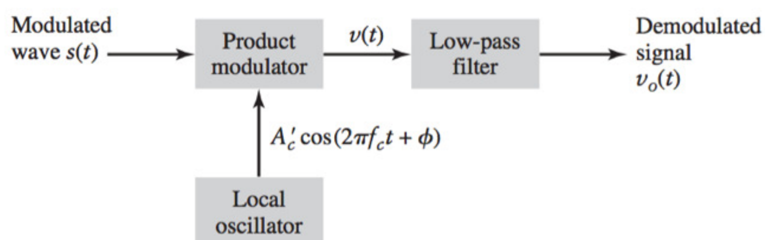

FIGURE 3.12 Block diagram of coherent detector, assuming that the local oscillator is out of phase by ϕ with respect to the sinusoidal carrier oscillator in the transmitter.

Figure 2: Block Diagram of Coherent Detector, assuming that the local oscillator is out of phase by ϕ with respect to the sinusoidal carrier oscillator in the transmitter

DSB-SC offers no advantage over AM in bandwidth occupancy. But it saves transmission power, which is important enough when the available transmitted power is at a premium.

2 Design Requirements

2.1 Part 1: Signal Representation in Time Domain and Frequency Domain

Understand the given code below. What is the sampling rate? What is the sample duration? What if `sig` is a mixture of two sinusoidal waves with 50Hz and 40Hz?

What if `sig` is a mixture of three sinusoidal waves with 200Hz, 100Hz, and 50Hz? What happens if `sig` is a mixture of three sinusoidal waves with 600Hz, 100Hz, and 50Hz?

```
% ECE403/591
% A sinusoidal wave in time and frequency domain
clear all;
close all;

fc = 50;
Ts = 0.001;
t = [0:Ts:1];
n = length(t);
fs = 1/Ts;
sig = cos(2*pi*fc.*t);
Y=fft(sig);
Y=fftshift(Y);
fshift = (-n/2:n/2-1)*(fs/n);
powershift = abs(Y).^2/n;

figure(1);
plot(t, sig);
xlabel('Time');
ylabel('Amplitude');
title('Oscilloscope');
figure(2);
plot(fshift, powershift);
xlabel('Frequency');
ylabel('Power');
title('Spectrum Analyzer');
```

2.2 Part 2: Display and Analyze DSB AM Modulated Wave

Use the following parameters to display and analyze an DSB AM modulated wave in Matlab and Simulink. Discuss your observations.

- Carrier Amplitude $A_c = 1$
- Carrier Frequency $f_c = 0.4Hz$
- Modulation frequency $f_m = 0.05Hz$
- Display and analyze 10 full cycles of the modulated wave, corresponding to a total duration of 200 seconds. Assume the modulated wave is sampled at $f_s = 10Hz$, obtaining a total of $200 \times f_s = 2000$ data points. The frequency band occupied by the modulated wave is $-5Hz \leq f \leq 5Hz$.

- When $\mu = 0.5$, the envelope of the modulated wave is clearly seen to follow the sinusoidal modulating wave. So we can use an envelope detector for demodulation. $f_c = \pm 0.4Hz$, $f_c - f_m = \pm 0.35Hz$, and $f_c + f_m = \pm 0.45Hz$.
- When $\mu = 1$, the modulated wave is on the verge of overmodulation.
- When $\mu = 10$, the modulated wave is overmodulated. There is no clear relationship between the envelope of the overmodulated wave and the sinusoidal modulating wave. The envelope detector will not work for $\mu > 1$.

2.3 Part 3: Spectrum Analysis with RTL-SDR Radio

Run the Simulink "Spectrum Analysis with RTL-SDR Radio" example, and measure signal characteristics.

2.4 Part 4: Build Your Own AM Radio

Use Pluto and RTL-SDR to build your own AM radio using MATLAB or Simulink. You may reference the examples in the free ebook of "Software Defined Radio using RTL-SDR, MATLAB and Simulink" at `https://www.mathworks.com/campaigns/offers/download-rtl-sdr-ebook.html`.

3 Grading

- Part 1: 10%
- Part 2: 20%
- Part 3: 10%
- Part 4: 40%
- Report: 20%

9 Appendices