ECE 203
Lab 7: Radio Havana

Verification: Complete the warm-up section of the lab during your assigned lab time. Make sure an instructor signs the verification sheet found at the end of the lab report.

Lab Report: Submit a lab report for the entire lab using MATLAB publisher.

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1 Introduction

The objective of this lab is to use MATLAB to demodulate real AM radio signals. In particular, we'll demodulate and listen to a number of radio stations that broadcast around the frequency of 6.0 MHz. Because of the propagation characteristics of radio waves at these frequencies, a single location in the U.S. can pick up stations from around the world. To learn more, check out http://en.wikipedia.org/wiki/Shortwave_radio.

2 Overview

Amplitude Modulation, or simply AM, was the first technique for transmitting voice and music using radio waves. To generate an AM radio, an audio signal is simply multiplied by a sine wave. If $a_1(t)$ is the audio signal, then the signal transmit by the radio station is

$$x(t) = a_1(t)\cos(2\pi f_1 t + \phi_1) \tag{1}$$

where f_1 is the carrier frequency of the radio station, and ϕ_1 is the phase. One of the stations we'll listen to in this lab is Radio Havana, transmit out of Havana Cuba. Radio Havana uses $f_1 = 6.0$ MHz. The maximum frequency component in $a(t)$ is 5 kHz, which means the radio station occupies frequencies from about 5995 kHz to 6005 kHz.

The over the air signal we'll use in this lab contains a number of radio stations (5 at least, perhaps up to 15). Ignoring amplitude gains or radio propagation effects, we can write the received signal at our antenna as a sum of the transmit signals from individual stations:

$$x(t) = \sum_i a_i(t)\cos(2\pi f_i t + \phi_i) \tag{2}$$

where f_i is the carrier frequency of each station, and ϕ_i is some unknown phase.

Over the air data was collected using an antenna and a simple receiver. The receiver was setup to sample frequencies from about 5900 kHz to 6100 kHz, well beyond the bandwidth of Radio Havana (and thus collecting a number of other stations too). After the over the air

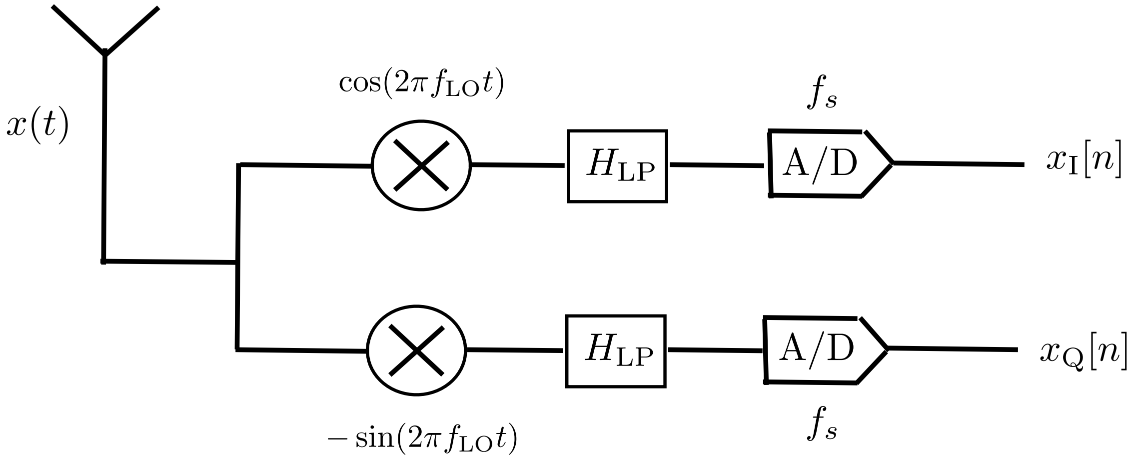

Figure 1: I/Q Receiver. The data in this lab was collected with $f_{\mathrm{LO}} = 6000$ kHz, and $f_s = 196.078$ kHz.

signal arrives at the antenna (and perhaps some input filtering and amplification), the radio receiver makes two analog multiplications with a cosine and a sine:

$$x_{\mathrm{I}}(t) = x(t) \times \cos(2\pi f_{\mathrm{LO}} t) \qquad x_{\mathrm{Q}}(t) = x(t) \times -\sin(2\pi f_{\mathrm{LO}} t) \tag{3}$$

Here, I and Q stand for 'in-phase' and 'quadrature', and $f_{\mathrm{LO}} = 6000$ kHz is the frequency of the *local oscillator* at the receiver. This multiplication step is called down conversion, as it is brings the high frequency signal down to a lower frequency.

Both $x_{\mathrm{I}}(t)$ and $x_{\mathrm{Q}}(t)$ are low pass filtered with H_{LP}, an *analog* low pass filter that is designed to reject high frequency components and prevent aliasing. Analog filters operate on continuous time signals and are made of capacitors and inductors. Conceptually, they can be thought of in a similar way to discrete time filters – for example, a low pass filter only passes low frequency continuous time sinusoids.

After $x_{\mathrm{I}}(t)$ and $x_{\mathrm{Q}}(t)$ pass the analog low pass filter H_{LP}, they are sampled by the analog to digital converters in the receiver, and saved into a `.mat` file. The sampling rate is $f_s = 196.078$ kHz. The block diagram of this receiver is shown in Fig. 1.

3 Warm-up

3.1 Plotting the Spectrum

The goal of this section is to plot the radio frequency spectrum of the over the air data. Download the file `'shortwave.mat'` from the course website. To load the file, you can type

```
load shortwave.mat;
```

The vector `raw` should contain two rows, corresponding to $x_{\mathrm{I}}[n]$ and $x_{\mathrm{Q}}[n]$ shown in Fig. 1. Since the in-phase and quadrature components are 90 degrees out of phase, we can combine them in a single complex number. In communications, this is called *complex baseband* notation.

```
x = raw(:,1) + 1i*raw(:,2);
```

Next, take the DFT of `x`. Use `fftshift` to center the spectrum:

```
X = fftshift(fft(x));
```

Plot the spectrum of the signal. Make sure the x-axis on your plot represents the frequencies of the signal that arrives at the antenna – i.e., the x-axis should start at $(f_{\mathrm{LO}} - f_s/2)$ Hz and go through $(f_{\mathrm{LO}} + f_s/2)$, and be centered around f_{LO}. Also, the y-axis should be in decibels, obtained by typing `20*log10(abs(X))`. Make sure to label your axis.

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You should be able to see a number of shortwave AM radio stations. Radio Havana should be at the center, at 6.0 MHz, and have bandwidth of about 10 kHz. Include this plot in your final lab report.

3.2 Understanding the Receiver

There are a number of reasons why the analog processing must take place before the signal is sampled.

Question 3.1. Say we wanted to implement a receiver that directly sampled the over the air signal (without the multiplication step shown in Fig. 1). Since the highest frequency components of Radio Havana are at 6.005 MHz, at what sampling rate would the A/D have need to operate to avoid aliasing?

Question 3.2. How many samples would we need to collect 1 minute of audio at this sampling rate? Explain why this might be prohibitive.

Question 3.3. The receiver above samples at $f_s = 196078$ Hz, much slower than the answer you got for question 3.1.

a. In order to avoid aliasing, what should the cutoff frequency of H_{LP} be? The cutoff frequency of a low pass filter is the highest frequency that can pass through the filter.

b. *In terms of frequencies that arrive at the antenna*, what are the highest and lowest frequencies that should get past H_{LP}?

c. If you zoom in on the edges of the spectrum plot, you should be able to see the effect of H_{LP}. What is the true cutoff frequency of the low pass filter (approximately)?

Question 3.4. It might not be immediately clear why both an in-phase and quadrature multiplication are necessary in the receiver. To see why, imagine the phase of the transmit signal of Radio Havana is $\phi_1 = \pi/2$. If we only look at the $x_{\mathrm{I}}(t)$ path, we have:

$$x_{\mathrm{I}}(t) = a_1(t)\cos(2\pi f_1 t + \phi_1) \times \cos(2\pi f_{\mathrm{LO}} t).$$

Recall $f_1 = 6000$ kHz and $f_{\mathrm{LO}} = 6000$ kHz as well. Write $x_{\mathrm{I}}(t)$ using complex exponentials. What happens to this signal after low-pass filtering if $\phi_1 = \pi/2$?

Question 3.5. How does a second multiplication with the sine wave mitigate this problem?

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4 Lab Exercise

4.1 A simple demodulator

In order to listen to the audio being played on Radio Havana and the other stations, we need to demodulate the signal. Demodulation is the process of extracting $a(t)$ from $x(t)$. For AM signals like the shortwave radio signal used in this lab, demodulation is fairly straight forward.

Recall we used *complex baseband* notation in Section 3.1 – in MATLAB, we represented the sampled signal as $x[n] = x_{\mathrm{I}}[n] + jx_{\mathrm{Q}}[n]$. To see why this is useful, if there were only *one* radio station, we could write the continuous time in-phase and quadrature components before the low pass filter as (before H_{LP} in Fig. 1) as:

$$x_{\mathrm{I}}(t) = a_1(t)\cos(2\pi f_1 t + \phi_1) \times \cos(2\pi f_{\mathrm{LO}} t)$$

and

$$x_{\mathrm{Q}}(t) = a_1(t)\cos(2\pi f_1 t + \phi_1) \times -\sin(2\pi f_{\mathrm{LO}} t)$$

After low pass filtering (just past H_{LP} in Fig. 1), provided the difference between f_{LO} and f_1 is less than the cutoff frequency of H_{LP}, we can write

$$x_{\mathrm{I}}(t) = \frac{a_1(t)}{2}\cos(2\pi t(f_1 - f_{\mathrm{LO}}) + \phi_1)$$

and

$$x_{\mathrm{Q}}(t) = \frac{a_1(t)}{2}\sin(2\pi t(f_1 - f_{\mathrm{LO}}) + \phi_1)$$

Thus, after sampling, we can write the signal as

$$x[n] = x_{\mathrm{I}}[n] + jx_{\mathrm{Q}}[n] = \frac{a_1(n/f_s)}{2}\, e^{j\left(2\pi n \frac{f_1 - f_{\mathrm{LO}}}{f_s} + \phi_1\right)}.$$

To recover the sampled audio, $a_1(n/f_s)$, we need two steps:

Step 1) Multiply $x[n]$ by a complex exponential at the digital frequency $-(f_1 - f_{\text{LO}})/f_s$, which gives:

$$x[n] \times e^{-j2\pi n \frac{f_1 - f_{\text{LO}}}{f_s}} = \frac{a_1(n/f_s)}{2} e^{j\phi_1} \quad (4)$$

Step 2) Multiply by $e^{-j\phi_1}$, which removes the phase of the signal, and gives us back the sampled audio.

4.2 Finding the carrier frequency

To implement the demodulation described above, we'll need to find the value of $(f_1 - f_{\text{LO}})/f_s$. One thing that can make demodulation a bit more challenging is that the frequency of the radio station and of the local oscillator are not exact. In the spectrum plot you made in Section 3.1, notice that each radio station has a high, narrow peak at its center frequency. This is the carrier frequency of the station, f_i, and is about 40 dB above the rest of the signal. We can use this peak to get a good estimate of f_i.

To see that the carrier frequency is not exact, use the spectrum plot, and zoom in on the peak of Radio Havana. The carrier frequency, f_1, isn't at exactly at 6 MHz – it's closer to 6.0001 MHz. This could also be the result of a small inaccuracy in f_{LO}.

We'll need to find a good estimate for the carrier frequency of the radio station. To do this, write code that takes an approximate carrier frequency as an input (for example, 6000 kHz). The code should then use the DFT of the signal to find an estimate of the true carrier frequency of the station, f_i. The following lines might help. You have to define `minInd` and `maxInd`, which should be indices corresponding to the minimum and maximum range in which you expect to find the true carrier frequency (for example, `minInd` could be the index of `X` corresponding to 5999 kHz and `maxInd` to 6001 kHz).

```
load shortwave.mat
f_LO = 6000e3;
x = raw(:,1) + 1i*raw(:,2);
X = fftshift(fft(x)/length(x));
N = length(X);
freqs = linspace(f_LO - Fs/2,f_LO + Fs/2,N);
minInd = ???
maxInd = ???
[maxVal ind0] = max(abs(X(minInd:maxInd)));
f_i = freqs(minInd+ind0);
```

The output, `f_i`, should be the estimated carrier frequency of the radio station.

4.3 Implementing the demodulator

Once we have an estimate of f_i, the rest of the demodulator is relatively straightforward.

4.3.1 Removing the carrier frequency

Recall that the sampled audio signal, $a_i(n/f_s)$, has been multiplied by a complex exponential with frequency corresponding to the digital frequency $(f_{\text{LO}} - f_i)/f_s$. To undo this, we multiply the signal by a complex exponential, according to equation (4):

```
a_raw = x.*exp(-1i*2*pi*[1:N]'*(f_i - f_LO)/Fs);
```

This multiplication of the signal with a complex sinusoid will simply shift the center of radio station to a digital frequency of 0. Plot the spectrum of `a_raw` in dB. Make the x-axis digital frequencies. Zoom in on digital frequency zero. If your code worked, there should be a peak at zero.

4.3.2 Removing the phase

To finish demodulating the signal, we'd like to remove the phase according to (4). In theory, since $a_i(t)$ is a real number, we could use look at $a_i(n/f_s)e^{j\phi_1}$ to estimate ϕ_1. In practice, the phase of the signal is noisy, and varies as a function of time. Additionally, other radio stations are still present at higher frequencies, making phase estimation difficult. Since we don't have a good estimate of the phase, we can just take the real portion of the signal:

```
a = real(a_raw);
```

You can listen to the audio using `soundsc`:

```
soundsc(a,Fs)
```

Question 4.1. How does the audio on Radio Havana sound? How is the sound quality? Try a few other stations, and comment on the quality of the sound.

4.4 Low Pass Filtering

Demodulation shifted the radio station so that its spectrum is centered at 0 Hz, but the signals from all the other stations are still included at higher frequencies. To extract the signal of interest you must set all frequency components outside of the lowpass band to zero. The width of the lowpass band is proportional to the bandwidth of the radio station's signal (typically 3000 to 5000 Hz). There are two basic approaches to extracting the signal components of interest: (1) using the DFT; (2) using an FIR lowpass filter. After low pass filtering the signal using the `conv` command, you can play back the resulting output signal `y` the `soundsc` command.

```
soundsc(real(y),Fs)
```

1 – Using the DFT. DFT based methods process a section or "block" of the signal, so in a real software-defined radio this would require buffering the signal while each block of samples is processed. To process each block of N samples, the DFT is computed, interfering radio station signals at higher frequencies are set to zero, and the inverse DFT is used to reconstruct a signal that contains only the station of interest. *Devise an DFT based scheme for extracting radio stations of interest. Include your code and plots of the spectrum before and after processing. Also listen to the resulting signal and discuss the improvements in sound quality after processing.*

2 – Using FIR filters. DFT methods require breaking a streaming signal into blocks for processing, and this additional overhead and processing can be avoided using FIR filters. The streaming radio signal can be processed in (near) real-time by convolving it with an appropriate lowpass FIR filter that attenuates higher frequency components that correspond to interfering stations. *Construct lowpass FIR filters using two approaches:*

a. a truncated "sinc" filter;

b. a Parks-Mclellan filter using the Matlab function `firpm`.

Specify the same "cut-off" frequency and filter length (e.g., 200) for both filters. Plot the frequency response of the filter (e.g., using `freqz`*). Include your code and a discussion of your approach.*

Question 4.2. How does the cutoff frequency of the low-pass filter affect the quality of the audio on Radio Havana? Explain the effect on quality in terms of the frequency response characteristic of the filter.

Question 4.3. How does the filter length of the low-pass filter affect the quality of the audio on Radio Havana? Explain the effect on quality in terms of the frequency response characteristic of the filter.

Question 4.4. Compare the frequency responses of the sinc and Parks-Mclellan filters. Which filter has better high frequency suppression characteristics?

Question 4.5. Listen to the resulting signal after filtering with both filters. Which has better sound quality?

Question 4.6. Try applying the filter multiple times (i.e., a cascade of the same FIR filter). Does the cascade improve the sound quality? Explain why or why not in terms of the frequency response of the overall filter corresponding to the cascade.

4.5 Listening to the stations

Once you have your demodulation and filtering code working well, try listening to other stations. Below are questions pertaining to the dialog on two of the other stations. Give the answers along with the demodulation frequency you used to listen to the stations; i.e., the correct value of `f_i` needed in this step:

```
a_raw = x.*exp(-1i*2*pi*[1:N]'*(f_i - f_LO)/Fs);
```

Question 4.7. According to the British man, how many developers are there in China?

Question 4.8. What is the open house event you don't want to miss?

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Name: ______________________________

Lab 7, verification 1: __________________ Date: _____________

Lab 7, verification 2: __________________ Date: _____________