

ECE 203

PreLab 9: Radio Havana

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

You have learned about the basics of Amplitude Modulation, or simply AM, in the accompanying video. An AM radio signal is generated by multiplying an audio signal a sine wave. If $a(t)$ is the audio signal, then the signal transmitted by the radio station is

$$x_T(t) = a(t) \cos(2\pi f_1 t) \quad (1)$$

where f_1 is the carrier frequency of the radio station. One of the stations we'll listen to in this lab is Radio Havana, transmitted 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 received signal experiences attenuation (which we will ignore) and a phase shift due to transmission. Thus we may write the received signal as

$$x(t) = a(t) \cos(2\pi f_1 t + \phi_1) \quad (2)$$

If both f_1 and ϕ_1 are known exactly, then we can recover the message $a(t)$ by multiplying $x(t)$ with $\cos(2\pi f_1 t + \phi_1)$ and lowpass filtering as presented in the video. In a practical AM transmission neither of these are known exactly, so a slightly more complicated approach is used to recover the message.

3 Practical AM Demodulation: Uncertain Phase and Carrier Frequency

Uncertainty about the phase shift ϕ_1 and the exact value of the carrier frequency f_1 motivate use of a so-called quadrature receiver that multiplies $x(t)$ with both a cosine and a sine

$$x_I(t) = x(t) \times \cos(2\pi \tilde{f}_1 t) \quad x_Q(t) = x(t) \times -\sin(2\pi \tilde{f}_1 t) \quad (3)$$

Here \tilde{f}_1 is the estimated carrier frequency. A block diagram depicting the quadrature receiver for obtaining $x_I(t)$ and $x_Q(t)$ is shown in Fig. 1. Note that the lowpass filters H_{LP} are not shown in Eq. 3. Substituting for $x(t)$ using Eq. 2, simplifying the products of sinusoids, and

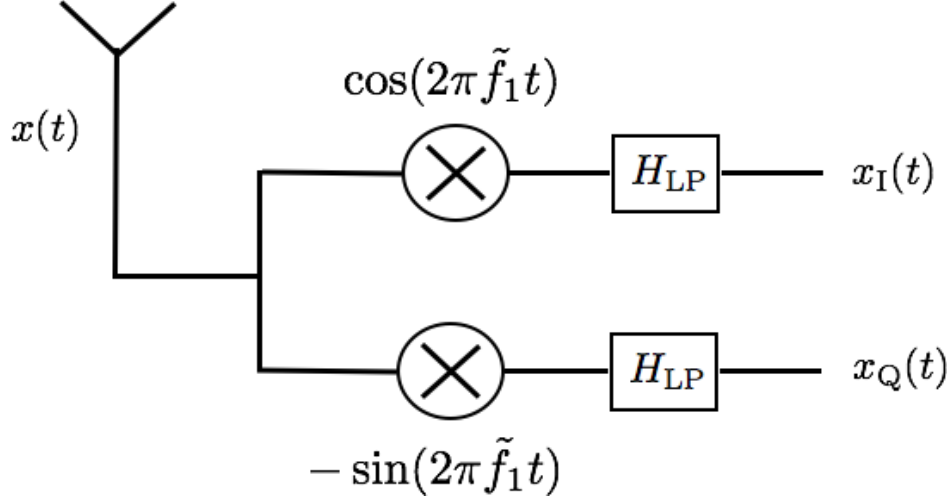


Figure 1: Block diagram of quadrature receiver.

using the lowpass filter to eliminate the terms at $f_1 + \tilde{f}_1$ gives

$$x_I(t) = a(t) \cos(2\pi(f_1 - \tilde{f}_1)t + \phi_1) \quad x_Q(t) = a(t) \sin(2\pi(f_1 - \tilde{f}_1)t + \phi_1) \quad (4)$$

Note that we have ignored the amplitude factor of $1/2$ associated with simplifying the products of sinusoids. As with attenuation due to propagation, such amplitude effects are addressed by amplification in the receiver and are not central to understanding how AM radio functions.

Having access to both $x_I(t)$ and $x_Q(t)$ allows us to compensate for uncertainty in f_1 and ϕ_1 . The details of how this compensation is done in a real AM receiver are complex and covered in later courses on communication systems. In this lab we will use the DFT to find f_1 exactly. We will not further address finding an unknown ϕ_1 .

It is very useful to combine $x_I(t)$ and $x_Q(t)$ into a complex-valued signal

$$x_{IQ}(t) = x_I(t) + jx_Q(t) \quad (5)$$

Note that using Euler's representation we may write

$$x_{IQ}(t) = x(t)e^{-j2\pi\tilde{f}_1t} \quad (6)$$

when $x_I(t)$ and $x_Q(t)$ are defined according to Eq. 3. In writing $x_{IQ}(t)$ this way we have assumed a lowpass filter is applied as shown in Fig. 1. Equation 6 is very helpful for developing frequency-domain intuition about the quadrature receiver.

We may express $x_{IQ}(t)$ in terms of $a(t)$, similar to Eq. 4, as

$$x_{IQ}(t) = a(t)e^{j2\pi(f_1 - \tilde{f}_1)t} \quad (7)$$

4 Practical AM Demodulation: Intermediate Frequencies

AM demodulation is normally performed in two (or more) stages as illustrated in Figure 2. First the received signal is demodulated to a lower, intermediate frequency between that of the carrier and the band containing the original message. This is accomplished by multiplying the received signal with a sine and cosine wave and lowpass filtering. Let the intermediate frequency be f_{IF} and the complex-valued signal be $x_{IF}(t)$. The message is shifted from frequency f_1 to frequency f_{IF} by multiplying the received signal with a complex sinusoid having frequency $-f_{LO}$ where $f_{LO} = f_1 - f_{IF}$. Following Eq. 6, we may write

$$x_{IF}(t) = x_R(t)e^{-j2\pi f_{LO}t} \quad (8)$$

The lowpass filter is not shown in the equation.

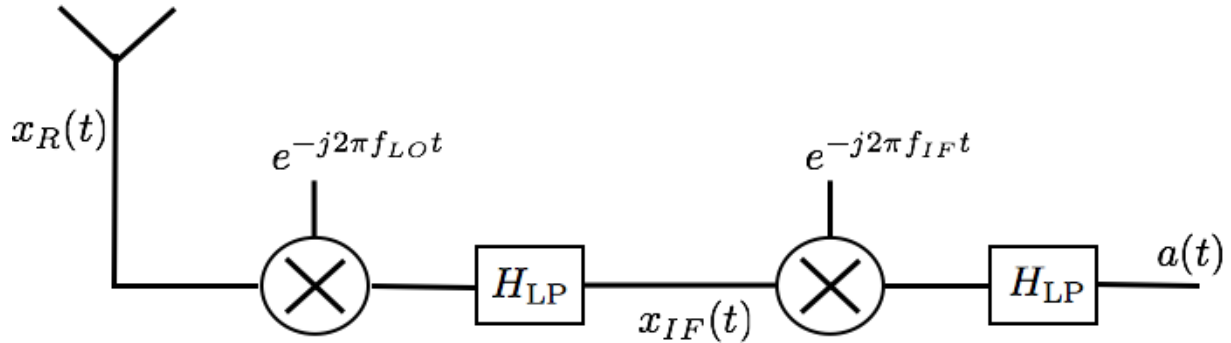


Figure 2: Block diagram of two-stage receiver.

In the second stage, $x_{IF}(t)$ is multiplied by another complex sinusoid of frequency $-f_{IF}$ and lowpass filtered to the message bandwidth. In equations,

$$a(t) = x_{IF}(t)e^{-j2\pi f_{IF}t} \quad (9)$$

Here again the lowpass filter is not shown. The lowpass filter passes the message which is now centered on 0 Hz, but attenuates other signals and noise at higher frequencies. Note that the multiplications by the complex sinusoids shown in Fig. 2 are implemented in an actual system by multiplying with sines and cosines.

A big advantage of two-stage demodulation is that it reduces the cost of the filters and overall processing. Typically the first stage is performed with analog electronics. Analog filters are typically composed of resistors and capacitors. Conceptually they behave similarly to discrete-time filters - they have a frequency response that describes how continuous-time sinusoids of different frequencies are affected by the filter. Old AM radios also performed the second stage with analog electronics, but newer receivers implement the second stage in discrete-time using digital electronics. The data we will use in the lab has undergone a first stage of demodulation using analog electronics. We will perform the second stage digitally using MATLAB.

5 Decibels

Decibels or dB are a logarithmic measure of relative power or amplitude (assumed positive) that are used in situations with very large dynamic range. That is, dB units are used when we are interested in relatively large and small signals. Frequently dB are used as units on plots to show large and small values on the same axis, but they also provide a convenient verbal description for situations with large dynamic range.

We define the size of a positive amplitude A relative to a reference amplitude A_r in dB as

$$20 \log_{10}(A/A_r)$$

Often we assume $A_r = 1$. Since power P is proportional to A^2 , this means the relative power in dB is defined as

$$20 \log_{10}(A/A_r) = 10 \log_{10}(A^2/A_r^2) \quad (10)$$

$$= 10 \log_{10}(P/P_r) \quad (11)$$

This means that each factor of 10 in amplitude increase of a signal adds 20 dB, while each factor of 10 decrease in amplitude subtracts 20 dB. If one signal is 60 dB larger than another, this means its amplitude is a factor of 1000 larger. Similarly, if one signal is 40 dB smaller than another, that is, at -40 dB, then its amplitude is a factor of 100 smaller than the other.

Logarithms convert multiplication/division to addition/subtraction which is what effectively compresses the dynamic range to a smaller scale. Suppose the amplitude of one sinusoid is 1,000 times smaller than that of a sinusoid at a different frequency. It would be very difficult to see the smaller sinusoid on a plot of the spectrum. However, conversion to dB puts the smaller sinusoid 60 dB below the larger one and allows both to be easily visualized. The plot below contrasts these two cases assuming the amplitude of the larger sinusoid is one, the smaller sinusoid is 10^{-3} and the reference amplitude is one when computing dB. The top panel uses a linear scale and the weaker sinusoid is not evident due to limited graphing resolution. The bottom panel uses a dB scale and clearly shows evidence of both sinusoids.

