EE 224: Signals and Systems I
Lab 11 : AM Communication System

If you have a PC-compatible microphone (and headphones), please bring them to lab for the warm-up.
Pre-Lab and Warm-Up: You should read at least the Pre-Lab and Warm-up sections of this lab assignment and go over all exercises in the Pre-Lab section before going to your assigned lab session.

Verification: The Warm-up section of each lab must be completed **during your assigned Lab time** and the steps marked *Instructor Verification* must also be signed off **during the lab time**. One of the laboratory instructors must verify the appropriate steps by signing on the **Instructor Verification** line. When you have completed a step that requires verification, simply demonstrate the step to the TA or instructor. Turn in the completed verification sheet to your TA when you leave the lab.

Lab Report: It is only necessary to turn in a report on Section 3 with graphs and explanations. You are asked to **label** the axes of your plots and include a title for every plot. In order to keep track of plots, include your plot *inlined* within your report. If you are unsure about what is expected, ask the TA who will grade your report.

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1 Introduction & Objective

The goal of this laboratory project is to illustrate the inner workings of a communication system based on AM (Amplitude Modulation). Ideally your lab report should demonstrate that you can produce a functional AM system that operates on a voice signal.

1.1 Think Independently

One objective of your education is to **"learn how to learn"** and how to extend your skills on your own. Thus the second goal in this lab is for you to demonstrate that you can "Think independently." As a result, the lab instructions are minimal, but should be well within your capability as an experienced MATLAB user.

Your written lab report should document that you can produce an interesting demonstration of AM. There are many ways to do this, so formulate your own plan of action.

2 Warmup

In this warm-up you must do four things:

(a) Download the file `generatelpb.m`. The script generates a vector `b`, which is a set of FIR filter coefficients. The FIR filter is a low-pass. For sampling frequency $f_s = 44.1$ kHz, the pass band is $[-2000, 2000]$Hz.

(b) Demonstrate to your TA that you can record a few seconds of your voice at $f_s = 44.1$ kHz. Use the Windows Accessory called *Sound Recorder* and save the signal as a WAV file. When you need the signal in MATLAB, use the function `wavread()`.
Instructor Verification (separate page)

(c) Use `freqz( )` to compute the frequency response of the filter whose coefficients are the vector `b` that you just downloaded. Make a plot of the frequency response over a band of frequencies corresponding to $0 \le f \le 4$ kHz assuming that the sampling rate is $f_s = 44.1$ kHz.
Instructor Verification (separate page)

(d) First use `soundsc( )` to listen to the voice signal that you recorded. Then filter your voice signal through the lowpass filter and play out the result. It should sound similar to the original recording but qualitatively different. How would you describe the difference? Convince your TA that the output has no frequency components above 2 kHz (by using an appropriate spectrogram plot).
Instructor Verification (separate page)

3 Lab Exercises: AM Communication System

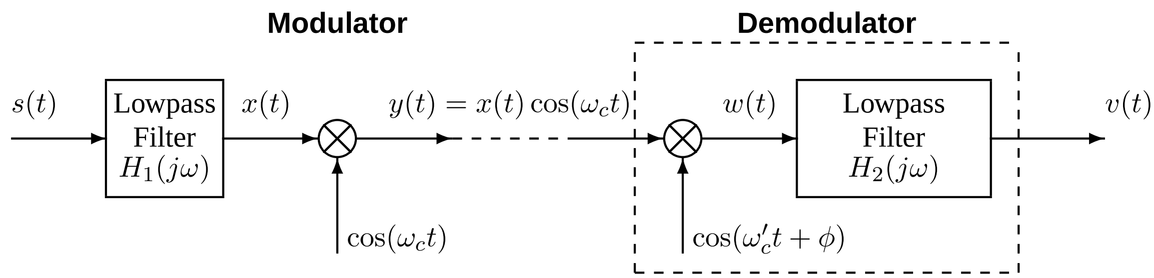

Figure 1: Block diagram of sinusoidal modulation followed by demodulation with variable phase.

In this lab, you must implement a single channel AM communication system (Fig. **1).** Since you will be *simulating* this system with MATLAB, the signal $s(t)$ will be sampled, the filters and modulators will be discrete-time systems, and the output $v(t)$ will be the output of your computer's D/A converter. See Section 3.2. Here are the steps:

1. Make an input signal by recording your voice for a one or two seconds. Use a sampling rate of 44.1 kHz. The best test signal would have lots of vowels. Plot the waveform using plot. Determine the beginning and end of significant speech activity and select only that region for processing.

2. Filter the voice signal with the given FIR lowpass filter.

3. Display the spectrogram of your filtered voice and *estimate* the bandwidth of the speech signal.

4. Make an AM signal by multiplying your voice by a cosine. Use a "carrier frequency" $f_c = \omega_c/(2\pi)$ that is between 8 kHz and 9 kHz.[1]

5. Simulate the demodulator for the AM signal. Write the demodulator as a MATLAB function with input arguments for the frequency and phase. In other words, implement a demodulator that can have a different carrier frequency $f_c' = \omega_c'/(2\pi)$ and non-zero phase ϕ.
Note: The analysis of this demodulator is a homework problem in Chapter 12.
Work this problem before doing the experiment.

[1]To make your choice of frequency unique, pick the carrier frequency as follows: Add the last THREE digits of your SSN to 8,000 and round to the nearest multiple of 100.

McClellan, Schafer, and Yoder, *Signal Processing First*, ISBN 0-13-065562-7.

6. First experiment with the effect of the phase difference. Set the demodulator carrier frequency exactly equal to the modulator carrier frequency. You can hear the effect of phase if you normalize your input signal $s(t)$ so that its maximum magnitude is 1. Then you can listen to the various signals using `sound( )` rather than `soundsc( )`. If you use `sound( )` you will hear amplitude differences because this function does not rescale the signal, however, it does assume that the signal amplitude is less than or equal to 1. Demonstrate how the demodulator output depends on the phase. Show that for one choice of the phase that you get zero output.

7. Next set the phase to zero and make the demodulator carrier frequency 10 Hz higher than the carrier frequency of the modulator. Listen to the demodulated signal. How do you characterize the output?

8. Demonstrate your working system to your lab TA.

Make spectrograms at the appropriate points to show what is happening to the signal in the frequency domain, e.g., show the spectrogram at each point in the demodulator.

3.1 Explanations

Explain using Fourier transforms, mathematics and sketches how your AM system works One aspect of this is to explain why "phase matters" in the receiver's mixer, but it is also important to draw sketches of the Fourier transform to show where the signal's spectrum lies at any point in the system. Make a spectrogram at certain key places in the system to explain what is going on. You don't need every one, but you should use the Fourier knowledge that you gained in the previous part to decide which ones are useful. In particular, make a spectrogram of the AM signal $y(t)$. Point out the upper and lower sidebands in the spectrogram. Since the theory shows that the lower sideband comes from the "negative frequencies" in $x(t)$, this is in some sense a demonstration that those negative frequencies do indeed exist.

3.2 Note on Simulation Frequency

A true AM system is a continuous-time (or analog) system. However, in this lab we are using MATLAB to simulate the analog system. The simulation rate is 44.1kHz. Therefore, every signal in the simulation must be sampled at $f_s = 44.1$ kHz, and every digital filter must be designed for 44.1 kHz. Also, the sampling frequency places a limit on how high we can make the carrier frequency since modulation shifts the input spectrum to higher frequencies, we might incur aliasing if we shift too far. Also, in the demodulation process, we generate even higher frequencies. This is why we used a bandwidth of 2 kHz and a carrier frequency around 8 kHz.

Lab 11
INSTRUCTOR VERIFICATION SHEET

For each verification, be prepared to explain your answer and respond to other related questions that the lab TA's or professors might ask. Turn this page in at the end of your lab period.

Name: ________________________________ Date of Lab: __________

Part **??**(a): Record your voice and show a spectrogram in MATLAB of the resulting signal.

Verified:____________________ Date/Time:__________

Part **??**(b): Use `freqz( )` to compute the frequency response of the filter whose coefficients are the vector `b` that you downloaded. Make a plot of the frequency response over a band of frequencies corresponding to $0 \leq f \leq 4000$ kHz assuming that the sampling rate is $f_s = 44.1$ kHz. Determine the stopband region from the plot.

Verified:____________________ Date/Time:__________

Part **??**(c): Filter your voice signal. Convince your TA that it has no components above 2 kHz (approximately).

Verified:____________________ Date/Time:__________

Evaluation: Demonstrate your working system to you TA. Explain how it works. Listen to the output signal for the maximum output.

Verified:____________________ Date/Time:__________

Verify the output when the phase of the demodulator is changed to produce zero and demonstrate the effect of a mismatch between the modulator and demodulator carrier frequencies.

Verified:____________________ Date/Time:__________

McClellan, Schafer, and Yoder, *Signal Processing First*, ISBN 0-13-065562-7.
Prentice Hall, Upper Saddle River, NJ 07458.