GEORGIA INSTITUTE OF TECHNOLOGY

SCHOOL of ELECTRICAL and COMPUTER ENGINEERING

ECE 2025 Spring 2004
Lab #11: AM Communication System

Date: 5–8 April 2004

You should read the Pre-Lab section of the lab and do all the exercises in the Pre-Lab section before your assigned lab time. You **MUST** complete the online Pre-Post-Lab exercise on Web-CT at the beginning of your scheduled lab session. You can use MATLAB and also consult your lab report or any notes you might have, but you cannot discuss the exercises with any other students. You will have approximately 20 minutes at the beginning of your lab session to complete the online Pre-Post-Lab exercise. The Pre-Post-Lab exercise for this lab includes some questions about concepts from the previous Lab report as well as questions on the Pre-Lab section of this lab.

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**. After completing the warm-up section, turn in the verification sheet to your TA.

Forgeries and plagiarism are a violation of the honor code and will be referred to the Dean of Students for disciplinary action. You are allowed to discuss lab exercises with other students and you are allowed to consult old lab reports but the submitted work should be original and it should be your own work.

The lab report for this lab will be an **Informal Lab Report,** but it will probably be **LONG.** Include derivations as well as plots of the spectrogram to illustrate the spectral content of the different AM channels in part 4.

The report will **due during the week of 19-April** (the last week of the semester).

1 Introduction & Objective

The goal of this laboratory project is to illustrate the inner workings of a communication system based on AM (Amplitude Modulation). Once a working AM system is programmed, it is possible to use the spectrogram to see how the signal information is being placed into different "channels" in the frequency domain. This also provides a link to the Fourier transform and a mathematical description of the frequency domain behavior of the AM system. Two versions of AM will be investigated: DSBAM-SC and QAM. These acronyms stand for double sideband AM, suppressed carrier and for quadrature amplitude modulation. The lab project will involve the simulation of a receiver for DSB-AM, and the extension of that system to quadrature modulation where two signals can be broadcast simultaneously on one channel by exploiting the real and imaginary parts of the complex exponential. In communication systems, the real and imaginary parts are usually called the *in-phase* and *quadrature* channels, or simply the I and Q channels.
Note: It would be useful to read Chapter 12 in *SP-First* for a discussion of AM communication systems.

1.1 Simulation of a Continuous-Time System

Since AM Radio is an analog communication system, we cannot implement it on a computer (i.e., in MATLAB). However, we can carry out a *simulation* of an actual system. This is done by using a very high sampling rate for all the signals so that absolutely no aliasing will occur. Then we can be assured that the digital signals are being processed exactly the same way that the analog signals would be filtered and modulated. If we denote the sampling interval for the simulation as T_{sim}, then we are saying that as $T_{\text{sim}} \to 0$ the digital simulation will behave exactly like the analog hardware.

When writing MATLAB programs, we would prefer to talk about the sampling rate of the system, which in this case will be $f_s = 1/T_{\text{sim}}$. If $T_{\text{sim}} \to 0$, then $f_s \to \infty$, so the higher the sampling rate, the more memory we need for the simulation. To keep the MATLAB vectors down to a reasonable size, we will use 44.1 kHz for the sampling rate and use relatively low frequencies for the AM carrier frequencies. At 44.1 kHz we will also be able to "listen to the simulation," because 44.1 kHz is sampling rate for audio CDs.

2 PreLab

Since this lab involves AM communication systems, review the material in Chapter 12 of *SP-First* so that you understand how the FDM system of Fig. 1 works.

2.1 Frequency shifting

Demonstrate that you can create a frequency-shifted signal. For the test signal use a sinusoidally modulated chirp:

$$x_1(t) = \cos(2\pi(800)t + 30\cos(2\pi(20)t))$$

(a) Determine the instantaneous frequency of $x_1(t)$ by doing a simple mathematical derivation. What is the maximum (and minimum) frequency in $x_1(t)$?

(b) Write some MATLAB code to create $x_1(t)$ with a duration of 0.6 seconds, then use $x_1(t)$ to modulate a carrier signal with a frequency of $\omega_c = 2\pi(4400)$ rad/sec. Use a simulation sampling rate of 44.1 kHz.

(c) Make a spectrogram of both $x_1(t)$ and the modulated signal to verify that the modulated signal has the desired frequency location. **Since the sampling rate for the simulation is so high, you should change the parameters `NFFT` and `window` in `specgram.m` to be longer than their default values.** Good choices would be 1024 and 512, so use

```
specgram(yy, 1024, fsim, 512)
```
or
```
plotspec(yy+j*1e-12, fsim, 1024)
```

Another consequence of the high simulation sampling rate is that you will have to zoom to see the (relatively small) frequency region of interest near 4400 Hz.
Note: the MATLAB `plotspec.m` command can make a plot of the *negative frequency region*, if you add a tiny imaginary part to `yy`.

3 Warmup

In this warm-up you must test out the pieces of a simple AM system that can operate on one channel. Since we are trying to study an *analog system*, we can only do a *simulation*. We know that any simulation requires a sampling rate for the simulation—in this case, we will use $f_s = 44,100$ Hz. This makes it possible to do the simulation in MATLAB, but keep in mind that real AM transmission systems operate with carrier frequencies near 1 MHz (10^6 Hz). Simulating a system operating at those rates would require a "simulation sampling rate" in the range of 5 to 10 MHz.

3.1 Modulation

Demonstrate that you can create a modulated signal. For the test signal use a sinusoidally modulated chirp:

$$x_1(t) = \cos(2\pi(800)t + 30\cos(2\pi(20)t))$$

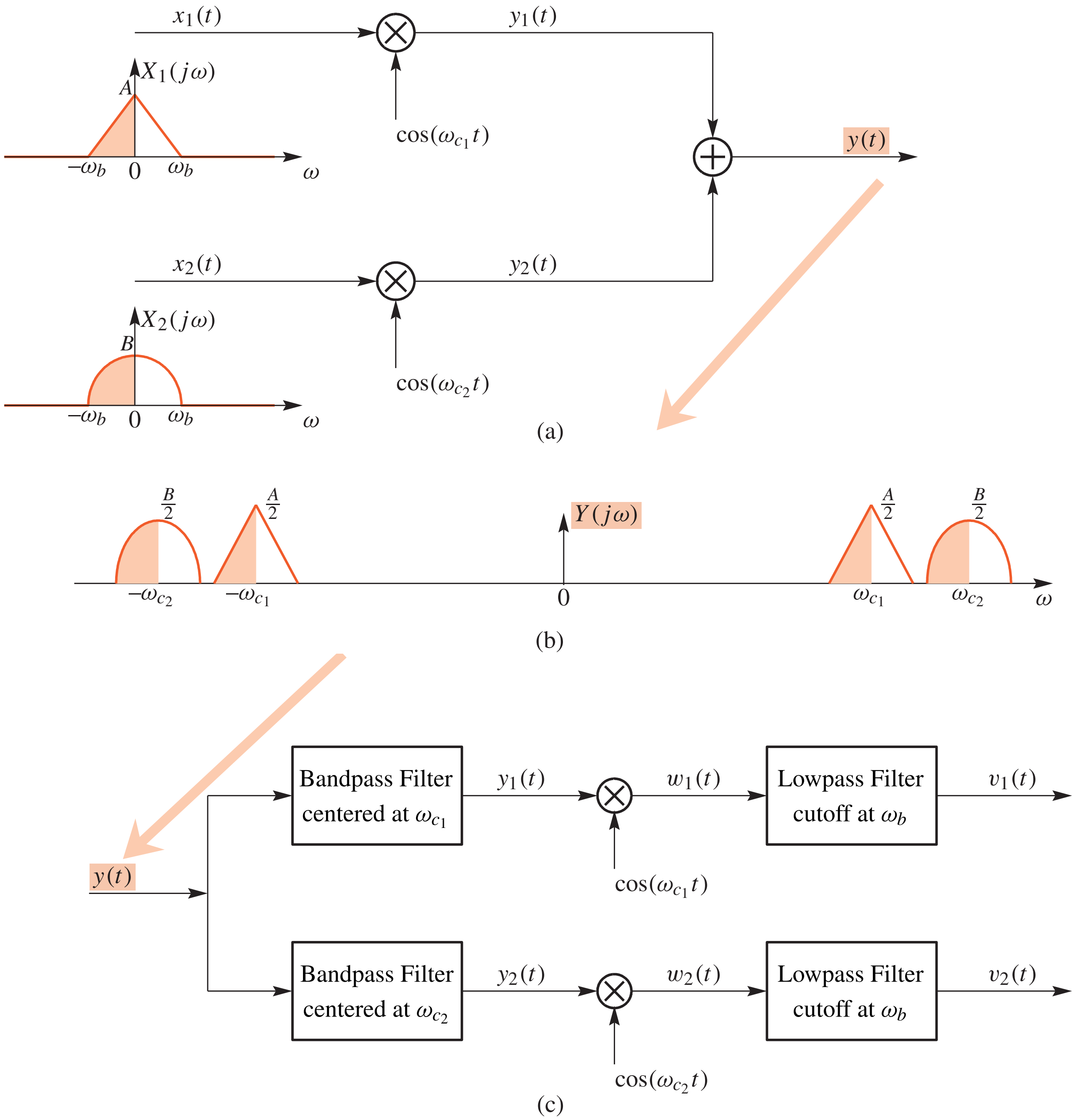

Figure 1: Block diagram of a Frequency-Division Multiplexing (FDM) communication system.

(a) Write some MATLAB code to create $x_1(t)$ with a duration of 0.6 seconds, then use $x_1(t)$ to modulate a carrier signal with a frequency of $\omega_c = 2\pi(4400)$ rad/sec.

(b) Make a spectrogram of both $x_1(t)$ and the modulated signal to verify that the modulated signal has the desired frequency location. Include positive and negative frequencies in the spectrogram. Explain to your TA where you see the frequency content of $x_1(t)\cos(\omega_c t)$ and why it is correct.

Instructor Verification (separate page)

3.2 Analog Filter Simulation

The demodulator will contain an analog bandpass filter to separate out the channel of interest.[1] Use your MATLAB function called `myBPF` to perform this "simulation" of an "almost-ideal" analog filter.[2] Since the filter specs will be given in Hz, the filter design will have to convert the cutoff frequencies by the simulation sampling rate `fsim` in Hz for the analog system. The output from `hh = myBPF( )` is the impulse response of the analog filter, but, of course, the impulse response is being sampled at the simulation rate, `fsim`.

Normally, the frequency response of the analog filter should be obtained by "taking the Fourier transform" of the impulse response. But, this is where things get a little confusing, because the simulation of the analog filter is actually a digital filter. To get the frequency response you should follow a three step procedure:

i. Recognize that the impulse response is a discrete-time signal, $h[n]$.

ii. Use `freqz()` to get the frequency response of the digital filter that is doing the simulation. If you made this plot, you would use a horizontal axis whose variable is $\hat{\omega}$ from $-\pi$ to π.

iii. Convert the frequency axis from $\hat{\omega}$ to ω. This requires that you use the simulation sampling frequency to get the correct units for the frequency axis.[3] A convenient way to figure out this rescaling is to remember that $\hat{\omega} = \pi$ corresponds to half the (simulation) sampling frequency.

(a) The task in this part is to design an analog filter whose passband extends from 3000 Hz to 5000 Hz; and whose stop bands are $f < 2000$ Hz and $f > 6000$ Hz. Determine the filter length (L) that will be required.

(b) Then make a plot of the frequency response versus ω in rad/sec to verify that the passband is in the correct place.

(c) For this system we will measure the passband width at the 0.995 and 1.005 points for the Hamming-sinc filters. Thus, you should examine the location of the passband and stopband of the simulated filter using the plot from part (b). If you measure the location of the passband as the region where the filter gain is above 0.995, verify that the passband width is correct. Likewise, if you measure the stopband edges as the points where the filter gain is below 0.005, verify that the stopband edges are correct. The distance between passband and stopband edges is called the "transition zone" of the BPF. It must be taken into account when trying to pack AM channels close together, because we cannot build ideal BPFs to demultiplex the AM channels.

Instructor Verification (separate page)

[1] This bandpass filter is not essential, but we'll use it in this lab.

[2] Since it is a simulation, the filter is actually an FIR digital filter.

[3] you will be creating what is called the "effective analog filter."

3.3 Demodulation: the Mixer

The demodulation section consists of two parts: a cosine multiplier, usually called a *mixer,* followed by a lowpass filter. In this part, you should examine the spectrum of the signal at the output of the mixer.

(a) Take the signal from Section 3.1(b) and multiply by a cosine of the correct frequency to do the mixing.

(b) Plot the spectrogram of the signal at the output of the mixer. Include positive and negative frequencies. Explain all the spectral components that you see in the spectrogram. Zoom in to see details if necessary.

Instructor Verification (separate page)

As a final comment, the lowpass filter needed when implementing the AM demodulator (in the next section) can be designed with your function `myLPF.m`.

3.4 MATLAB **Demo**

In MATLAB's signal processing toolbox, there is a demo called `moddemo` which illustrates the AM system, as well as some other modulation schemes. The demo can show the time waveforms and also the spectrograms in a DSBAM-TC system. It works fine for the speech input, but not well for the sine input.

4 AM Communication System

In this part of the lab project you must do the entire implementation of a two-channel AM communication system. The transmitter and receiver sections should be written as separate MATLAB M-files. The assigned channel frequencies are 3200 Hz and 8500 Hz, and each channel will be 5000 Hz wide, i.e., $\pm$2500 Hz, after taking into account the realistic bandpass filters that will be used in the receiver.

4.1 Transmitter: `mymodulator.m`

Write a MATLAB function, called `mymodulator.m`, that will take two baseband signals[4] $x_1(t)$ and $x_2(t)$ and produce the modulated signal which is called $y(t)$ in Fig. 1.

(a) Provide the MATLAB code for `mymodulator.m` in your report.

(b) For testing, use the signal from the warm-up Section 3.1(b) for $x_1(t)$ but change its duration to 1.25 secs. For the second signal, $x_2(t)$, use a constant-frequency sinusoid at $\omega = 2\pi(1000)$ rad/sec. with the same duration. Plot a spectrogram of the modulated signal $y(t)$ to show that the signals are in the correct place in the frequency domain.

4.2 Receiver: `mydemod.m`

In this section, you need to implement the *synchronous* AM receiver, which was shown in Fig. 1.

(a) Write a MATLAB function that will "tune" to any AM channel, like the dial on a radio. Call the function

```
vv = mydemod( yy, fstation, mixerphase )
```

where the arguments are `yy`, the signal into the demodulator, `fstation`, the frequency of the station you want to hear, and `mixerphase`, the phase of the multiplier sinusoid in the mixer. The output `vv` is the demodulated signal. Having these two parameters will facilitate the experiment in Section 4.3. Write the function `mydemod(yy,fstation,mixerphase)` with the following capabilities:

[4]The term *baseband* refers to the fact that a signal such as speech or music has a Fourier transform that is centered around $\omega = 0$.

- Design two BPFs to accommodate the desired channel bandwidth (± 2500 Hz) for the two given stations (3200 Hz and 8500 Hz). Use the Hamming-sinc filter design and obey the following constraint: the neighboring channel should be in the stopband of the filter so that it doesn't interfere, i.e., it should be less than 0.5%. This is the usual spec for the Hamming-sinc filters, and it will determine the length of the filter that has the correct stopband edges and passband edges.
- When the receiver tunes to a new station, the bandpass filter must move in frequency. Thus, your M-file should call the BPF design function "on the fly" using the frequency of the station.
- Design the lowpass filter (after the mixer) with `myLPF.m` so that any other interfering spectral components are removed. Use the same LPF for both channels. Determine the minimum filter length and state which frequency you used for the cutoff frequency in `myLPF` in order to obtain the desired passband and stopband edges of the LPF.

(b) Plot the frequency response of one of the BPFs; plot both magnitude and phase. Since the passband is the most interesting part, plot only the region near the passband (and the beginning of the stopbands) to show that your filter design meets the stopband and passband specs.

(c) Since the filter doing the simulation is a digital filter, the implementation of the filter for processing signals must be done in the "index domain," n. Therefore, use the appropriate MATLAB function for the digital filtering. For your lab report, include a listing of your MATLAB program that simulates the receiver.

(d) Based on your BPF design in part (a), explain how the non-ideal characteristic of the analog BPF limits the channel bandwidth to a value that is less than the maximum possible channel bandwidth, which is the separation of the carrier frequencies.

4.3 Testing and Listening

Some signals have been provided for testing your system.

(a) First of all, verify that your system works on the synthetic signals from Section 4.1(b). This can be done with a "listening test" because the simulation rate is 44.1 kHz. Listen to the input signal and then the demodulated output signal to verify that they sound identical. Both the chirp and the sinusoid should be well within the bandwidth of the DSBAM system, so there should be virtually no loss of signal quality. Make plots of the input and output signals, and then measure the amplitudes of the input and output signals to verify that they differ by a factor of two (as predicted by the theory when using filters whose passband is one).

(b) Now test the entire AM mod-demod system with two "real" signals from the file `harp2k.mat`. The sampling rate for both is 44.1 kHz. The signals are called `xx1` and `xx2` when loaded into MATLAB with the load command. Run `mymodulator()` followed by `mydemod()` to verify that the system works correctly and that you can identify the two signals at the output.
Note: you do not have to turn in any plots for this part; just write a brief description of the two "real" signals and what you heard.

(c) Now try different values for the phase of the mixing sinusoid in an attempt to make the output signals zero; remember that you've already been using $\psi = 0$. Record the two values of ψ for which the output is exactly zero (one value for each channel).
Note: This is not as simple as it seems, the reason being that the bandpass filters change the phase of the received signal prior to the multiplication by the sinusoid in the mixer. Although this phase change can be predicted from the length of the FIR bandpass filter, it is a frequency-dependent phase.

(d) Explain the observations from the previous part by drawing sketches (by hand) of the Fourier transform at the modulator output and mixer output. Do this for the 8500 Hz channel only. If you make a spectrogram, it will help in drawing the sketches of the Fourier transform, but note the scale on the spectrogram because functions like `specgram()` will automatically scale to the maximum value.

4.4 QAM: Quadrature AM System

In a quadrature AM system, the transmitter uses two carrier signals that are "90° out of phase" when doing the modulation. Usually the carrier signals are $\cos(\omega_c t)$ and $\sin(\omega_c t)$, but another general choice would be $\cos(\omega_c t + \phi)$ and $\cos(\omega_c t + \phi + \pi/2)$. As illustrated in the previous parts, the receiver works correctly when its mixing cosine is "in-phase" with the transmitter. Putting these two ideas together, it is possible to *double the capacity* of the AM transmission system by *exploiting the phase dependence* of the receiver.

For this part, you should use your function `mydemod()` in a "trial-and-error" mode to find all the signals buried in the "mystery transmission" called `lab11s04.mat` which can be downloaded from Web-CT. The mystery signal is a few secs. in duration and is sampled at the simulation rate of 44.1 kHz. The transmit channels are still 3200 Hz and 8500 Hz, but the transmitter uses a *quadrature* modulation scheme.

(a) Process the "mystery transmission" multiple times to see how many signals you can extract from it. Use your function `mydemod()`, but vary the phase as you did in Section 4.3(c). Listen to the outputs each time. Describe in detail what you heard and how you extracted the different signals. How many did you find? Include a list of the descriptions of the extracted signals in your lab report, and give the settings that you used for `mydemod()` in each case.

(b) Explain using Fourier transforms, mathematics and sketches how the system is able to send more than two signals over the two AM channels. One aspect of this discussion is to explain why "phase matters" in the receiver's mixer, but it is more important to figure out and explain how the transmitter is forming the signals. Give a formula for the time signal that is the output of the transmitter, and also its Fourier transform. Write the formula as a summation in terms of all the input signals, $x_i(t)$ and the sinusoids at the two carrier frequencies.

(c) 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 decide which ones are useful. Use the Fourier knowledge that you gained in the previous part to pick the key spectrograms.

4.5 Summarize with a Concept Map

For the Summary section of your lab report, draw a concept map that contains at least five concepts that were used during this lab. Since experts use many links between concepts, try to produce a map that has a high "link to node" ratio. Use the Concept Navigation Tool (***CNT***) to produce the map.

Since you are now experienced at making concept maps, no list of concepts is given and you should try to discover the list of possible concepts on your own. One approach might be to make a list of concepts before doing the map, and then determine how to link the concepts together.

Please print out your concept map for your lab report, but also ***save it to the web*** by using that option in ***CNT.*** Include an identifier that refers to Lab #11.

Lab #11
ECE-2025
Spring-2004
INSTRUCTOR VERIFICATION PAGE

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 3.1: Show the spectrogram (positive and negative frequency regions) of an amplitude modulated chirp signal. Explain to your TA the expected range for the frequency content of $x_1(t)\cos(\omega_c t)$.

Verified:____________________ Date/Time:__________

Part 3.2: Design an analog filter with a specified passband, and make a plot of its frequency response (magnitude) versus ω in rad/sec. Determine the stopband region from the plot.

Verified:____________________ Date/Time:__________

Part 3.3: Determine the output of the mixer in an AM demodulator. Show the spectrogram with positive and negative frequencies. Explain all the spectral components that you see in the spectrogram.

Verified:____________________ Date/Time:__________