GEORGIA INSTITUTE OF TECHNOLOGY
SCHOOL of ELECTRICAL and COMPUTER ENGINEERING

ECE 2025 Fall 2004
Lab #11: AM Communication System

Date: 9–15 Nov 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 you cannot give or receive written material or electronic files. Your submitted work should be original and it should be your own work.

The lab report for this week will be an **Informal Lab Report.** It is only necessary to turn in Section 4 as this week's lab report. The report will be due **on Monday, 22-Nov in Lecture (before noon),** or if you have lab on that Monday afternoon you can turn it in at the beginning of your lab time.

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1 Introduction & Objective

The goal of this laboratory project is to illustrate frequency shifting as used in communication systems based on AM (Amplitude Modulation). It is possible to use the spectrogram to see how the signal's spectral information can be moved 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. The lab project will involve the simulation of two frequency shifters: a "speech scrambler" that encodes a signal by flipping its spectrum, and a standard AM double sideband system.
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 the sampling rate for audio CDs.

2 PreLab

Since this lab involves AM communication systems, review the material in Chapter 12 of *SP-First* to gain an understanding of 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 MATLAB code to create $x_1(t)$ with a duration of 0.6 secs, then use $x_1(t)$ to modulate a carrier signal with a frequency of $\omega_c = 2\pi(5000)$ 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 and sidebands. **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 5000 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, which makes it possible to do the simulation in MATLAB. However, 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" greater than 5 or 10 MHz.

3.1 Modulation

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

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

(a) Write MATLAB code to create $x_1(t)$ with a duration of 0.6 secs, and then use $x_1(t)$ to modulate a carrier signal with a frequency of $\omega_c = 2\pi(5000)$ 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 and sidebands. 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)

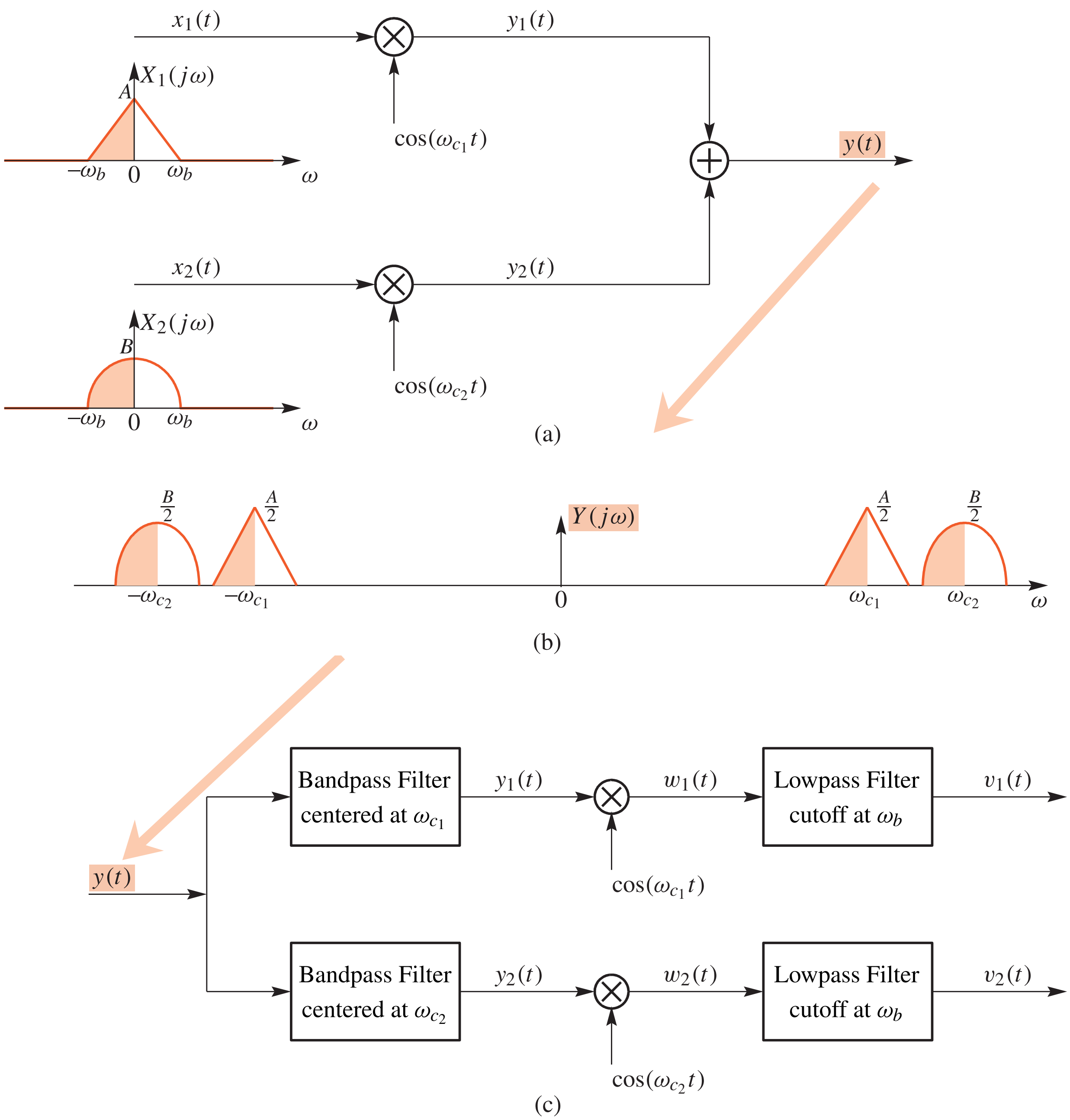


Figure 1: Block diagram of a Frequency-Division Multiplexing (FDM) communication system.

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 `VHBPF` 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 = VHBPF( )` is the impulse response of the analog filter, but, keep in mind that 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 3500 Hz to 6500 Hz; and whose stop bands are $f < 3000$ Hz and $f > 7000$ 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 0.9925 and 1.0075 points for the vonHann-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 0.9925, verify that the passband width is correct. Likewise, if you measure the stopband edges as the points where the filter gain is below 0.0075, 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)

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)

[1] This bandpass filter is not essential, but we'll use it in this lab.
[2] Since we are doing a simulation, the filter is actually an FIR digital filter.
[3] You will be creating what is called the "effective analog filter."

As a final comment, the lowpass filter needed when implementing the AM demodulator (in the next section) can be designed with your function `VHLPF.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 Transmitting and Receiving Scrambled Speech

In this part of the lab project you must do the entire implementation of an AM communication system that transmits scrambled speech. The scrambling is done to hide the content, but it's a very simple method of secrecy coding that's not very hard to break. The assigned channel frequency is $f_c = 9000$ Hz, and each channel will be 8000 Hz wide, i.e., ± 4000 Hz. The system consists of four major blocks as shown in Fig. 2.

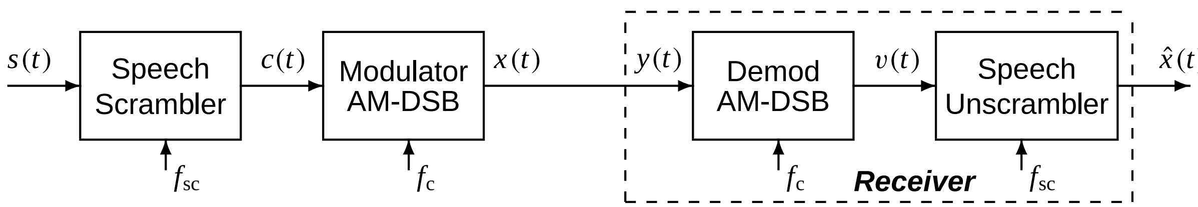


Figure 2: Block diagram of the AM communication system for send scrambled speech. For this lab, the channel frequency will be $f_c = 9000$ Hz, and the scrambling frequency $f_{sc} = 4000$ Hz.

4.1 Speech Scrambler: `scrambler.m`

Write a MATLAB function, called `scrambler.m`, that will take a baseband speech signal[4] $s(t)$ and produce the scrambled signal which is called $c(t)$ in Fig. 2. The process involves two steps: multiply the speech signal by a sinusoid, $\cos(2\pi f_{sc} t)$, and then filter with a LPF whose passband edge is at $0.925 f_{sc}$ Hz, where f_{sc} is called the *scrambling frequency*. The LPF will have to be quite sharp because its stopband must reject a portion of the spectral content that is just a little bit higher than f_{sc}; so use $1.075 f_{sc}$ for the stopband edge.

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

(b) For testing, use the following chirp signal

$$s(t) = \cos(3000\pi t^2) \qquad \text{for } 0 \le t \le 0.9$$

Listen to the original and scrambled signals; describe the difference.

(c) Plot spectrograms of the original chirp signal and the scrambled signal to show what is happening in the frequency domain. Notice that the output of the scrambler should have little or no frequency content for $f > f_{sc}$ Hz.

4.2 Speech Unscrambler: `unscramble.m`

Write a MATLAB function, called `unscramble.m`, that will recover a signal from its scrambled version. Prove that it works by processing the test case in the previous section.

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

4.2.1 Related Homework Problem

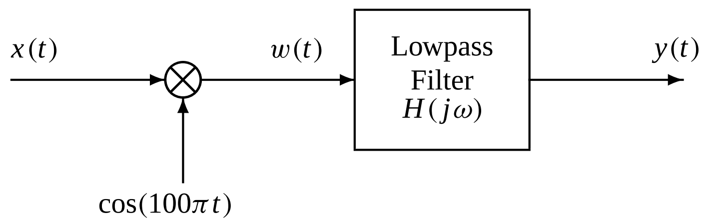


Figure 3: A spectrum scrambler consists of a cosine multiplier followed by a lowpass filter.

In the modulation/filtering system of Fig. 3, assume that the input signal $x(t)$ has a bandlimited Fourier transform $X(j\omega)$ as depicted in Fig. 4:

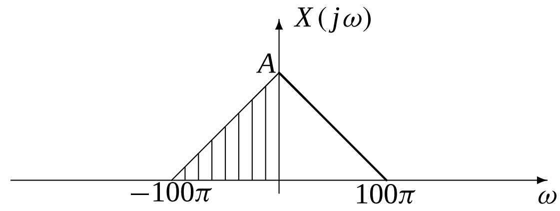
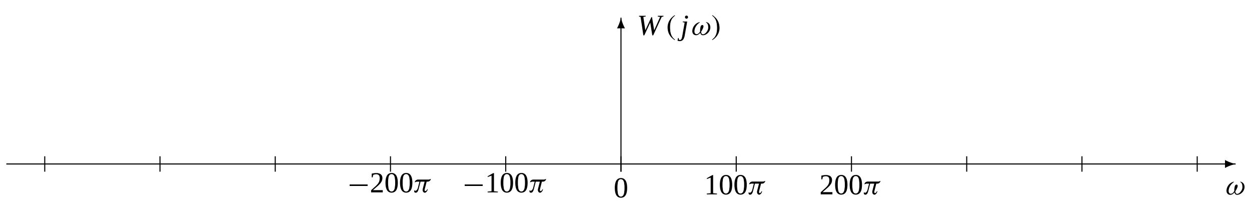


Figure 4: Input spectrum for analyzing the scrambler.

(a) First, give the general equation that expresses $W(j\omega)$, the Fourier transform of $w(t) = x(t)\cos(100\pi t)$, in terms of $X(j\omega)$.

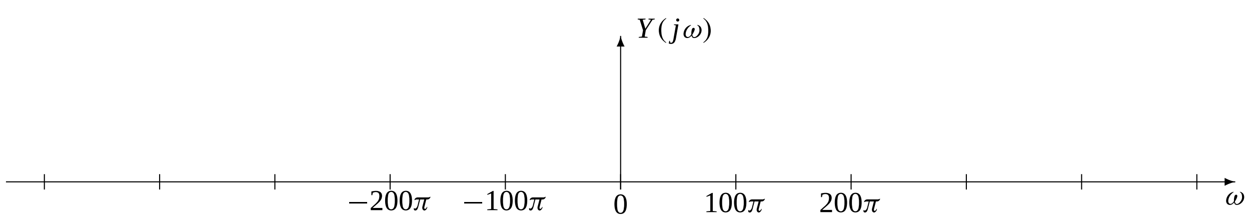
(b) Now **carefully** plot the Fourier transform $W(j\omega)$ for the specific input $x(t)$ whose Fourier transform $X(j\omega)$ is given in Fig. 4. *Note that the negative frequency portion of the Fourier transform $X(j\omega)$ is shaded. Mark the corresponding region or regions in your plot of $W(j\omega)$.*



(c) The frequency response of the ideal lowpass filter is

$$H(j\omega) = \begin{cases} 1 & |\omega| \le 100\pi \\ 0 & |\omega| > 100\pi \end{cases}$$

Plot the Fourier transform $Y(j\omega)$ below for the $X(j\omega)$ given in Fig. 4. *Carefully label both amplitudes and frequencies and also shade the region corresponding to the negative frequencies of the input.*



(d) Define a system consisting of multipliers and filters that will recover $x(t)$ from $y(t)$.

4.3 Transmitter: `modscram.m`

In this section, you need to implement the combined modulator and scrambler for the AM system shown in Figs. 2 and 1.

(a) Provide the MATLAB code for your function.

(b) Show spectrograms from testing on the chirp test case of Section 4.1.

(c) Show spectrograms from testing on a speech signal, `sptest.wav`.

4.4 Receiver: `demodunscram.m`

In this section, you need to implement the combined demodulator and unscrambler for the AM system shown in Figs. 1 and 2.

(a) Provide the MATLAB code for your function.

- Design the BPF to accommodate the desired channel bandwidth ($\pm$4000 Hz) for the given station at 9000 Hz. Use the vonHann-sinc filter design and obey the following constraint: the stopband of the bandpass filter should be at $\pm$5000 Hz. This spec for the vonHann-sinc filters will determine the length of the filter that has the correct stopband edges and passband edges.
- Plot the frequency response of the BPF; 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.
- Design the lowpass filter (after the mixer) with `VHLPF.m` so that any other interfering spectral components are removed. Determine the minimum filter length and state which frequency you used for the cutoff frequency in `VHLPF` in order to obtain the desired passband and stopband edges of the LPF.
- 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.

(b) Show spectrograms from testing on the chirp test case of Section 4.1.

(c) Show spectrograms from testing on a speech signal, `sptest.wav`.

Extra: There is a way to combine the demodulator and unscrambler into one unified demodulator, and thereby eliminate some of the operations. If you are able to figure this out, then do the implementation of `demodunscram.m` using the more efficient structure.

4.5 Mystery Signal

For this part, you should use your function `demodunscram()` to find all the signals buried in the "mystery transmission" called `spmyst.wav` in `lab11f04.zip` which can be downloaded from Web-CT.

(a) Process the "mystery transmission" to determine what is being said in the speech signal.

(b) Explain using Fourier transforms, mathematics and sketches how the system handles the scrambling.

(c) Make a spectrogram at certain key places in the demodulator/unscrambler to explain what is going on. Use the Fourier knowledge that you gained in the previous part to pick the key spectrograms.

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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:__________