EE 451 – LAB 7

Amplitude Modulation and Demodulation

This laboratory is divided into two parts. In the first part you will learn how to design an AM modulator using the C6713 DSK environment, and in the second part you will implement the demodulation of an AM signal.

Introduction

A very common method of transmitting information is through Amplitude Modulation (AM) of a message. The transmission of a low-frequency signal over a channel requires a process to transform the signal to a high-frequency. At the receiver end, the signal is demodulated and filtered to extract the low-frequency signal. There are four major types of modulation of signals: amplitude modulation, frequency modulation, phase modulation, and pulse amplitude modulation.

In amplitude modulation a transmitted signal includes the carrier signal. An AM signal has the following form

$$s(t) = A_c[1 + k_a m(t)]\cos(w_c t)$$

Where A_c is the amplitude of the carrier, k_a is the amplitude sensitivity of the modulator, m(t) is the message, and $\cos(w_c t)$ is the carrier. In standard AM modulation, $1+k_a m(t) \geq 0$ for all t, so the message can be recovered from the envelope to within a scale factor and constant offset.

Two methods can be used for envelope detection and are particularly suited for DSP implementation: the square-law and Hilbert transform detection. The square-law demodulation method consists of squaring the signal, passing it though a lowpass filter, obtaining the square root and finally, scaling and removing an offset. The first step can be expressed as

$$s^2(t) = A^2{}_c[1 + k_a m(t)]^2 \cos^2(w_c t) = 0.5A^2{}_c[1 + k_a m(t)]^2 + 0.5A^2{}_c[1 + k_a m(t)]^2 \cos(2w_c t)$$

The right-hand side of the equation above consists of a lowpass signal whose cutoff frequency has been modified to $2\omega_s$ by the squaring operation, and a second term that has a spectrum centered at $\pm 2\omega_c$. For positive frequencies, the spectrum of the signal has a range of $2\omega_c \pm 2\omega_s$. The spectra of these two terms must not overlap

$$\omega_c > 2\omega_s$$

The lowpass filter has a cutoff frequency of $2\omega_s$ and its output is $0.5A^2{}_c[1 + k_a m(t)]^2$. Square-root of this signal results in an output signal that is proportional to m(t) with a DC offset, which can be removed by a highpass filter.

Assuming that m(t) does not have spectral components below 50Hz, the following highpass filter can be used to remove the DC offset

$$G(z) = \frac{1+c}{2}\frac{1-z^{-1}}{1-cz^{-1}}$$

Where c is a constant slightly less than 1 so that m(t) does not get distorted.

The Prelab

For this lab you will need to create an AM modulator. Use f_s =48 kHz. For your message, have the program generate a sinusoidal signal. You should be able to vary the message frequency dynamically as your code is running.

Amplitude Modulation

1. Why do we modulate?
2. How would you go about generating your message signal. Write all the equations that you will need.
3. Write a MATLAB code to generate your AM signal.
4. Sketch the magnitude of the frequency response.

Amplitude Demodulation

1. On a magnitude spectrum plot identify: (a) where is the passband, (b) where is the stopband, (c) where is the transition band. Also, explain what is meant by passband ripple and stopband ripple.
2. Use MATLAB to create an elliptic filter with the following specifications: R_p = 0.1 dB and R_s = 50 dB. What is the order of the filter?
3. Use MATLAB to express the filter as a cascade of second-order systems. You can use *fdatool*.

The Lab

1. Write a program that implements the amplitude modulated signal you implemented in the prelab. Start CCS and begin a new project. Create and add a configuration file to the project. Select File → New → DSP/BIOS Configuration. We will use a HWI to (i) read a new input sample from the codec, (ii) generate an AM signal, and (iv) output to the codec.

2. Connect a function generator to the board and vary the frequency and verify that the program can generate an AM signal through the LINE OUT.

3. Write a program that implements the demodulation of a signal. You will need another DSP board to generate the AM signal. Run the code that generates an AM signal on one board and connect it to the second board that is running your demodulation code.