Amplitude Modulation and Demodulation

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.

For this project 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. You may use *fdatool* from MATLAB to design a filter with the following specifications: R_p = 0.1 dB and R_s = 50 dB, and express it as a cascade of second-order systems. Connect a function generator to the board and vary the frequency and verify that the program can generate an AM signal. You will need another DSP board to generate the demodulator.