University of Rhode Island Department of Electrical and Computer Engineering
ELE 435: Communication Systems

Envelope and Envelope Recovery Lab Report and Grading Format

1 Report Format

1. Briefly explain what you did in the lab
2. Show the Results/plots, you can use excel or a snap shot or draw it by hand. T6, T7, T9, T10, T12
3. Matlab Code (print the code with the report and also submit the .m file on Sakai, please save it as AMlabb_yourname.m)
   (a) Modulation: Generation of an AM signal
      - Generate discrete-time version of the message signals $m_1(t), m_2(t)$, given by $m_1(nT_s) = sin(2\pi f_1 nT_s)$, and $m_2(nT_s) = sin(2\pi f_2 nT_s)$, where $f_1 = 100$ Hz, $f_2 = 200$ Hz and T_s is the sampling interval $f_s = \frac{1}{T_s}$.
      - Generate a carrier signal with $f_c = 5KHz$.
      - Generate an AM signal with $A_c = 10$ and $f_c = 5KHz$, $s(nT_s) = A_c[2.5 + m_1(nT_s) + m_2(nT_s)]cos(2\pi f_c nT_s)$.
      - Plot the time domain waveforms and the spectrum (use fft) seperately for all cases. Choose a sufficiently large f_s such that the modulated signal is not aliased. I am suggesting a f_s of 20KHz.
      - Brief overview of fft: Fast Fourier Transform (FFT) function is an effective tool for computing the Discrete Fourier Transform (DFT) of a signal. DFT is the way computer calculates the Fourier spectrum of a signal. Typical syntax of fft: fft(x,N), where x(n) is the signal and N is the number of points in the fft. DFT of a signal is complex valued. So it is necessary to look at the magnitude and phase spectrum or real and imaginary parts of the fft. It is periodic, One period extends from 0 to f_s. With in this period, it is symmetric about $\frac{f_s}{2}$ called Nyquist frequency. Therefore typically the fft spectrum is shown from 0 to $\frac{f_s}{2}$.

   (b) Demodulation: Build an AM receiver/demodulator. You can use the same signal generated for part a. You can use one of the following method
      i. Peak detection: Consider the modulated (time domain) signal. You can get the maximum value of the time domain modulated signal at a point, store this value until the next peak and so forth. Then plot the results. The plot should give an approximate envelope. This is a crude envelope. Then it can be passed throught

a Low Pass filter to result in a smooth plot, LPF removes the jumps from one value to other.

ii. hilbert transform: When you take the hilbert transformed signal, the result is $s(nT_s) = A_c[2.5+m_1(nT_s)+m_2(nT_s)]e^{(2\pi f_c nT_s)}$. When the magnitude (absolute value) is taken the signal is obtained.

4. Answer the following questions TUTORIAL QUESTIONS
   (a) [redacted] Q1, Q2
   (b) [redacted] Extra Credits: Q3, Q4

2 Grading format

The lab will be graded for a total of 10 points. There is an option for extra credit, you will not loose any points.

1. 4 points for Steps 1.1 - 1.2
2. 3 points for Steps 1.3 (Matlab)
3. 3 points for Step 1.4 (for answering the questions)

You can work in groups but everybody should submit an individual report. Feel free to write any other comments (what did you like about the lab and what you didn't, so forth).

[redacted]