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

Exam 1: Matlab

DUE OCTOBER 16 2009

This matlab section accounts to 50% of Exam-I grade. Print the code with the report and also send me the .m file. Please save it as exam1_yourname.m

1. Modulation: Generation of an AM signal (Text book Reference: Page 303)

   - 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 $\frac{f_s}{2}$.

2. Demodulation: Build an AM receiver (demodulator). (Text book Reference: Pages 266,267)

   - Coherent Demodulation: multiply with $cos(\omega_c nT_s)$ and pass it through a low pass filter. You can use an fir1, fir2 commands in matlab to design an FIR filter.
   - Incoherent Demodulation: multiply with $cos(\omega_c nT_s + \theta)$ and pass it through a low pass filter. Plot the result for different phase angles, $\theta = 0, \frac{\pi}{4}, \frac{\pi}{2}, \pi, \frac{3\pi}{2} etc...$ What happens to the amplitude of the demodulated signal.
   - Matlab hints: [b] = fir1(N, f_{cutoff_norm},'low'), where N is the order of the filter and f_{cutoff_norm} is the normalized cut-off frequency, $\frac{f_{cutoff}}{f_s/2}$. And the output of the filter can be obtained by convolving the input with filter coefficients (b) computed above

3. Hilbert Transform: (single sideband, Text book Reference: Page 313)

    - Generate a signal $m(nT_s) = 20cos(2\pi f_1 nT_s) + 10cos(2\pi f_2 nT_s)$.
    - Take the Hilbert transform of the signal. Use the hilbert command in matlab.
    - Take the conjugate of the Hilbert transform. Generate the spectrum for all cases.
    - Hilbert command Matlab help:
      >> help hilbert HILBERT Discrete-time analytic signal via Hilbert transform. X = HILBERT(Xr) computes the so-called discrete-time analytic signal X = Xr + j*Xi (j = $\sqrt{-1}$) such that Xi is the Hilbert transform of real vector Xr.

Good luck.