EE341 Fall 2013
Problem Set 3 Solutions

1)
4.2.8 To descramble the signal use the same system you used to scramble the signal.

4.4.1 Note that $2 \sin(a) \cos(b) = \sin(a - b) + \sin(a + b)$. This results in the equations shown in 4.4.1). This shows the key detriment of QAM. If your synchronous detector has frequency and phase errors not only do you get distortion from your modulated signal, but you get interference from the other signal.

4.4.2b Let $\hat{m}(t)$ be the Hilbert transform of $m(t)$. Then we have that

$$x_{USB}(t) = 2m(t) \cos(1000\pi t) - 2\hat{m}(t) \sin(1000\pi t)$$
and

$$x_{LSB}(t) = 2m(t) \cos(1000\pi t) + 2\hat{m}(t) \sin(1000\pi t)$$

2) Assume that $v(t)$ is an AM signal with $|n(t)| < 1$ and that $f_0 >> 3dm(t)/dt$.

a) $e_v(t) = 1 + n(t), \phi_v(t) = 3m(t)$, and $v_{LP}(t) = (1 + n(t)) \exp(j3m(t))$.

b) $e_w(t) = \sqrt{m^2(t) + n^2(t)}, \phi_w(t) = -\tan^{-1}(m(t)/n(t)) + \pi/4$, and $w_{LP}(t) = \sqrt{m^2(t) + n^2(t)} \exp(j\pi/4 - j\tan^{-1}(m(t)/n(t)))$. 
c) $x(t)$ is a delayed version of $v(t)$. Express in envelope and phase representation to get that $e_x(t) = 1 + n(t-\tau)$, $\phi_x(t) = 3m(t-\tau) - \omega_0\tau$, and $x_{LP}(t) = (1 + n(t-\tau)) \exp(j(3m(t-\tau) - \omega_0\tau))$.

d) The output of the nonlinearity $z(t) = \text{sgn}(1 + n(t)) \text{sgn}((\cos(\omega_0t + 3m(t)))$. Since $v(t)$ is an AM signal the sgn of the first term is 1 and the second term approximates a square wave at frequency around $f_0 + 3dm(t)/dt/(2\pi)$. Output of BP filter gives $u(t) = K \cos(\omega_0t + 3m(t))$, where $K$ is a constant. We then have that $y(t) = -K(\omega_0 + 3dm(t)/dt) \sin(\omega_0t + 3m(t))$. This signal looks like an AM signal with $e_y(t) = K(\omega_0 + 3dm(t)/dt)$, $\phi_y(t) = 3m(t) - \pi$, and $y_{LP}(t) = K(\omega_0 + 3dm(t)/dt) \exp(j3m(t) - j\pi)$.

3) We set sampling rate of pulse train to $T_0 = .05\text{sec}$. Diagram of transmitter and receiver along with plots shown in Figure 1 and Figure 2).

matlab code:

```matlab
t=-10:.01:10;
triang05=.5-abs(t(951:1051));
m=conv(randn(1,1901),triang05)*.05;% this produces a random lowpass signal with bandwidth 2Hz
p=0*t; p(1:5:2001) = 1; p = conv(p,ones(1,3)); p=p(2:2002);
w=m.*p;
lpf=butt4; % butt4 is lowpass butterworth filter of order 4 and bandwidth of 5Hz
bpf= lpf.*cos(40*pi*t);
x=conv(w,bpf)*.01;
```
x=x(1001:3001);
z=x.*2.*cos(40*pi*t);
mhat=conv(lpf,z)*.01;
mhat=mhat(1001:3001);

Figure 1:

4)

a) The selective filtering is shown in Figure 3). It consists of two modulators and two filters to produce the LSB of the SSB-SC AM system.

b) Figure 4) shows where each of the signals resides in the frequency domain. Note that with the given modulators and filters that the LSB resides at 950KHz. For $Z(f)$ and $X(f)$ only positive frequencies are plotted.

c) This system is better than using one modulator and filter. Using one filter requires that the filter be very accurate going from passband to stopband in 300Hz which is difficult for a filter dealing with frequencies around 1MHz. It is easier to use two filters as the second filter goes from passband to stopband in around 100KHz.

5) Solution is shown in Figure 5).
Figure 2:
Selective Filtering method to generate SSB-SC AM
\[ f_1 = 50\text{KHz} \& f_0 = 950\text{KHz} \]

Figure 3:

Figure 4:
Figure 5: