Assignment 10
Adaptive Filtering

Due April 2, 1996

Please show your work. For those questions that ask you to use Matlab show the exact commands you used to compute the results.

Question 1

In this question you will use an adaptive filter to implement an adaptive equalizer.

To do this question you'll need to use the `lms()` function which is available in the directory `~elec466/matlab`. You will be able to use this function if you have set up the MATLABPATH environment variable as described in Assignment 4 Question 4. The use of this function is as follows:

```
e = lms ( y, x, w, mu )
```

This function implements the LMS algorithm as described in the lecture notes and computes the error vector e_k from two input vectors y and x. The initial filter coefficients w are adapted using the LMS algorithm with convergence parameter mu.

Create a square wave training signal composed of 100 samples of a square wave of frequency $F_s/10$ (*Hint: create a sine wave and use the "greater-than" operator to compare it to 0.*).

Filter the square wave through a channel with impulse response $h(n) = \{0.1, 0.3, 1, -0.2, -0.3, \}$.

Use the `lms()` function so that the training signal is used as the reference signal and the filter adapts to reduce the effect of the channel. Compute the value of μ using the approximate equation given in the notes. Use half of this value of μ and an adaptive filter of length twice that of the channel impulse response.

Plot the original signal, the signal as sent through the channel, the error and the equalized signal. All plots are as a function of time (sample number).