The University of Texas at Austin
Dept. of Electrical and Computer Engineering
Midterm #2

Prof. Brian L. Evans

Date: May 5, 2017 Course: EE 445S

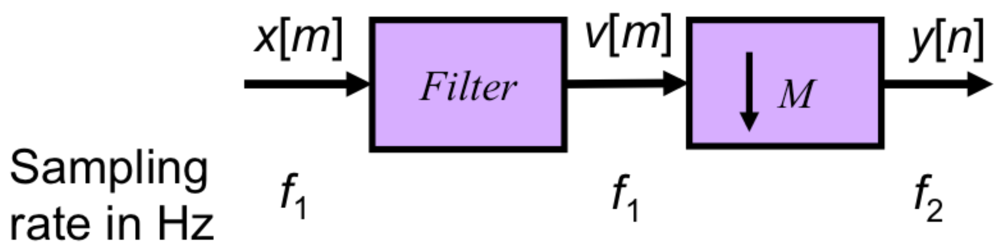
Name: ______________________________________________________________
Last, First

- The exam is scheduled to last 50 minutes.
- Open books and open notes. You may refer to your homework assignments and the homework solution sets. You may not share materials with other students.
- Calculators are allowed.
- You may use any standalone computer system, i.e. one that is not connected to a network. **Disable all wireless access from your standalone computer system**.
- Please turn off all smart phones and other personal communication devices.
- Please remove headphones.
- All work should be performed on the quiz itself. If more space is needed, then use the backs of the pages.
- **Fully justify your answers unless instructed otherwise**. When justifying your answers, you may refer to the Johnson, Sethares & Klein (JSK) textbook, the Welch, Wright and Morrow (WWM) lab book, course reader, and course handouts. Please be sure to reference the page/slide number and quote the particular content in your justification.

| Problem | Point Value | Your score | Topic |
|---|---|---|---|
| 1 | 21 | | Decimation |
| 2 | 27 | | QAM Communication Performance |
| 3 | 28 | | Blind Channel Equalization |
| 4 | 24 | | Channel Equalization With Training |
| Total | 100 | | |

**Problem 2.1.** *Decimation*. *21 points*.

Decimation can change the sampling rate of discrete-time signal $x[n]$ through discrete-time operations of filtering and then downsampling by $M$.




(a) Give a formula for $y[n]$ in terms of $v[\ ]$. *3 points.*

(b) Give a formula for $f_2$ in terms of $f_1$. *3 points*.

(c) Specify the filter's passband frequency $\omega_{pass}$ and stopband frequency $\omega_{stop}$ in rad/sample to pass as many frequencies in $x[m]$ as possible and reduce as many artifacts due to downsampling in $y[n]$ as possible. *6 points*.

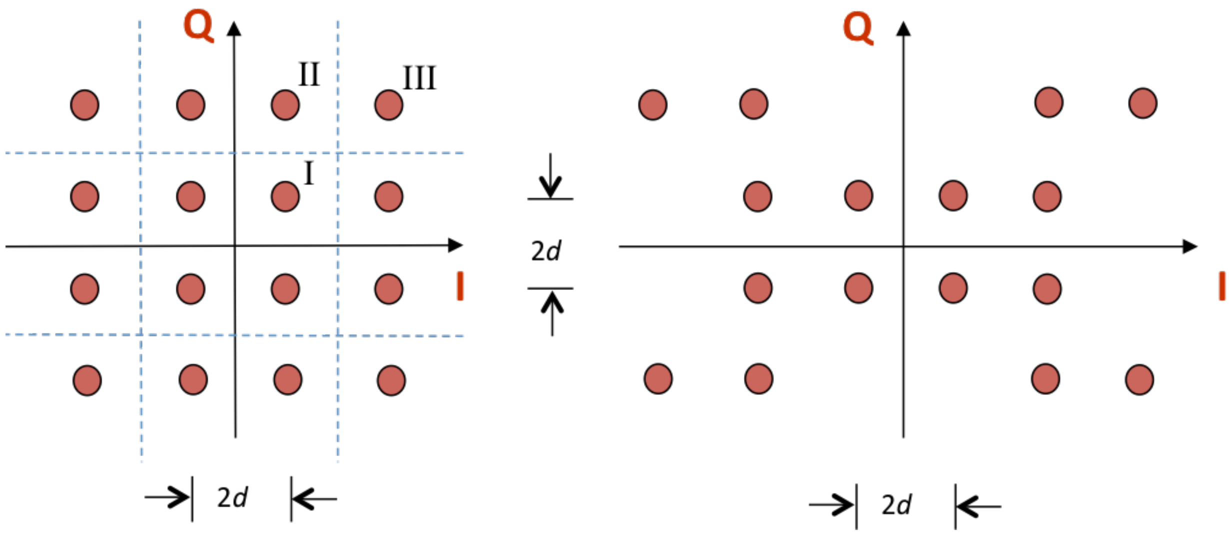
(d) In converting an audio signal sampled at 48 kHz to a speech signal sampled at 8 kHz,

i. What is the value of $M$? *3 points*.

ii. Would you use a finite impulse response filter or an infinite impulse response filter. Why? *6 points*

**Problem 2.2** *QAM Communication Performance. 27 points.*

Consider the two 16-QAM constellations below. Constellation spacing is $2d$.

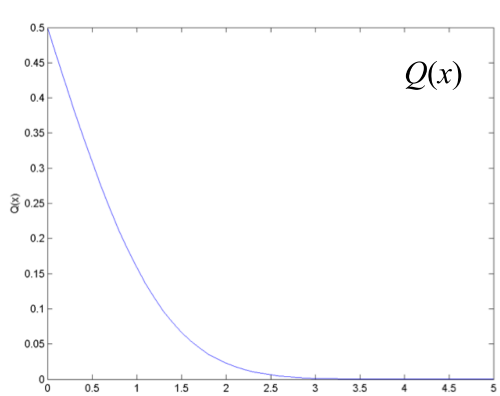



Energy in the pulse shape is 1. Symbol time $T_{sym}$ is 1s. The constellation on the left includes the decision regions with boundaries shown by the in-phase (I) axis, quadrature (Q) axis and dashed lines.

Each part below is worth 3 points. **Please fully justify your answers.**

| | Left Constellation | **Right Constellation** |
|---|---|---|
| (a) Peak transmit power | $18d^2$ | |
| (b) Average transmit power | $10d^2$ | |
| (c) Draw the decision regions for the right constellation on top of the right constellation. | | |
| (d) Number of type I regions | 4 | |
| (e) Number of type II regions | 8 | |
| (f) Number of type III regions | 4 | |
| (g) Probability of symbol error for additive Gaussian noise with zero mean & variance $\sigma^2$ | $3Q\left(\frac{d}{\sigma}\right)-\frac{9}{4}Q^2\left(\frac{d}{\sigma}\right)$ | |

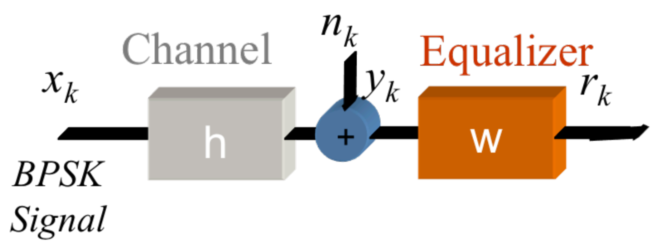
(h) Which constellation has a lower probability of symbol error vs. signal-to-noise ratio? Why? *6 points.*



**Problem 2.3.** *Blind Channel Equalization. 28 points.*

Blind channel equalization occurs without a training sequence, as shown below.

When the transmitted sequence $x[k]$ is binary phase shift keying (BPSK), i.e. is +1 or -1, an adaptive method can be based on the fact that $x^2[k] = 1$.




Assume a two-tap finite impulse response (FIR) equalizer with its first coefficient fixed at one:

$$w[k] = \delta[k] + w_1\, \delta[k-1]$$

(a) Using the objective function

$$J(k) = ¼\, (1 - r^2[k]\,)^2$$

derive the adaptive update equation for $w_1$. *16 points*

(b) Give an initial value for $w_1$. Why did you choose that value? *3 points*

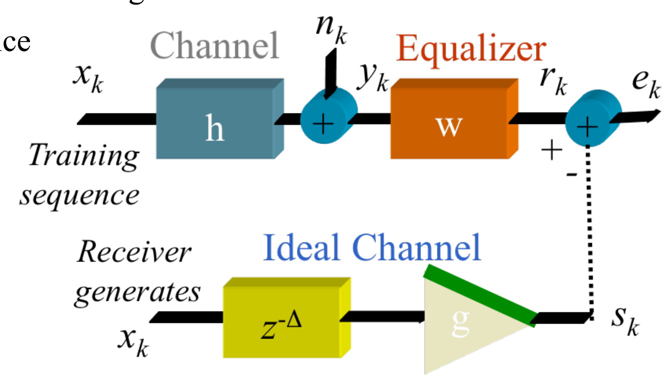
(c) What range of values would you use for the step size $\mu$? Why? *3 points*

(d) How would you adjust the objective function for 4-level pulse amplitude modulation? *6 points*

**Problem 2.4.** *Channel Equalization With Training. 24 points*

For the finite impulse response (FIR) channel equalizer on the right:

(a) Give two reasons why pseudo-noise is a good choice for the training sequence. *3 points.*




(b) Here is the update equation for an adaptive least mean squares FIR filter with $N$ coefficients **w**:

$\mathbf{w}[k+1] = \mathbf{w}[k] - \mu\, e[k]\, \mathbf{y}[k]$

where $\mathbf{y}[k] = [\ y[k]\ \ y[k-1]\ \ \ldots\ \ y[k-(N-1)\ ]$ and $e[k] = r[k] - g\, x[k - \Delta]$ and $r[k] = \text{FIR}\{\ y[k]\ \}$

i. How many multiplications are needed per iteration? How does this compare with an FIR filter? *6 points*

ii. How many words of memory are needed the adaptive FIR filter? How does this compare with an FIR filter? *6 points*

iii. What range of values would you use for the step size $\mu$? Why? *3 points*

(c) For a training sequence of length $2N$, would you advocate using a least squares equalizer or an adaptive least mean squares equalizer? *6 points*