The University of Texas at Austin

Dept. of Electrical and Computer Engineering

Midterm #2

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Date: May 3, 2013 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 cell phones and other personal communication devices.
* 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 textbook, the Welch, Wright and Morrow lab book, course reader, and course handouts. Please be sure to reference the page/slide number and quote the particular content you are using in your justification.

|  |  |  |  |
| --- | --- | --- | --- |
| **Problem** | **Point Value** | **Your score** | **Topic** |
| 1 | 30 |  | Channel Equalization |
| 2 | 24 |  | Quadrature Amplitude Modulation |
| 3 | 24 |  | Data Conversion |
| 4 | 22 |  | Potpourri |
| Total | 100 |  |  |

**Problem 2.1.** *Channel Equalization*. *30 points*.

In the discrete-time system on the right, the  
equalizer operates at the sampling rate.

Equalizer has real coefficients *w*0 and *w*1:

*r*[*k*] = *w*0 *y*[*k*] + *w*1 *y*[*k*-1]

You may ignore the noise signal *nk*.

For the adaptive least mean squares (LMS)  
equalizer, the objective function is

During training, the update equation for *w*1 for iteration *k* +1 is

where  is the constant step size.

1. Derive the update equation for *w*0 for an adaptive LMS equalizer. *12 points*.
2. Prior to training, what initial values would you give *w*0 and *w*1? Why? *6 points*.
3. Let the vector of equalizer coefficients be **w** = [*w*0 *w*1]. Using the result from (a), write the update in one equation in vector form. Please define any new vectors that you introduce. *6 points*.
4. For an adaptive LMS equalizer with *n* coefficients, how many multiplications are needed per training sample? *6 points*.

**Problem 2.2** *Quadrature Amplitude Modulation (QAM). 24 points.*

Consider the two 32-QAM constellations below. Constellation spacing is 2*d*.

Energy in pulse shape is 1. Symbol time *T*sym is 1s.

|  |  |  |
| --- | --- | --- |
|  | Left Constellation | Right Constellation |
| (a) Peak power | 58 *d* 2 |  |
| (b) Average power | 25.875 *d* 2 |  |
| (c) Number of type I regions | 12 |  |
| (d) Number of type II regions | 16 |  |
| (e) Number of type III regions | 4 |  |

Fill in each entry (a)-(e) for the right constellation. Each entry is worth *3 points*.

Which of the two constellations would you advocate using? Why? *9 points*.**Problem 2.3.** *Data Conversion*. *24 points*.

For an analog-to-digital converter running at sampling rate of *fs* and quantizing to *B* bits, here is a block diagram of a sigma-delta modulation implementation of the A/D:



The internal clock runs at *M* *fs*.

1. Replace the quantizer with a constant gain of *K* and assume *K* ≥ 2. Derive the signal transfer function in the *z*-domain for input *x*[*m*] and output *b*[*m*]. Please set the dither to zero. *12 points*.
2. We can design the FIR filter prior to downsampling as a cascade of an equalizer and an anti-aliasing filter. Assuming that *h*[*m*] is an FIR filter, please define the equalizer as the FIR filter that cancels the poles in the signal transfer function found in (a). *6 points.*
3. Give a filter specification for the anti-aliasing filter. *6 points*.

**Problem 2.4.** *Potpourri*. *22 points*.

Please determine whether the following claims are true or false. If you believe the claim to be false, then provide a **counterexample**. If you believe the claim to be true, then give **supporting evidence** that may include formulas and graphs as appropriate. If you give a true or false answer without any justification, then you will be awarded **zero points** for that answer. If you answer by simply rephrasing the claim, you will be awarded **zero points** for that answer.

1. In a certain QAM system, pseudo-noise sequence is sent at the beginning of transmission. In the receiver, one would correlate against the known the PN sequence to determine when transmission has begun instead of an energy detector because the correlator has lower complexity. *7 points.*
2. The symbol recovery method based on appendix M of the course reader and discussed in lecture 16 on QAM Receivers uses the Fourier property that a shift in time corresponds to a shift in frequency. That is why the method locks onto frequencies ωc – ωsym and ωc + ωsym for QAM symbol recovery. *7 points*.
3. In communication channel modeling, we model the frequency selectivity using a finite impulse response (FIR) filter because the linear time-invariant properties of all physical channels are FIR. *8 points.*