The University of Texas at Austin

Dept. of Electrical and Computer Engineering

Midterm #2

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Date: December 6, 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.

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| --- | --- | --- | --- |
| **Problem** | **Point Value** | **Your score** | **Topic** |
| 1 | 25 |  | Channel Equalization |
| 2 | 27 |  | Receiver Design |
| 3 | 30 |  | Pre-emphasis |
| 4 | 18 |  | Potpourri |
| Total | 100 |  |  |

**Problem 2.1.** *Channel Equalization*. *25 points*.

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

Equalizer is a finite impulse response (FIR) filter with two 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*.

1. For the adaptive FIR equalizer, derive the update equation for *w*1 for the following objective function: *12 points.*
2. One of the problems with the adaptive FIR equalizer in part (a) is that its convergence depends on the initial value for *w*1.

Consider finding the roots of the polynomial

i. Give the iterative update equation for estimates for *x*. *6 points*.

ii. From the iterative update equation in part i, give the range of initial values of *x* to guarantee convergence. The range of values may depend on the step size . *7 points*

**Problem 2.2** *Receiver Design. 27 points.*

Consider the baseband pulse amplitude modulation (PAM) receiver blocks below with

* sampling rate *f*s
* downsampling factor *L* = 6 samples/symbol where *f*s = *L* *f*sym
* square root raised cosine pulse shape *g*[*m*] with rolloff parameter  = 1:

1. Why is placing the FIR equalizer immediately after the A/D converter inefficient? Completing steps (b)-(d) below might help you here. *6 points*.
2. The first step to remove the inefficiency is to swap the order of the equalizer and matched filter. How can this be justified? *6 points*.
3. Show in the discrete time domain that downsampling by 6 is the same as downsampling by 3 followed by downsampling by 2. *6 points*.
4. The second step to reduce the inefficiency is to exchange the FIR equalizer with the downsampling by 3. *9 points*.
5. How can this exchange be justified?
6. What is the frequency band in Hz over which the FIR equalizer has to equalize?

**Problem 2.3.** *Pre-emphasis*. *30 points*.

Consider an upconverted baseband 2-PAM signal *x*(*t*) = *s*(*t*) cos(2  *f*c *t*) where   
*s*(*t*) is a baseband 2-PAM signal with

Constellation spacing 2 *d*

Symbol rate *f*sym

Sampling rate *f*s

Samples per symbol *L* = 20

Rolloff factor  = 1

and where

Carrier frequency *f*c = 2 *f*sym

Transmission bandwidth *B* = 2 *f*sym

The received signal is *r*(*t*) = *x*(*t*) + *n*(*t*) where *n*(*t*) is spectrally-flat Gaussian noise.

Here is the block diagram for pre-emphasis filtering where *q* is an integer and *q* > 1:

The nonlinearity raises the input to the *q*th power.

(a) Give the passband and stopband frequencies for bandpass filter (BPF) #1. *6 points*.

1. Pre-emphasis of carrier frequency *f*c. *12 points*.

i. give all possible values for *q*

ii. which value of *q* would you use and why?

iii. give the center frequency for BPF #2

(c) Pre-emphasis of symbol clock *f*sym. *12 points*.

i. give all possible values for *q*

ii. which value of *q* would you use and why?

iii. give the center frequency for BPF #2

**Problem 2.4.** *Potpourri*. *18 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.

(a) Applying a lowpass filter to a spectrally-flat Gaussian noise signal always produces an output signal with lower average noise power than that of the input signal. *6 points*.

1. In discrete time, an ideal channel can be modeled as

The input *x*[*m*] can always be exactly recovered by discarding the first  samples of *y*[*m*] and scaling each subsequent sample by 1/*g*. *6 points*.

(c) For a synchronized quadrature amplitude modulation (QAM) receiver and an additive spectrally-flat Gaussian noise channel, the in-phase noise will always be statistically independent of the quadrature noise when measured at the input to the decision block. *6 points*.