#### S-72.227 Digital Communication Systems



Spring 2002 Tutorial#7, 11.03.2002

### E 7.1

Binary PAM is used to transmit information over an unequlaized linear filter channel. When a=1 is transmitted, the noise free output of the demodulator is

$$x_{m} = \begin{cases} 0.3 & (m=1) \\ 0.9 & (m=0) \\ 0.3 & (m=-1) \\ 0 & (\text{otherwise}) \end{cases}$$

a) Design a three-tap zero-forcing equalizer so that the output is

$$q_m = \begin{cases} 1 & (m=0) \\ 0 & (m=\pm 1) \end{cases}$$

b) Determine  $q_m$  for  $m=\pm 2$ ,  $\pm 3$  by convolving the impulse response of the equalizer with the channel response.

# E 7.2

Repeat problem (E 7.1) using the MMSE as the criterion for optimizing the tap coefficients. Assume that the noise power spectral density is 0.1W/Hz.

### E 7.3

A time-dispersive channel having an impulse response h(t) is used to transmit four-phase PSK at a rate R=1/T symbols/s. The equivalent discrete-time channel is shown in Figure 7.1. The sequence  $\{\eta_k\}$  is white noise sequence having zero mean and variance  $\sigma^2 = N_o$ .

a) What is the sampled autocorrelation function sequence  $\{x_k\}$  for this channel , where  $\{x_k\}$  defined by

$$x_k = \int_{-\infty}^{\infty} h^*(t)h(t+kT)dt$$

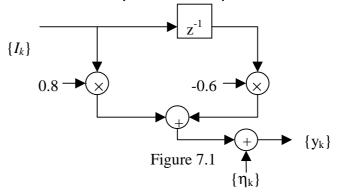
b) The minimum MSE performance of a linear equalizer and a decision feedback equalize having an infinite number of taps depends on the folded spectrum of the channel

$$\frac{1}{T}\sum_{n=-\infty}^{\infty} \left| H(\omega + \frac{2\pi n}{T}) \right|^2$$

where  $H(\omega)$  is the Fourier transform of h(t).

Determine the folded spectrum.

- c) Use your answer in (b) to express the minimum MSE of a linear equalizer in terms of the folded spectrum.
- d) Repeat (c) for an infinite-tap decision equalizer.



## Homework-7 Deadline 08 April 2002 at 10.00

Homework return box is located at Otakaari 5, 2nd floor, near the E-wing. You can also return the answers to the assistant just before the class.

The real and imaginary parts of the filter coefficients  $\{c_n(i)\}$  as well as the signal phase  $\phi$  at the output of the linear equalizer can be adjusted iteratively towards their optimum values using the steepest descent algorithm,

$$\begin{aligned} &\operatorname{Re}\{c_{n}(i)\} = \operatorname{Re}\{c_{n}(i)\} - \Delta \frac{\partial |e_{k}|^{2}}{\partial [\operatorname{Re}\{c_{n}\}]} \\ &\operatorname{Im}\{c_{n}(i)\} = \operatorname{Im}\{c_{n}(i)\} - \Delta \frac{\partial |e_{k}|^{2}}{\partial [\operatorname{Im}\{c_{n}\}]} \\ &\phi\{i+1\} = \phi(i) - \Delta_{\phi} \frac{\partial |e_{k}|^{2}}{\partial \phi} \end{aligned}$$

where the error signal is defined as

$$e_k = z_k - \hat{b}_k = \left(\sum_{m=-M}^{M} c_m r_{k-m}\right) \exp(j\phi) - \hat{b}_k$$

Derive the results given in the handouts by setting this expression of  $e_k$  into the iteration equations shown above.

For definition of variables, please refer to lecture handouts.