

# ELEG-636: Statistical Signal Processing

Gonzalo R. Arce

Department of Electrical and Computer Engineering  
University of Delaware

Spring 2010

# Course Objectives & Structure

**Objective:** Given a discrete time sequence  $\{x(n)\}$ , develop

- Statistical and spectral signal representation
- Filtering, prediction, and system identification algorithms
- Optimization methods that are
  - Statistical
  - Adaptive

**Course Structure:**

- Weekly lectures [notes: [www.ece.udel.edu/~arce](http://www.ece.udel.edu/~arce)]
- Periodic homework (theory & Matlab implementations) [15%]
- Midterm & Final examinations [85%]

**Textbook:**

- Haykin, Adaptive Filter Theory.

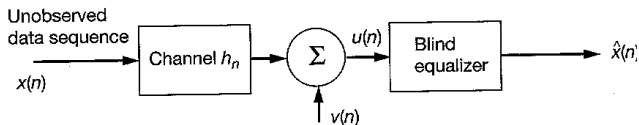
# Course Objectives & Structure

- Broad Applications in Communications, Imaging, Sensors.
- Emerging application in
  - Brain-imaging techniques
  - Brain-machine interfaces,
  - Implantable devices.
- Neurofeedback presents real-time physiological signals from MRIs in a visual or auditory form to provide information about brain activity. These signals are used to train the patient to alter neural activity in a desired direction.
- Traditionally, feedback using EEGs or other mechanisms has not focused on the brain because the resolution is not good enough.

# Blind Deconvolution

## Motivation:

- Adaptive equalizers typically require a training period during which they operate on known signals/statistics.
- This known signal training is not always appropriate, e.g., in mobile communications
  - Cost is too high (time/bandwidth)
  - Multipathing or other time varying interference
- In such cases, we must use blind equalization



System components and assumptions:

- Channel introduces distortion (dominant)
- System has additive noise (not dominant)
- Assume a baseband model of communications
  - Multilevel pulse amplitude modulation ( $M$ -ary PAM)
  - Received signal:

$$u(n) = \sum_k h_k x(n - k)$$

- Dominating interference is due to intersymbol interference (ISI) from channel distortion
  - ⇒ The noise is ignored

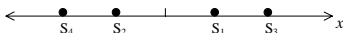
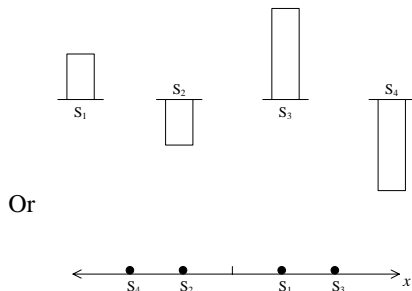
Also assume that:

$h \neq 0$  for  $n < 0$  (noncausal)

$$\sum_k h_k^2 = 1 \quad (\text{to keep the variance of the output constant})$$

### Example (4-ary PAM modulation)

A 4-ary PAM modulation scheme uses 4 signals



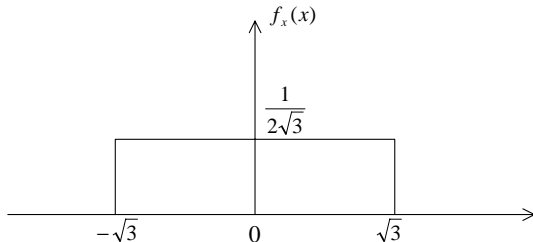
To solve the equalization problem, we need a statistical model of the data. Assume,

- 1 The data is white

$$E\{x(n)\} = 0$$

$$E\{x(n)x(k)\} = \begin{cases} 1 & k = n \\ 0 & \text{otherwise} \end{cases}$$

- 2 The pdf of  $x(n)$  is symmetric and uniform



**Deconvolution Objective:** If  $\{w_i\}$  are the coefficients of the ideal inverse filter, then

$$\sum_i w_i h_{l-i} = \delta_l = \begin{cases} 1 & l = 0 \\ 0 & \text{else} \end{cases}$$

If this is the case, the output of the equalizer is

$$\begin{aligned} y(n) &= \sum_i w_i u(n-i) \\ &= \sum_i \sum_k w_i h_k x(n-i-k) && \text{[let } k = l - i\text{]} \\ &= \sum_l x(n-l) \sum_i w_i h_{l-i} \\ &= \sum_l x(n-l) \delta_l \\ &= x(n) \end{aligned}$$

**Problem:**  $h_n$  is not known  $\Rightarrow$  the exact inverse can not be used



**Solution:** Use an iterative procedure to find the filter.

Let the output at iteration  $n$  be given by

$$y(n) = \sum_{i=-L}^L \hat{w}_i(n)u(n-i)$$

where a  $2L + 1$  tap filter is used

Setting  $\hat{w}_i(n) = 0$  for  $|i| > L$ , we can write

$$\begin{aligned} y(n) &= \sum_i \hat{w}_i(n)u(n-i) \\ &= \sum_i w_i u(n-i) + \sum_i [\hat{w}_i(n) - w_i]u(n-i) \\ &= x(n) + v(n) \quad \left[ \text{since } x(n) = \sum_i w_i u(n-i) \right] \end{aligned}$$

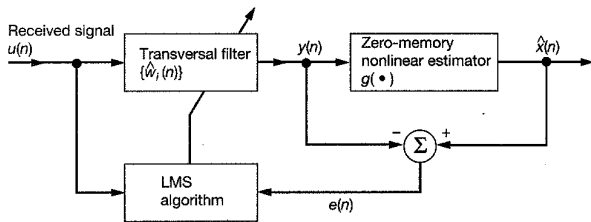
Note  $v(n) = \sum_i [\hat{w}_i(n) - w_i]u(n-i)$  is the **residual ISI**

**Interpretation:**  $v(n) = \sum_i [\hat{w}_i(n) - w_i] u(n - i)$  is the convolution noise (residual ISI) resulting from the fact that ideal filter was not used

**Estimation Approach:** Apply the output  $y(n) = x(n) + v(n)$  to a zero memory nonlinear estimator

$$\hat{x}(n) = g[y(n)]$$

Complete System:



- The nonlinear estimate  $g[y(n)]$  can be used to update the equalizer to produce a better estimate at time  $n + 1$

Define the equalizer error to be

$$e(n) = \hat{x}(n) - y(n)$$

Use  $e(n)$  in the LMS algorithm to update the equalizer weights

$$\hat{w}_i(n+1) = \hat{w}_i(n) + \mu u(n-i)e(n) \quad i = 0, \pm 1, \pm 2, \dots, \pm L$$

Note that in this case, the cost function is

$$\begin{aligned} J(n) &= E\{e^2(n)\} \\ &= E\{(\hat{x}(n) - y(n))^2\} \\ &= E\{(g[y(n)] - y(n))^2\} \end{aligned}$$

**Observation:**  $J(n)$  is a nonconvex function of the filter weights ( $g[\cdot]$  is nonlinear)

- The cost function can have numerous local minima

To solve the optimization, first evaluate the convolution noise,

$$u(n) = \sum_k h_k x(n-k)$$

$$\Rightarrow u(n-i) = \sum_k h_k x(n-i-k)$$

Next, use this in

$$y(n) = \underbrace{\sum_i w_i u(n-i)}_{x(n)} + \underbrace{\sum_i [\hat{w}_i(n) - w_i] u(n-i)}_{v(n)}$$

where recall

$w_i \equiv$  perfect equalizer weights

$\hat{w}_i(n) \equiv$  finite approximate equalizer with  $\hat{w}_i(n) = 0$  for  $|i| > L$

Then using  $u(n-i) = \sum_k h_k x(n-i-k)$

$$\begin{aligned}
 v(n) &= \sum_i [\hat{w}_i(n) - w_i] u(n-i) \\
 &= \sum_i \sum_k h_k [\hat{w}_i(n) - w_i] x(n-i-k) \\
 &= \sum_l x(l) \nabla(n-l)
 \end{aligned}$$

where the last line follows from letting  $n-i-k = l$  and defining

$$\nabla(n) = \sum_k h_k [\hat{w}_{n-k}(n) - w_{n-k}]$$

- $\nabla(n)$  is the residual impulse response of the channel due to imperfect equalization
- $\nabla(n)$  is small in value, but long and oscillatory

Since the convolution noise is given by

$$v(n) = \sum_l x(l) \nabla(n-l)$$

$$\Rightarrow E\{v(n)\} = \sum_l E\{x(l)\} \nabla(n-l) = 0$$

Also

$$E\{v(n)v(n-j)\} = E\left\{\sum_l x(l) \nabla(n-l) \sum_m x(m) \nabla(n-m-j)\right\}$$

$$= \sum_l \sum_m \nabla(n-l) \nabla(n-m-j) E\{x(l)x(m)\}$$

$$= \sum_l \nabla(n-l) \nabla(n-l-j)$$

Since  $\nabla(n)$  is long and oscillatory

$$E\{v(n)v(n-j)\} = \sum_l \nabla(n-l)\nabla(n-l-j)$$

tends to average to 0 for  $j \neq 0$ . Thus

$$E\{v(n)v(n-j)\} = \begin{cases} \sigma^2 & j = 0 \\ 0 & j \neq 0 \end{cases}$$

where

$$\sigma^2 = \sigma^2(n) = \sum_l \nabla^2(n-l)$$

**Observation:**  $v(n) = \sum_l x(l)\nabla(n-l)$  is a weighted sum of i.i.d. RVs  $\Rightarrow$   
 $v(n)$  is Gaussian (central limit theorem)

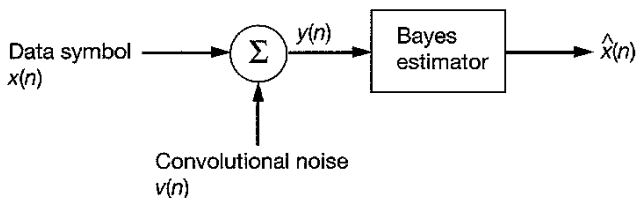
Lastly, consider the cross correlation of  $x(n)$  and  $v(n)$ ,

$$\begin{aligned}
 E\{x(n)v(n-j)\} &= E\{x(n) \sum_l x(l) \nabla(n-l-j)\} \\
 &= \sum_l \nabla(n-l-j) E\{x(n)x(l)\} \\
 &= \nabla(-j) \\
 &\ll \sum_l \nabla^2(l) = \sigma^2
 \end{aligned}$$

**Observation:** Thus we can say  $x(n)$  and  $v(n)$  are essentially independent



Finally, consider the nonlinear estimation of  $x(n)$



Component statistics:

- $x(n)$  is uniformly distributed with zero mean and unit variance
- $v(n)$  is white Gaussian noise with zero mean and variance  $\sigma^2(n)$
- $x(n)$  and  $v(n)$  are independent

**Estimation Approach:** Utilize Bayes estimation, which exploits knowledge of the distributions

The estimation risk is defined by

$$\begin{aligned} R &= E\{c(x, \hat{x}(n))\} \\ &= \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} c(x, \hat{x}(n)) f_{xy}(x, y) dy dx \end{aligned}$$

Recall that if  $c(x, \hat{x}(n)) = (x - \hat{x}(n))^2$ , then the risk is minimized by

$$\begin{aligned} \hat{x} &= \int_{-\infty}^{\infty} x f_x(x|y) dx \\ &= E\{x|y\} \quad \text{[conditional expectation]} \end{aligned}$$

where we can use the fact that

$$f_x(x|y) = \frac{f_y(y|x) f_x(x)}{f_y(y)}$$

Employ the current model,

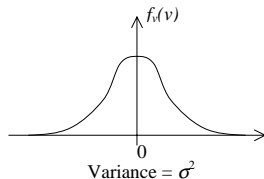
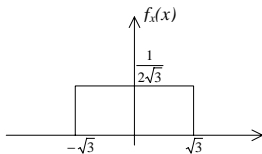
$$y = c_0 x + v$$

where  $c_0 < 1$  is a scaling factor included to ensure  $E\{y^2\} = 1$ .

Thus since  $x$  and  $v$  are independent,  $f_y(y|x) = f_v(y - c_0 x)$  and

$$\begin{aligned} \hat{x} &= \int_{-\infty}^{\infty} x f_x(x|y) dx \\ &= \frac{1}{f_y(y)} \int_{-\infty}^{\infty} x f_y(y|x) f_x(x) dx \\ &= \frac{1}{f_y(y)} \int_{-\infty}^{\infty} x f_v(y - c_0 x) f_x(x) dx \end{aligned}$$

where



Evaluating this yields (Berllini, 1988)

$$\hat{x} = \frac{1}{c_0 y} - \frac{\sigma}{c_0} \frac{Z(y_1) - Z(y_2)}{Q(y_1) - Q(y_2)}$$

where

$$y_1 = \frac{1}{\sigma}(y + \sqrt{3}c_0)$$

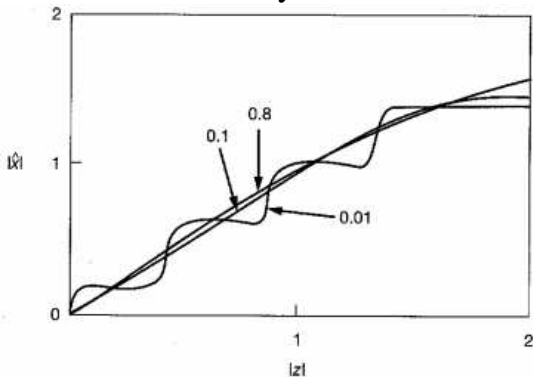
$$y_2 = \frac{1}{\sigma}(y - \sqrt{3}c_0)$$

and

$$Z(y) = \frac{1}{\sqrt{2\pi}} e^{-\frac{y^2}{2}} \quad \text{[normalized Gaussian pdf]}$$

$$Q(y) = \frac{1}{\sqrt{2\pi}} \int_y^{\infty} e^{-\frac{u^2}{2}} du \quad \text{[normalized Gaussian cdf]}$$

Results for an 8 level PAM system:



- Plotted for three different noise to (signal + noise) ratios
- Note that this tends to a step function as the noise goes to zero

**Implementation Point:** When the blind equalizer has converged, the algorithm is switched to decision-directed mode.

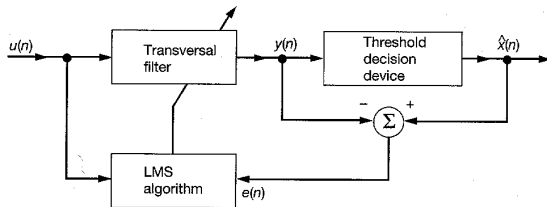
In the case of binary symbols,

$$x(n) = \begin{cases} +1 & \text{for symbol 1} \\ -1 & \text{for symbol 2} \end{cases}$$

which results in the sample estimate

$$\Rightarrow \hat{x}(n) = \text{sgn}(y(n)) = \begin{cases} +1 & \text{if } y(n) \geq 0 \\ -1 & \text{else} \end{cases}$$

## Final System:



**Final Observation:** This system works well as long as what?

# Blind Deconvolution

Adaptive equalizers typically require a training period during which they operate on known signals/statistics.

This known signal training is not always appropriate such as in mobile communications

- Cost is too high (time/bandwidth)
- Multipathing or other interference

In such cases, we must use blind equalization.