Further Discussions and Beyond EE630

Electrical & Computer Engineering University of Maryland, College Park

Acknowledgment: The ENEE630 slides here were made by Prof. Min Wu. *Contact: minwu@umd.edu*



UMD ENEE630 Advanced Signal Processing

End of Semester Logistics

- Project due
- Final exam: two hours, close book/notes
 - Mainly cover Part-2 and Part-3
 - May involve basic multirate concepts from Part-1 (decimation, expansion, basic filter bank)
- Office hours

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Discussions [2]

Higher-Order Signal Analysis: Brief Introduction

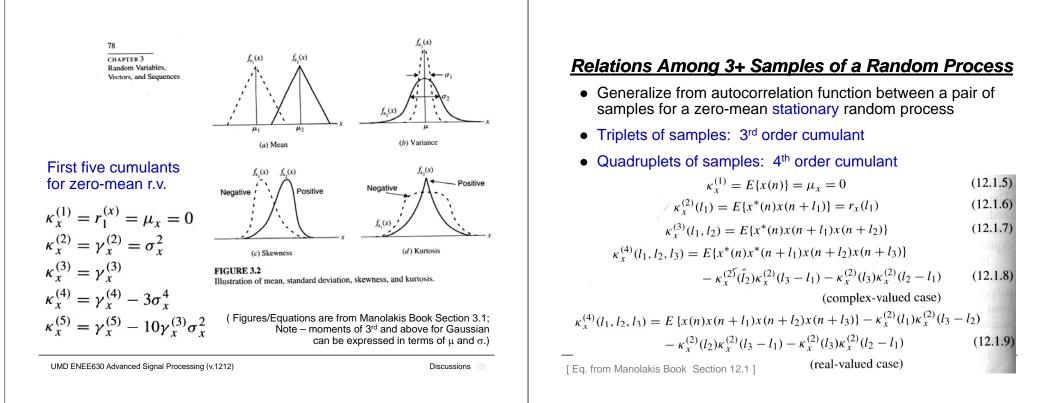
- Information contained in the power spectrum
 - Reflect the 2nd-order statistics of a signal (i.e. autocorrelation)
 - => Power spectrum is sufficient for complete statistical description of a Gaussian process, but not so for many other processes
- Motivation for higher-order statistics
 - Higher-order statistics contain additional info. to measure the deviation of a non-Gaussian process from normality
 - Help suppress Gaussian noise of unknown spectral characteristics.
 - The higher-order spectra may become high SNR domains in which one can perform detection, parameter estimation, or signal reconstruction
 - Help identify a nonlinear system or to detect and characterize nonlinearities in a time series

mth-order Moments of A Random Variable

- Moments: $m_k = E[X^k];$
- Central moments: subtract the mean $\gamma_k = E[(X \mu_X)^k]$
- Mean: $\mu_X = m_1 = E[X]$
 - Statistical centroid ("center of gravity")
- Variance: $\sigma_X^2 = \gamma_2 = E[(X \mu_X)^2]$
 - Describe the spread/dispersion of the p.d.f.
- 3rd Moment: normalize into $K_3 = \gamma_3 / \sigma_X^3$
 - Represent Skewness of p.d.f. → zero for symmetric p.d.f.
- \circ 4th Moment: normalize into K₄ = γ_4 / σ_X^4 3
 - "Kurtosis" for flat/peakiness deviation from Gaussian p.d.f. (which is zero)

See Manolakis Sec.3.1.2 for further discussions

Frequency estimation [3]



High-order Spectra

- Multi-variable DTFT on cumulant functions
 - Bispectrum & Trispectrum: may exhibit patterns in magnitude & phase
- Extend properties under LTI to high-order stats

$$R_x^{(2)}(e^{j\omega}) \triangleq \sum_{l_1=-\infty}^{\infty} \kappa_x^{(2)}(l_1)e^{-j\omega l_1} = R_x(e^{j\omega})$$
(12.1.15)

$$R_x^{(3)}(e^{j\omega_1}, e^{j\omega_2}) \triangleq \sum_{l_1=-\infty}^{\infty} \sum_{l_2=-\infty}^{\infty} \kappa_x^{(3)}(l_1, l_2) e^{-j(\omega_1 l_1 + \omega_2 l_2)} \quad \text{(bispectrum)} \quad (12.1.16)$$

and
$$R_x^{(4)}(e^{j\omega_1}, e^{j\omega_2}, e^{j\omega_3}) \triangleq \sum_{l_1=-\infty}^{\infty} \sum_{l_2=-\infty}^{\infty} \sum_{l_3=-\infty}^{\infty} \kappa_x^{(4)}(l_1, l_2, l_3)e^{-j(\omega_1 l_1 + \omega_2 l_2)}$$

(trispectrum) (12.1.17)

See Manolakis et al. McGraw Hill book "Statistical & Adaptive S.P." Sec.12.1 High-order statistics for further discussions

Resource on Signal Processing

- IEEE Signal Processing Magazine
 - E-copy on IEEE Xplore; Hard-copy by student membership
- IEEE "Inside Signal Processing eNewsletter" http://signalprocessingsociety.org/newsletter/
- Signal Processing related journals/transactions
- Related conferences: ICASSP, ICIP, etc.
- Additional 2-cents beyond courses
 - Attend talks/seminars to broaden your vision
 - Oral communications (oral exams, presentations, etc)

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Discussions [1

Related Courses Beyond EE630

- Adaptive and space-time signal processing: ENEE634*
- Image/video & audio/speech processing: ENEE631*, 632
- Detection/estimation & information theory: ENEE621*, 627*
 - See also SP for digital communication in ENEE623
- Pattern recognition and machine learning: ENEE633
- Special topic courses and seminars in signal processing: Occasionally offered. E.g. on info forensics & multimedia security, compressive sensing, etc.
- See also related applied math and statistics courses

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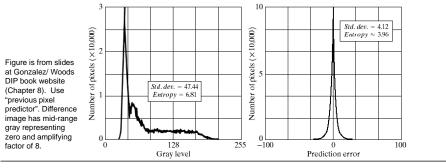
Discussions [11]

a b c

FIGURE 8.20 (a) The prediction error image resulting from Eq. (8.4-9). (b) Gray-level histogram of the original image.







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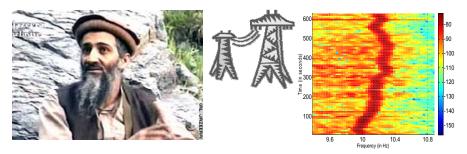
Discussions [12]

Digital Image and Video Processing (ENEE631)

- Human visual perception; color vision
- Image enhancement
- Image restoration
- Image transform, quantization and coding
- Motion analysis and video coding
- Feature extraction and analysis
- Security and forensic issues

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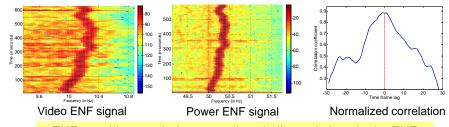
Forensic Question on "Time" and "Place"



- When was the video actually shot? And where?
- Was the sound track captured at the same time as the picture? Or super-imposed afterward?
- Explore the fingerprint influenced by power grid onto sensor recordings

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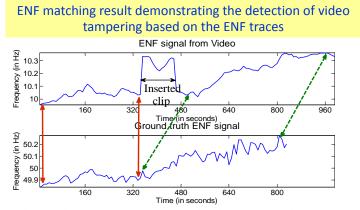
Ubiquitous Forensic Fingerprints from Power Grid



ENF matching result demonstrating similar variations in the ENF signal extracted from video and from power signal recorded in India

- Electric Network Frequency (ENF): 50/60 Hz nominal
 - · Varies slightly over time; main trends consistent in same grid
 - Can be "seen" or "heard" in sensor recordings
- Help determine recording time, detect tampering, etc.
- Other potential applications on smart grid & media management
 - → Ref: Garg et al. ACM Multimedia 2011, CCS 2012 and APSIPA 2012

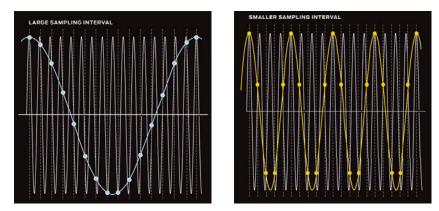
Tampering Detection Using ENF



- Adding a clip between the original video leads to discontinuity in the ENF signal extracted from video
- Clip insertion can also be detected by comparing the video ENF
 signal with the power ENF signal at corresponding time

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Aliasing Revisit: Downsample A Sinusoid



"If the RF signal [white] is not sampled at least twice per cycle, aliasing will occur. But by properly adjusting the sampling interval [indicated by vertical lines], you can down-convert the RF to whatever lower frequency is desired [blue and yellow]."

IEEE Spectrum Magazine April 2009 "Universal Handset" – Alias Harnessed for software-defined radio http://spectrum.ieee.org/computing/embedded-systems/the-universal-handset/0/cellsb01 ENEE630 Look Ahead

Introduction to Adaptive Filtering

Electrical & Computer Engineering University of Maryland, College Park

Acknowledgment: the additional overview/introductory slides for beyond ENEE630 were made by Prof. Min Wu and FFP Teaching Fellow Mr. Wei-Hong Chuang, with reference to textbooks by Hayes and Haykins and ENEE634 class notes by Prof. Ray Liu. Contact: minwu@umd.edu



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Stationarity Assumption in Wiener Filtering

- Wiener filtering is optimum in a stationary environment
 - Unfortunately, most real signals are non-stationary
- One remedy: process the non-stationary signal in blocks, where the signal is assumed to be stationary
- Not always effective
 - For rapidly varying signals, the block length may be too small to estimate relevant parameters
 - Can't accommodate step changes within analysis intervals
 - Solution imposes incorrect data model, i.e., piecewise stationary

=> Try to begin with non-stationarity to develop solutions

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Adaptive Filtering
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Recursive Update of Filter Coefficients

Wiener Filtering: solve the normal equation

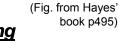
$$\mathbf{R}_{x}\mathbf{w} = \mathbf{r}_{dx}$$

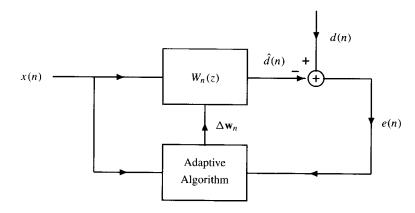
 If non-stationary, optimal filter coefficients will depend on time n

$$\mathbf{R}_{x}(n)\mathbf{w}_{n}=\mathbf{r}_{dx}(n)$$

- Not always feasible (e.g. high computational complexity)
- Can be much simplified with adaptive filtering:
 - => Form \mathbf{w}_{n+1} by adding correction $\Delta \mathbf{w}_n$ to \mathbf{w}_n at each iteration $\mathbf{W}_{n+1} = \mathbf{W}_n + \Delta \mathbf{W}_n$

General Structure of Adaptive Filtering



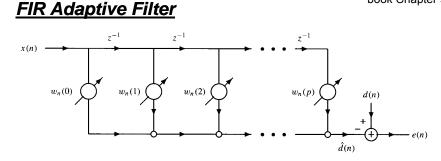


• Measure the error e(n) at each time *n*, determine how to update filter coefficients accordingly

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Adaptive Filtering

(Fig. from Hayes' book Chapter 9)



- Simple & efficient algorithms for coefficient adjustment
- Often perform well enough
- Stability is easily controlled

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Feasible performance analysis

Coefficient Update: Desired Properties

Corrections should reduce mean-square error

$$\xi(n) = E\left[\left|e(n)\right|^2\right]$$

 In a stationary environment, w_n should converge to Wiener-Hopf solution

$$\lim_{n\to\infty}\mathbf{w}_n=\mathbf{R}_x^{-1}\mathbf{r}_{dx}$$

- Avoid explicit signal statistics for $\Delta \mathbf{w}_n$ if possible
 - "Built-in" estimation of statistics
- If non-stationary, filter should track the solution

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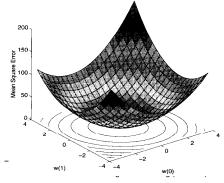
Adaptive Filtering

(Fig. from Hayes'

book Chapter 9)

Steepest-Descent Adaptive Filter

- Recall direct approach: minimizing the MSE by setting partial derivative = 0 (this may involve matrix inverse)
- => Alternative: search solution iteratively using a numerical method of steepest descent



book Chapter 9)

(Fig. from Hayes'

Adaptive Filtering

- - Find the filter coefficients that minimize the error on the error surface
 - At every iteration, moves along the direction of the steepest descent of error

Method of Steepest Descent

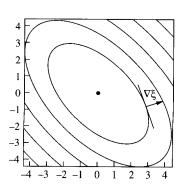
The steepest direction is given by gradient

$$\nabla \xi(n) = \begin{bmatrix} \frac{\partial \xi(n)}{\partial w(0)} \\ \frac{\partial \xi(n)}{\partial w(1)} \end{bmatrix}$$

Update Equation

$$\mathbf{w}_{n+1} = \mathbf{w}_n - \mu \nabla \xi(n)$$

 µ: step size, controls the rate at which the coefficients move



Adaptive Filtering

Method of Steepest Descent

- 1. Initialize with initial estimate w₀
- 2. Evaluate gradient $\xi(n)$ at time n
- Update estimate at time n by taking a step of size µ in negative gradient direction

$$\mathbf{w}_{n+1} = \mathbf{w}_n - \mu \nabla \xi(n)$$

4. Go back to 2.

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Stability of Steepest Descent

• It can be shown that

$$\mathbf{w}_{n+1} = \mathbf{w}_n + \mu E\{e(n)\mathbf{x}^*(n)\}$$

If x(n) and d(n) are w.s.s.,

$$\mathbf{w}_{n+1} = \mathbf{w}_n + \mu \big(\mathbf{r}_{dx} - \mathbf{R}_x \mathbf{w}_n \big)$$

- Correction term is zero if
$$\mathbf{W}_n = \mathbf{R}_x^{-1} \mathbf{r}_{dx}$$

(i.e., a fixed point of the update)

• Does the coefficient update converge?

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Adaptive Filtering [10]

Convergence of Steepest Descent

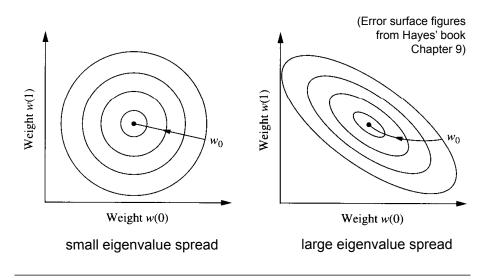
• For w.s.s. d(n) and x(n), the steepest descent adaptive filter converges to Wiener-Hopf solution

$$\lim_{n\to\infty}\mathbf{w}_n=\mathbf{R}_x^{-1}\mathbf{r}_d$$

if the step size satisfies $0 < \mu < \frac{2}{\lambda_{max}}$

- λ_{max} : maximal eigenvalue of \mathbf{R}_x
- Can be shown by diagonalizing ${\boldsymbol{\mathsf{R}}}_{{\boldsymbol{\mathsf{x}}}}$
- Convergence rate (how fast the update converges) is determined by the spread of eigenvalues

Effects of Condition Number on Convergence



Adaptive Filtering [11]

Adaptive Filtering

Least Mean Squares (LMS) Algotithm

• Recall the update in steepest descent:

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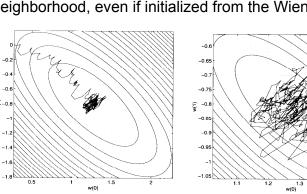
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 $\mathbf{w}_{n+1} = \mathbf{w}_n + \mu E \{ e(n) \mathbf{x}^*(n) \}$

- Practical challenge: the expectation may be unknown or difficult to estimate on the fly
- The LMS replaces the expectation by an "one-shot" estimate

 $\mathbf{w}_{n+1} = \mathbf{w}_n + \mu e(n) \mathbf{x}^*(n)$

- Very crude estimate, but often performs well in practice



Randomness of the LMS Algorithm

(Fig. from Hayes' book Chapter 9)

Adaptive Filtering

Adaptive Filtering

- The one-shot estimate approximates the steepest descent direction (i.e., the statistical average)
- The one-shot nature makes **w**_n move randomly in a neighborhood, even if initialized from the Wiener solution

LMS Algorithm for pth-order FIR Adaptive Filter

Parameters:
$$p =$$
 Filter order
 $\mu =$ Step sizeInitialization: $\mathbf{w}_0 = \mathbf{0}$ Computation:For $n = 0, 1, 2, ...$ (a) $y(n) = \mathbf{w}_n^T \mathbf{x}(n)$ (b) $e(n) = d(n) - y(n)$ (c) $\mathbf{w}_{n+1} = \mathbf{w}_n + \mu e(n) \mathbf{x}^*(n)$

(Algorithm from Hayes' book Chapter 9)

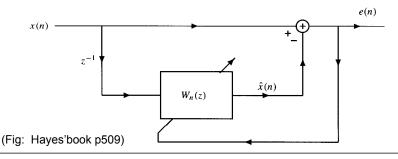
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Adaptive Filtering [14]

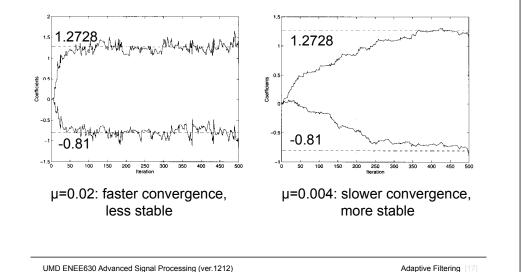
Example: Adaptive Linear Prediction

- x(n) = 1.2728 x(n-1) 0.81 x(n-2) + v(n)
 - v(n): unit-variance white noise
 - Optimum linear predictor: $\hat{x}(n) = 1.2728 x(n-1) 0.81 x(n-2)$

• Linear Prediction using LMS



Example: Adaptive Linear Prediction



Convergence of the LMS Algorithm

- Examine convergence properties of LMS under a statistical framework
- For w.s.s. d(n) and x(n) the LMS adaptive filter converges in the mean sense if

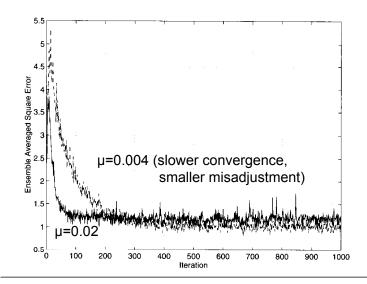
$$0 < \mu < \frac{2}{\lambda_{\max}}$$

• More stringent condition is required for convergence in mean square sense

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Adaptive Filtering [18]

Typical Learning Curves: MSE vs. time



Recursive Least Squares (RLS) Algorithm

- Mean-square error v.s. least squares error
 - Mean-square error does not depend on incoming data, but their ensemble statistics
 - Least squares error depends explicitly on x(n) and d(n)
- RLS: minimizes least squares error, where old data are gradually "forgotten"

$$\mathcal{E}(n) = \sum_{i=0}^{n} \lambda^{n-i} |e(i)|^2$$



Adaptive Filtering [20

Recursive Least Squares (RLS) Algorithm

• Least squares normal equation

$$\mathbf{R}_x(n)\mathbf{w}_n=\mathbf{r}_{dx}(n)$$

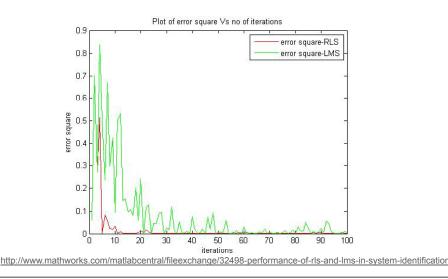
$$\mathbf{R}_{x}(n) = \sum_{i=0}^{n} \lambda^{n-i} \mathbf{x}^{*}(i) \mathbf{x}^{T}(i)$$
$$\mathbf{r}_{dx}(n) = \sum_{i=0}^{n} \lambda^{n-i} d(i) \mathbf{x}^{*}(i)$$

- $\mathbf{R}_{x}(n)$ and $\mathbf{r}_{dx}(n)$ can be calculated recursively
- R_x⁻¹(n) can also be calculated recursively using Matrix Inversion Formula

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Adaptive Filtering [22

Learning Rates of RLS and LMS



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Adaptive Filtering [23]

Characteristics of RLS Algorithm

- Convergence rate is an order of magnitude faster than that of LMS, at the cost of higher complexity
- Convergence rate is insensitive to eigenvalue spread
- In theory, RLS produces zero excess error or misadjustment
- RLS can be understood under the unifying framework of Kalman filtering

<u>Summary</u>

• Adaptive filters

- Address non-stationary signal processing
- Low computational complexity
- Recursive update of filter coefficients
- Method of steepest descent
 - Moves in the negative gradient direction
 - converges to Wiener-Hopf if stationary
- LMS algorithm
 - Crude one-shot gradient estimation; reasonable practical performance
- RLS algorithm: recursively minimizes least-squares error

Adaptive Filtering [24

<u>References for Further Explorations</u>

- M. Hayes, *Statistical Digital Signal Processing and Modeling*, Wiley, 1996. Chapter 9
 - All figures except one used in this lecture are from the book
- S. Haykin, *Adaptive Filter Theory*, 4th edition, Prentice-Hall, 2002, Chapters 4 & 5
- => See more detailed development in ENEE634 (offered in alternating spring semester)

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Adaptive Filtering [26]