4. Analog Adaptive Filters

Duration: Half Day

Tutorial Presenter:

- Prof. A. Chan Carusone, University of Toronto, Canada

Who Should Attend:

Researchers and designers of digital communication transceivers. This includes people working either at the circuit level (analog, digital, or mixed signal) or the system level. The material would also be of interest to adaptation algorithm researchers.

Abstract:

Adaptation is used whenever a filter's parameters must track poorly controlled or time varying conditions. At low speeds, adaptive filtering is easily and efficiently performed using digital circuits. However, analog filters are preferable when high speed, low power operation is required and moderate linearity can be tolerated. In these cases, the combination of flexibility and performance offered by analog adaptive filters can be an enabling technology. This tutorial will provide a overview of the area, including the algorithms, architectures, and circuits.

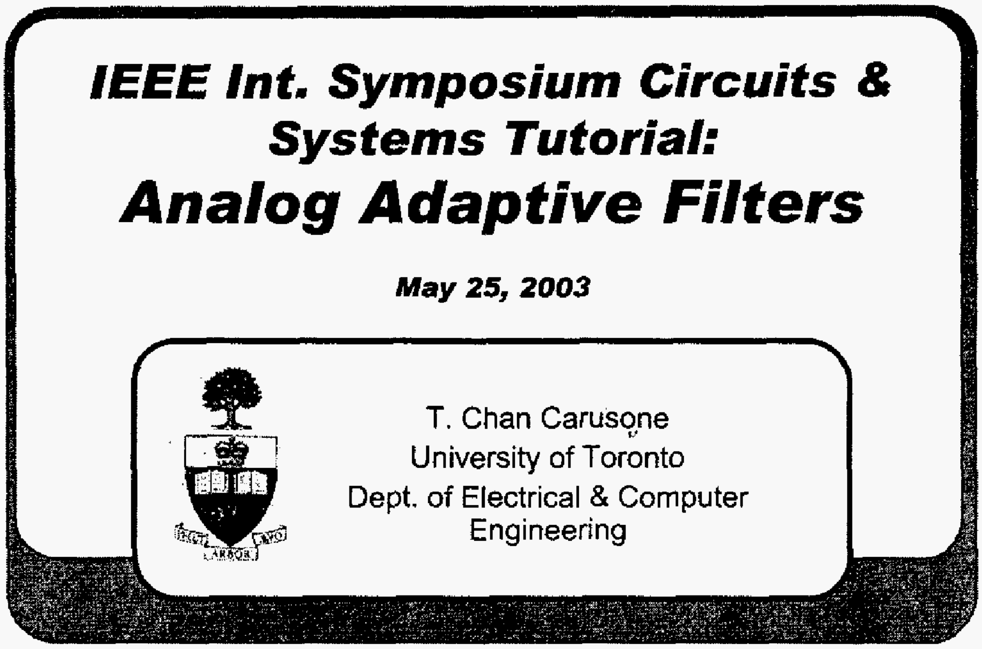
In the first part of the tutorial, analog filter structures suitable for adaptation are presented. Both analog discrete time (transversal, transpose, IIR) and continuous time (Laguerre and ladder filters) filter structures are covered including first order circuit designs.

The second part of the tutorial is a discussion of adaptation strategies suitable for analog filters. The traditional LMS algorithm has several problems when applied to analog adaptive filters. Algorithmic and circuit techniques for combating these problems will be discussed. Alternative adaptation algorithms will also be presented.

Finally, the third part of the tutorial brings together material presented in the first two sections by focusing on practical applications of analog adaptive filters. Both established applications and ongoing research areas will be considered including magnetic storage read channels, Ethernet transceivers, coaxial cable channels, backplanes, chip-to-chip connections, and optical communications. We shall see how the applications dictate the designers' choice of adaptation strategy, filter structure and circuit implementation.

Time Table

Tutorial 4: Analog Adaptive Filters	
13:30	Analog filter structures suitable for adaptation; analog discrete time (transversal, transpose, IIR) and continuous time (Laguerre and ladder filters) filter structures; first order circuit designs
14:30	Adaptation strategies for analog filters; algorithmic and circuit techniques
15:00	Coffee break
15:30	Alternative adaptation algorithms
16:00	Practical applications of analog adaptive filters; magnetic storage read channels, Ethernet transceivers, coaxial cable channels, backplanes, chip-to-chip connections, and optical communications
17:00	Close
18:00	Welcome reception



IEEE Int. Symposium Circuits & Systems Tutorial:
Analog Adaptive Filters

May 25, 2003

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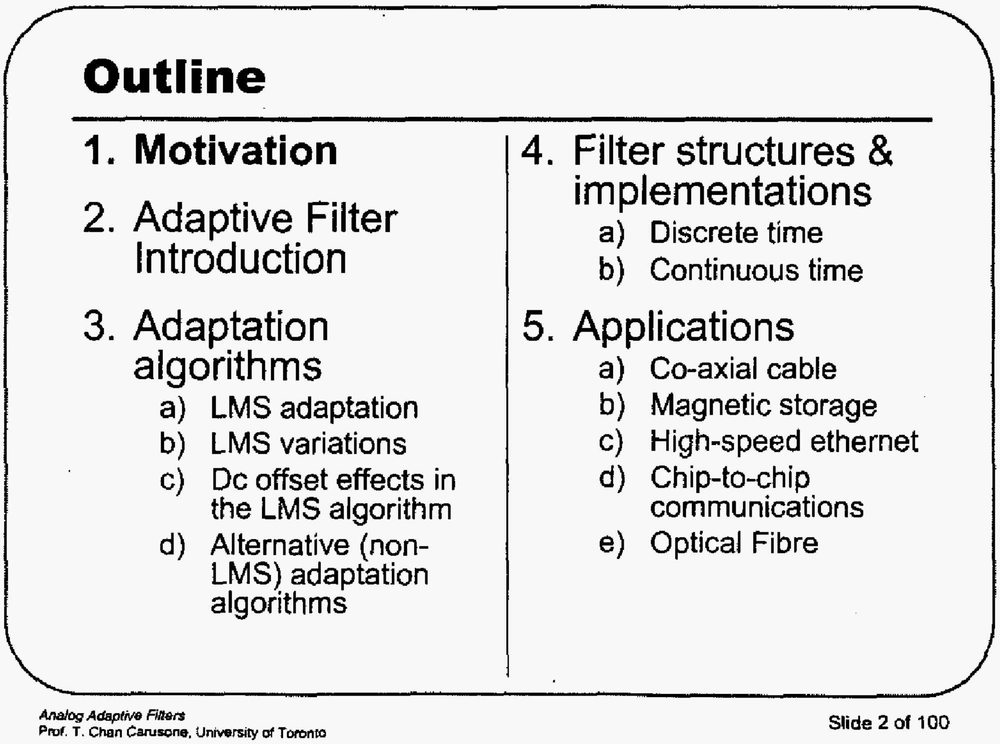
Dept. of Electrical & Computer Engineering



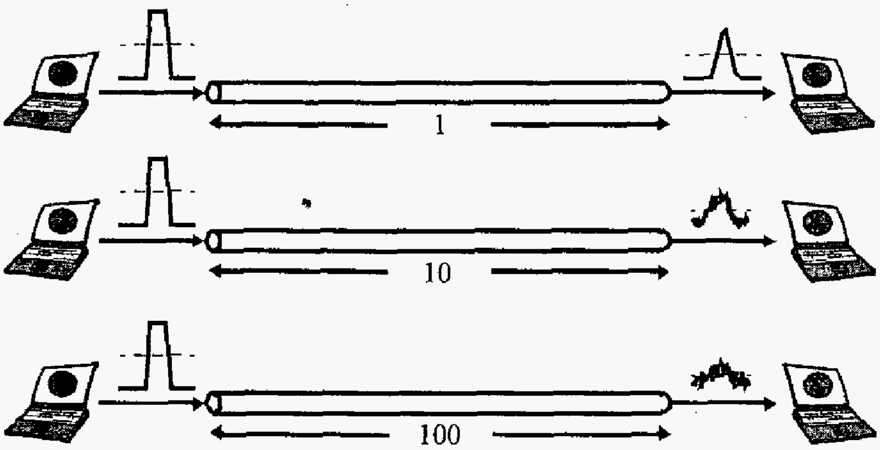
Outline

1. **Motivation**
2. Adaptive Filter Introduction
3. Adaptation algorithms
    a) LMS adaptation
    b) LMS variations
    c) Dc offset effects in the LMS algorithm
    d) Alternative (non-LMS) adaptation algorithms
4. Filter structures & implementations
    a) Discrete time
    b) Continuous time
5. Applications
    a) Co-axial cable
    b) Magnetic storage
    c) High-speed ethernet
    d) Chip-to-chip communications
    e) Optical Fibre

Digital Communications

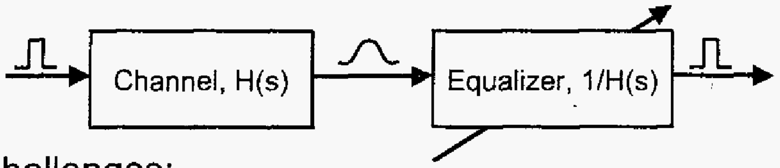
Why bother with adaptive filters at all?

- Channel impairments are unknown & time-varying



Equalization

- A communication channel changes the shape of pulses passing through it
- ➢ Intersymbol interference (ISI)


Channel, H(s)

Equalizer, 1/H(s)

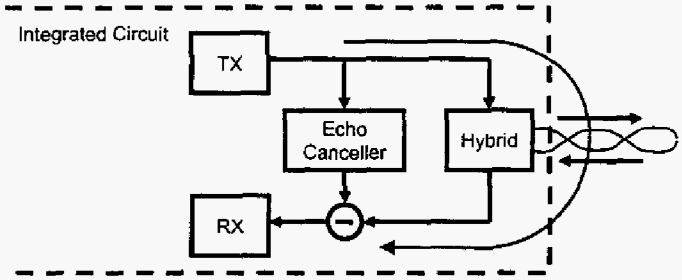
Challenges:

- only linear channels, H(s), can be equalized
- generally requires amplification, hence, noise enhancement
- channel H(s) may not be known *a priori*

  ⇒*Make equalizer adaptive*

Echo Cancellation

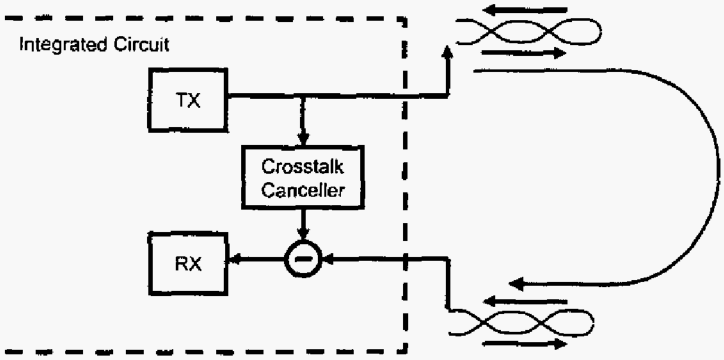
- *Full duplex*: transmit and receive over the same channel
- ➢ Some transmit signal leaks into receive path due to imperfect hybrid circuits
- Echo canceller output should mimic the echo signal


Integrated Circuit
TX
Echo Canceller
Hybrid
RX

- Challenges: similar to equalizer

Crosstalk Cancellation

- transmit signal from **neighboring channel** leaks into receive path
- ➢ Crosstalk canceller mimics the coupling path


Integrated Circuit
TX
Crosstalk Canceller
RX

Digital Communications

Why bother with ***analog*** adaptive filters?

- **Case #1**: Very high speed required
  - ⇒ A/D conversion not possible or practical
- **Case #2**: Moderate linearity & low power required
  - ⇒ Possible power savings resulting from reduction in A/D resolution and/or DSP required
  - ⇒ Improvement may be debateable as CMOS process technologies scale

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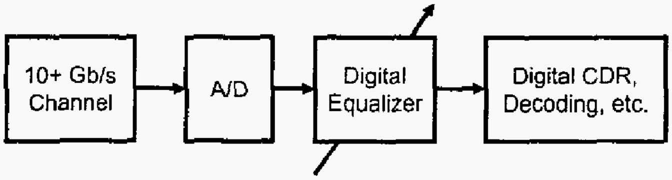
Digital Communications

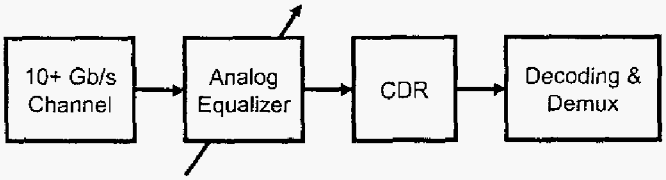
Example #1: Adaptive equalization of 10+ Gb/s data



- A/D must operate at 10+ GSamples/sec with several bits accuracy
  - ⇒ huge power consumption
  - ⇒ may not even be feasible, especially in CMOS

Digital Communications

Example #1: Adaptive equalization of 10+ Gb/s data

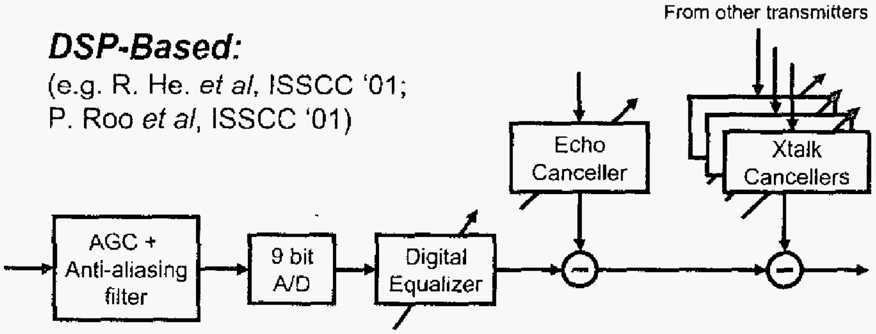


- no need for A/D
- may be implemented at the transmitter

Analog vs. Digital Adaptive Filters

Example #2: Signal conditioning in Gigabit ethernet

DSP-Based:
(e.g. R. He. *et al*, ISSCC '01;
P. Roo *et al*, ISSCC '01)

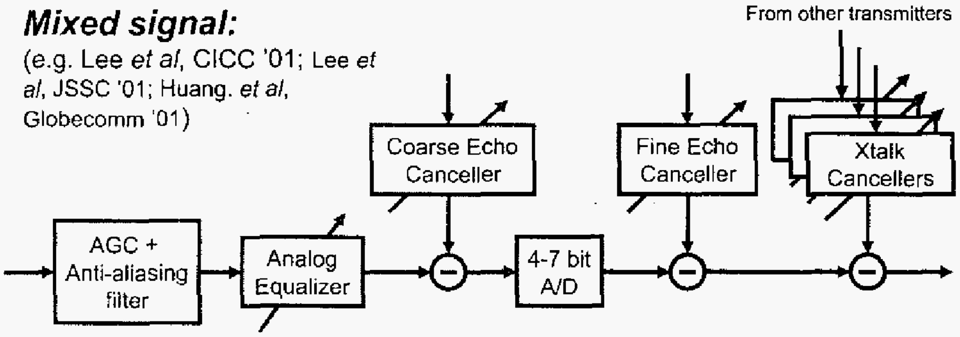


- Lots of high-speed DSP ⇒ power-hungry
- Considerable analog front end still required
- 9-bit A/D design is challenging

Analog vs. Digital Adaptive Filters

Example #2: Signal conditioning in Gigabit ethernet

Mixed signal:
(e.g. Lee *et al*, CICC '01; Lee *et al*, JSSC '01; Huang. *et al*, Globecomm '01)



From other transmitters

AGC + Anti-aliasing filter

Analog Equalizer

Coarse Echo Canceller

4-7 bit A/D

Fine Echo Canceller

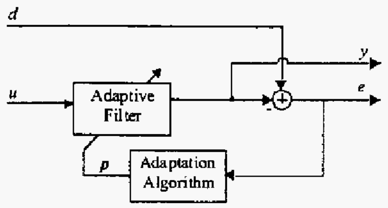
Xtalk Cancellers

- Analog equalizer may be combined with anti-aliasing filter
- Reduced power in both DSP and A/D
- Overall analog complexity may actually be *reduced*

Outline

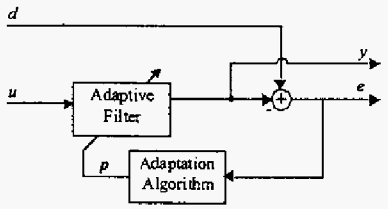
1. Motivation
2. **Adaptive Filter Introduction**
3. Adaptation algorithms
4. Filter structures & implementations
5. Applications

Adaptive Filter Introduction



- Adaptive filter has input *u* and output *y*
- The filter has programmable parameters, ***p*** (e.g. transfer function coefficients, location of poles, zeros, etc…)
- The filter's "desired" or "ideal" output is known, *d*
- Adaptation algorithm adjusts ***p*** to minimize the power in the error, *e* = *d* – *y*

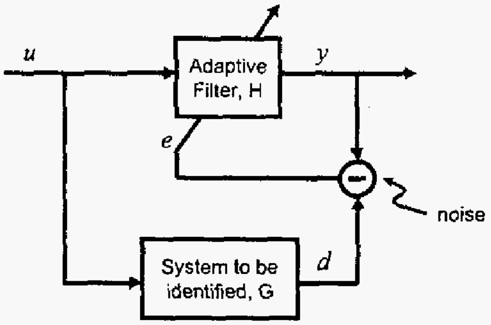
Adaptive Filter Introduction



Alternate interpretation:

- Adaptation algorithm adjusts ***p*** so that *y* is the best possible estimate of *d*

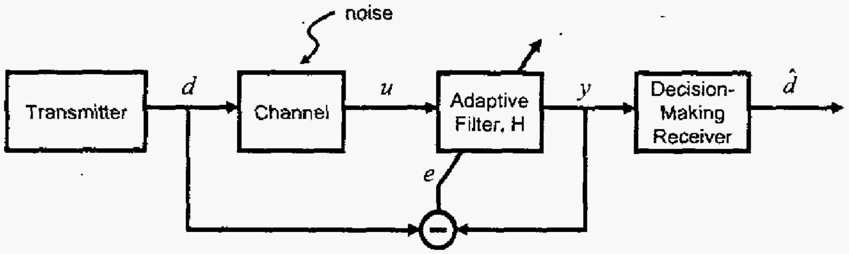
System Identification Model



- Define:
  - u = filter input
  - y = filter output
  - d = desired output
  - e = error in output with respect to desired output = $(d - y)$

➢ After convergence, H will match G as closely as possible ⇒ can then use H as a model for G (a.k.a. "model matching")

Example: Equalization

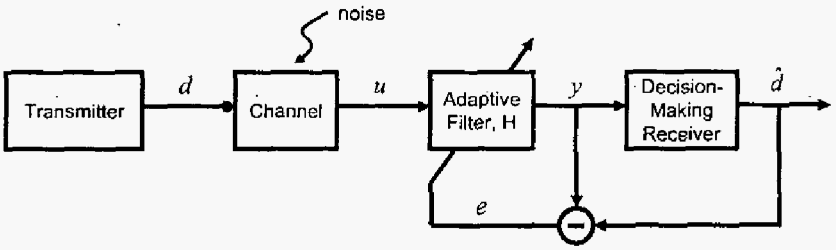


- Desired output, d = transmitted waveform

➢ Must have access to the transmitted data at the receiver in order to generate e (e.g. training sequence)

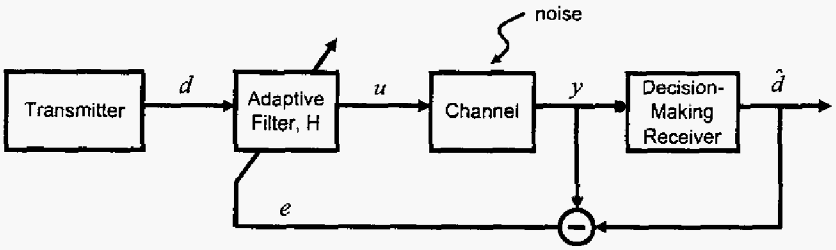
- Similar to the system identification problem except the system to be identified, G = $(\text{channel})^{-1}$

Decision-Directed Equalization



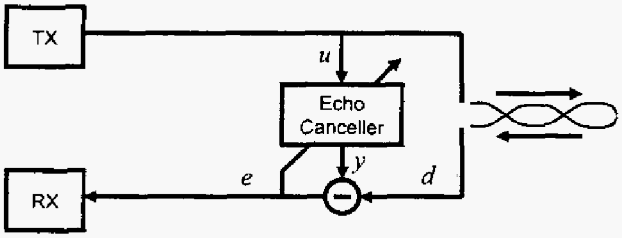
- Instead, may use receiver's decisions as the "desired" signal
  - "Decision-directed" operation
  - Assumes fairly low bit error rate (BER)
  - May use training sequence at the start to decrease BER, then switch to decision-directed

Adaptive Pre-emphasis



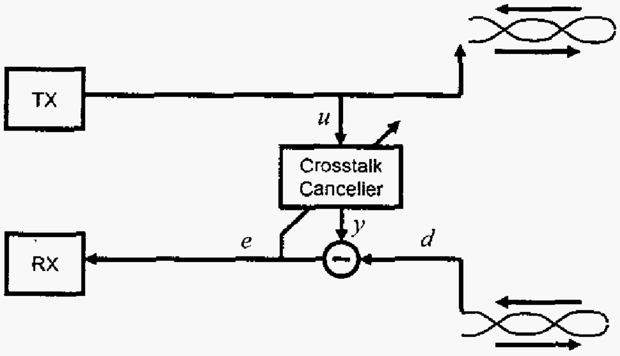
- (Theoretically) it does not matter what order you apply the adaptive filter and the channel
  - Sometimes it is beneficial to apply adaptive filtering at the transmitter: "Pre-emphasis" (usually high frequency components are amplified/emphasized prior to transmission)

Example: Echo Cancellation



- Recall:
  - u = filter input
  - y = filter output
  - d = desired output
  - e = error in output with respect to desired output = $(d-y)$

Example: Crosstalk Cancellation



- Recall:
  - u = filter input
  - y = filter output
  - d = desired output
  - e = error in output with respect to desired output = $(d-y)$

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    a) **LMS adaptation**
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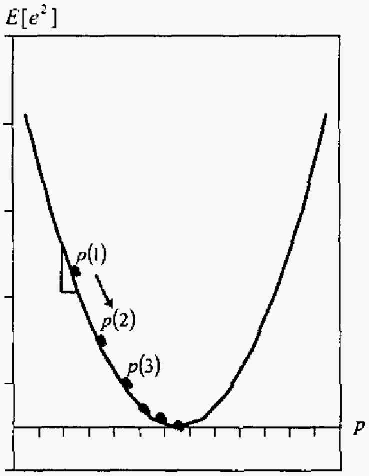
Gradient Descent Adaptation

- 1 parameter case:

$$\frac{\partial E\left[e^2\right]}{\partial p} < 0 \qquad \therefore\ \Delta p > 0$$

$$\Rightarrow \quad p(k+1) = p(k) + \Delta p = p(k) - \mu \frac{\partial E\left[e^2(k)\right]}{\partial p}$$

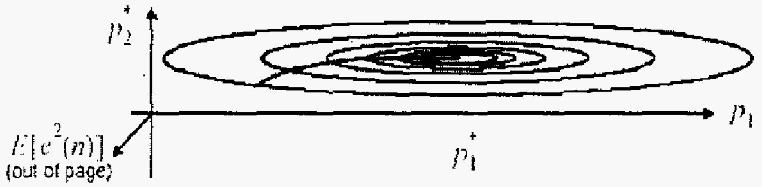
- μ determines the rate of adaptation: larger μ ⇒ faster convergence



Gradient Descent Adaptation

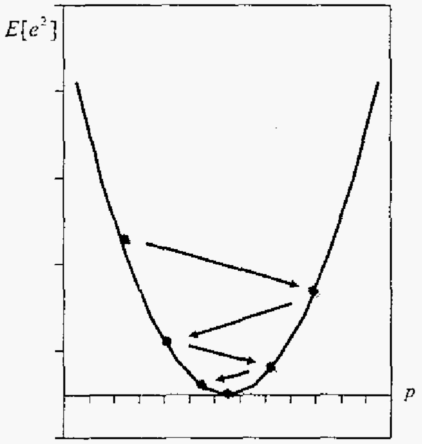
- Multiple parameters: Each is updated independently (simultaneously) depending on its own partial derivative
- e.g. 2 parameters:

$$p_1(k+1) = p_1(k) - \mu\frac{\partial E\left[e^2(k)\right]}{\partial p_1} \qquad p_2(k+1) = p_2(k) - \mu\frac{\partial E\left[e^2(k)\right]}{\partial p_2}$$

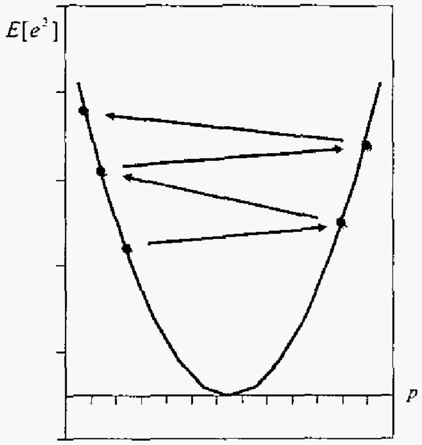


Gradient Descent Adaptation

- μ must also be chosen to ensure stability



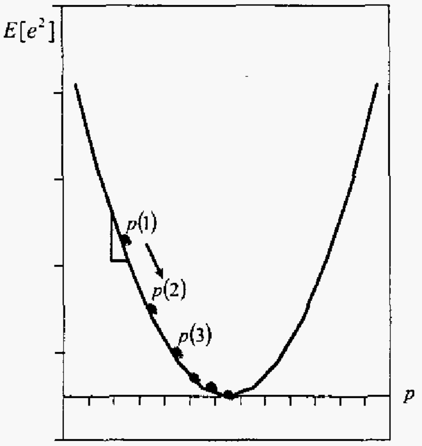
μ underdamped, but still stable



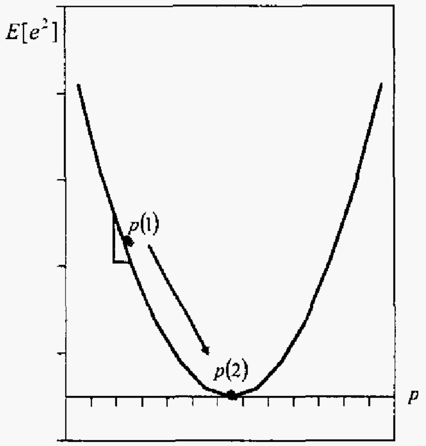
μ too large ⇒ unstable

Gradient Descent Adaptation

- Usually, μ is chosen conservatively for smooth "overdamped" convergence



μ overdamped ⇒ smooth convergence



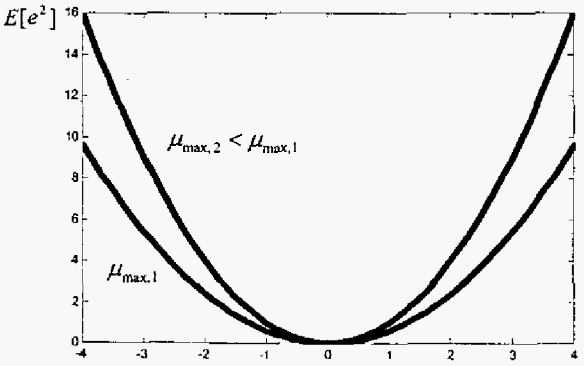
μ critically damped ⇒ requires a knowledge of $E[e^2]$ vs. p (not usually available)

Gradient Descent Adaptation

- μ_{max} = maximum μ for which adaptation is still stable

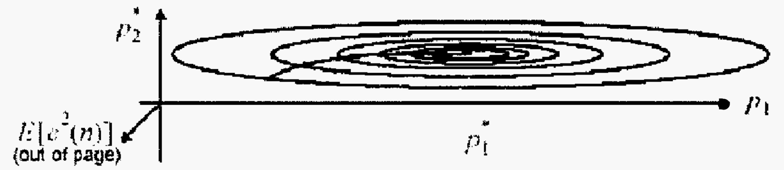
> will depend upon the shape of the performance surface

- "shallow" bowl ⇒ larger μ_{max} since the slope $\delta E[e^2(k)]/\delta p$ is always smaller



Gradient Descent Adaptation

- With multiple parameters, must choose μ conservatively for worst-case (steepest) parameter
- In this case, using the same μ for both parameters results in a μ that is slower than necessary for p_1



Gradient Descent Adaptation

- How can we calculate $\partial E[e^2]/\partial p$?
  - Operate filter at ($p + \Delta p$) and ($p - \Delta p$) for some period of time, T_0
  - Average e^2 over T_0 to estimate E[e^2] at ($p + \Delta p$) and ($p - \Delta p$)
  - Approximate: $\frac{\partial E[e^2]}{\partial p} \approx \frac{E[e^2]\big|_{p+\Delta p} - E[e^2]\big|_{p-\Delta p}}{\Delta p}$
  - Problem: only becomes accurate for large $T_0 \Rightarrow$ slow/inaccurate

LMS Algorithm

- How can we calculate $\partial E\left[e^2\right]/\partial p$?
  - Assume that $E[e^2(k)] \approx e^2(k)$

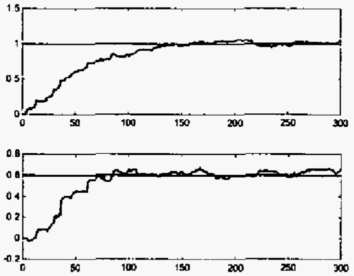
$$\therefore \frac{\partial E\left[e^2\right]}{\partial p} \approx \frac{\partial\left(e^2\right)}{\partial p} = \frac{\partial\left(e^2\right)}{\partial e}\cdot\frac{\partial e}{\partial p} = 2e\cdot\frac{\partial(d-y)}{\partial p} = -2e\cdot\frac{\partial y}{\partial p} \equiv -2e\phi$$

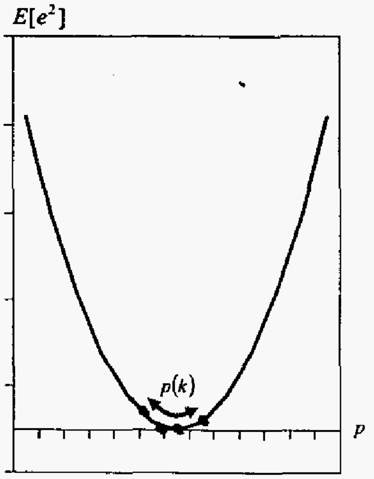
  - ϕ is the “gradient signal” for parameter p
  - Substitute this back into general gradient descent method:

$$\boxed{p(k+1) = p(k) + 2\mu e(k)\phi(k) \quad \textbf{LMS Algorithm}}$$

LMS Algorithm

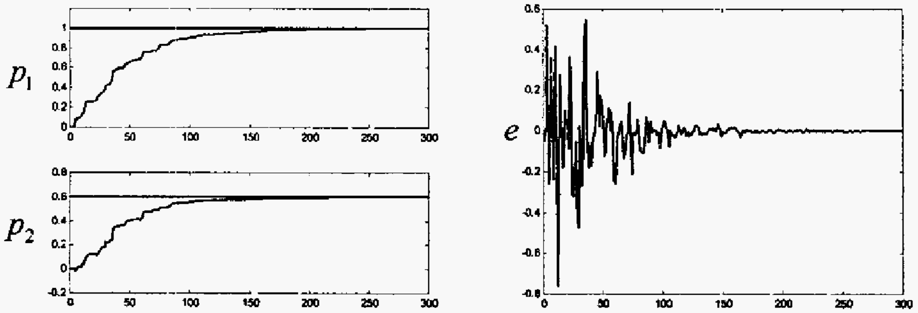
- **Problem**: What is the effect of the approx: $\delta E[e^2(k)]/\delta p \approx -2e\phi$?
- After convergence, $\delta E[e^2(k)]/\delta p = 0$ but $-2e\phi \neq 0$
- So, $\Delta p(k) \neq 0$ and parameters “bounce” around their optimal values





LMS Algorithm

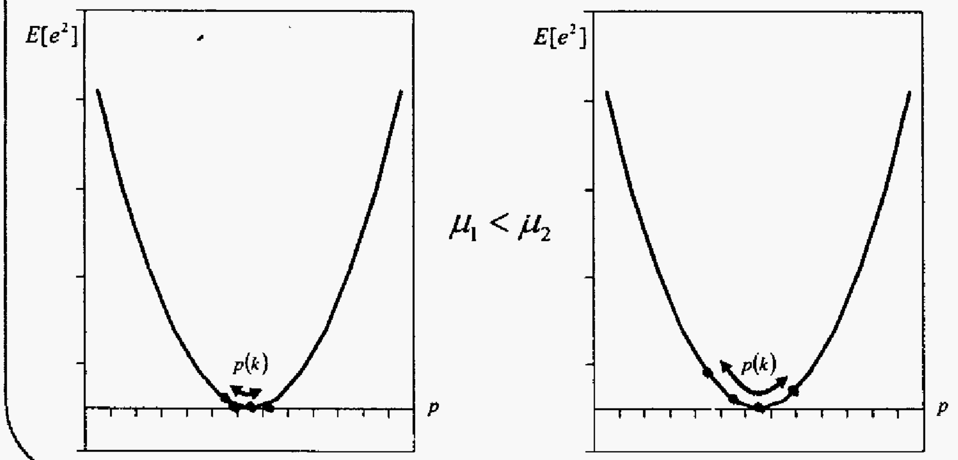
- **Exception:** If the adaptive filter can be made to match the "target" exactly $\Rightarrow y = d \Rightarrow e = 0 \Rightarrow -2e\phi = 0 \Rightarrow \Delta p(k) = 0$



- Never occurs in practice

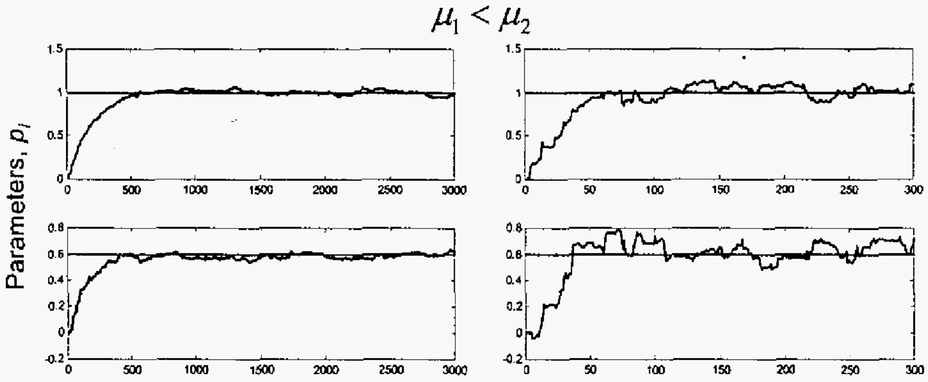
LMS Algorithm

- The amount of "bounce" can be reduced by decreasing μ
- The amount of bounce is the "***misadjustment***"



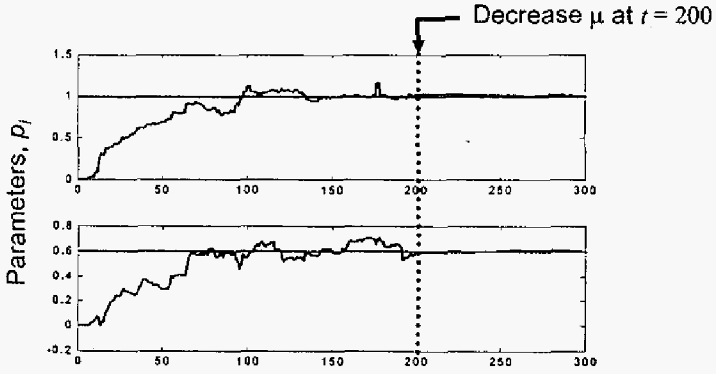
LMS Algorithm

- Tradeoff between speed and accuracy of convergence in choice of μ



LMS Algorithm

- Compromise can be reached by employing "gear-shifting" approach



Decrease μ at $t = 200$

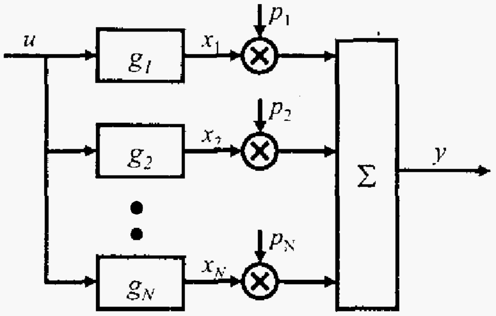
Obtaining Gradient Signals

- **Problem**: How can we obtain ϕ?

➢ Depends upon the filter structure employed:

  - Adapt only zeros while keeping poles fixed $\Rightarrow$ "adaptive linear combiner" (usual case)
  - Adapt poles and zeros $\Rightarrow$ "adaptive IIR filtering" (challenging!)

Adaptive Linear Combiner (ALC)



- Each block, g_i, represents a fixed LTI system
- Adapted parameters are the coefficients, p_i
- Changing p_i's only effects the filter's zeros (not the poles) $\Rightarrow$ always stable
- Overall impulse response & transfer function from u to y:

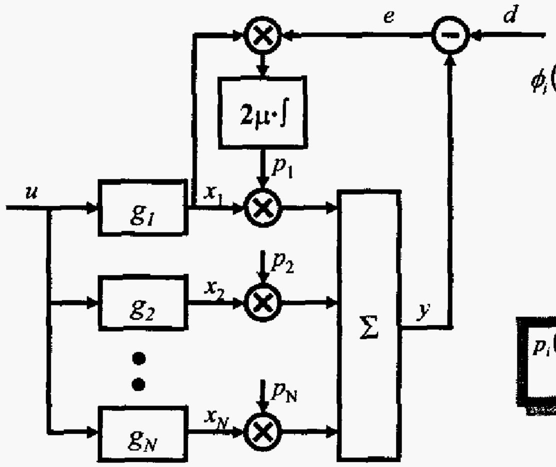
$$y(k) = p_1 x_1(k) + \cdots + p_N x_N(k) = \sum_{i=1}^{N} p_i x_i(k)$$

$$h(k) = p_1 g_1(k) + \cdots + p_N g_N(k) = \sum_{i=1}^{N} p_i g_i(k)$$

$$H(z) = \sum_{i=1}^{N} p_i G_i(z)$$

Adaptive Linear Combiner

- **Problem**: How can we obtain ϕ?



$$\phi_i(t) \equiv \frac{\partial y(t)}{\partial p_i} = \frac{\partial}{\partial p_i}\left[p_1x_1(t)+\cdots+p_Nx_N(t)\right] = x_i(t)$$

LMS for ALC

$$\boxed{\begin{aligned} p_i(k+1) &= p_i(k)+2\mu e(k)\phi_i(k)\\ &= p_i(k)+2\mu e(k)x_i(k)\end{aligned}}$$

Adaptive Linear Combiner

- Matrix notation: $\bar{p} = \begin{bmatrix} p_1 \\ p_2 \\ \vdots \\ p_N \end{bmatrix}$, $\bar{x} = \begin{bmatrix} x_1 \\ x_2 \\ \vdots \\ x_N \end{bmatrix}$ $\Rightarrow \bar{p}(k+1) = \bar{p}(k) + 2\mu e(k)\bar{x}(k)$

Let, $\xi = E[e^2]$, Note: $\xi = \xi(\bar{p})$ is the "performance surface"

Let, $\xi(\bar{p}^*) = \xi_{\min}$, where $\bar{p}^*$ are the optimal ALC coefficients

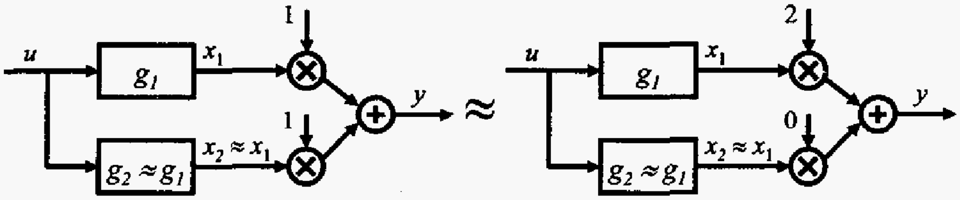
- It can be shown that:

$$\xi(\bar{p}) = \xi_{\min} + (\bar{p} - \bar{p}^*)R(\bar{p} - \bar{p}^*)^T$$

- Where R is the $N \times N$ state correlation matrix: $R_{ij} = E[x_i(k) \cdot x_j(k)]$
- ξ is a N-dimensional paraboloid

Adaptive Linear Combiner

- The shape of $\xi(p)$ is related to the state correlations, $E[x_i(k) \cdot x_j(k)]$
- Independent states ⇒ a "round bowl" ⇒ fast convergence of all parameters
- Dependent states ⇒ "ill-conditioned" performance surface ⇒ poor convergence
- **e.g.** 2 parameters with highly dependent states:



Adaptive Linear Combiner

- ALCs have several advantages:

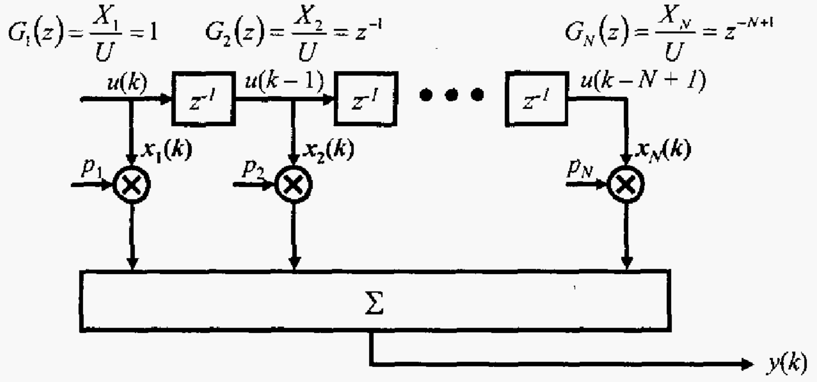
✓ Stability of the filter is guaranteed because the pole locations are fixed

✓ The MSE is a parabolic function of the parameters; hence, there is only one local minimum; stability of the adaptation procedure is guaranteed using a gradient descent algorithm (such as LMS)

✓ Relatively easy to obtain the gradient signals, ϕ, required for LMS adaptation

Analog Transversal Filters

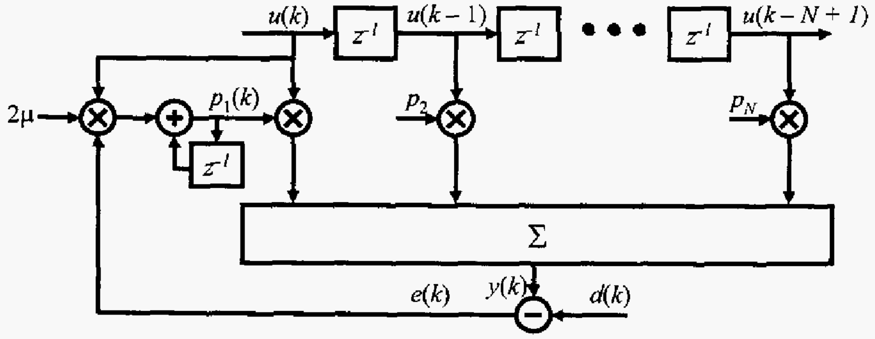
- **Example:** Analog transversal filter



- N-parameter FIR filter

Analog Transversal Filters

- If $u(k)$ is white, $E[u(k)u(k-i)]=0,\ \forall i\neq 0 \Rightarrow E[x_i(k)x_j(k)]=0,\ \forall i\neq j$
  - ➢ The states are uncorrelated
- **LMS adaptive transversal filter:**



Adaptive IIR Filtering

- In general, one may adapt the zeros **and poles** of a filter

✓ May provide better performance with fewer adapted parameters

× Poles may become unstable

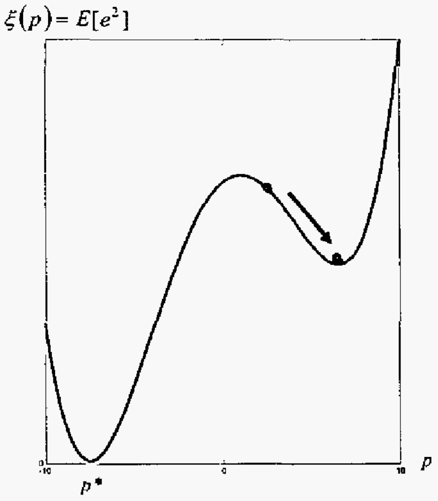
× Convergence to the optimum filter parameters is not guaranteed using gradient descent techniques (e.g. the LMS algorithm)

× More difficult to obtain the gradient signals, ϕ, for LMS adaptation

Adaptive IIR Filtering

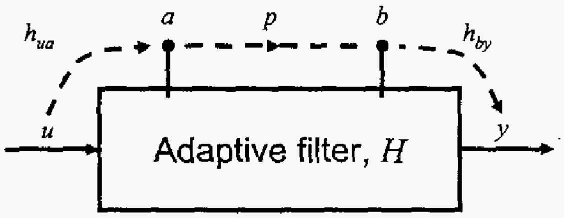
- "Multi-modal" performance surface
- Steady-state parameters depend upon the initial conditions
- No guarantee that you will converge to the optimal parameters
- Note: For ALC, you are guaranteed a (N-dimensional) parabolic performance surface ⇒ "unimodal"

$$\xi_{\text{ALC}}(\bar{p}) = \xi_{\min} + (\bar{p} - \bar{p}^*)R(\bar{p} - \bar{p}^*)^T$$



Adaptive IIR Filtering

- **Problem**: How can we calculate ϕ for an IIR filter parameter?

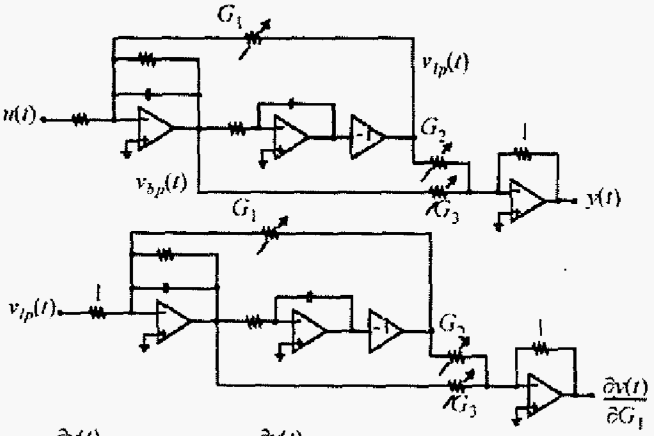


$$\begin{aligned} y &= b * h_{by} + [\text{other terms that don't depend on } b] \\ &= p \cdot a * h_{by} + \ldots \\ &= p \cdot u * h_{ua} * h_{by} + \ldots \end{aligned}$$

$$\therefore \frac{\partial y}{\partial p} = u * h_{ua} * h_{by}$$

Adaptive IIR Filtering

- Example:

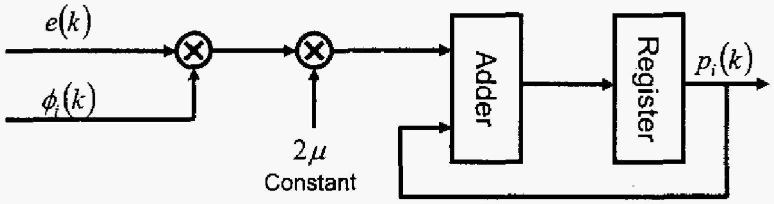


$$\frac{\partial y(t)}{\partial G_2} = -v_{lp}(t) \qquad \frac{\partial y(t)}{\partial G_3} = -v_{bp}(t)$$

LMS Algorithm Implementation

- LMS update rule can be implemented using either analog or digital circuits
- For digital filters, use digital LMS:

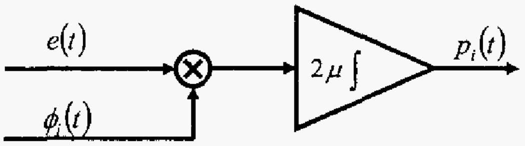
$$p_i(k+1) = p(k) + 2\mu e(k)\phi_i(k)$$



LMS Algorithm Implementation

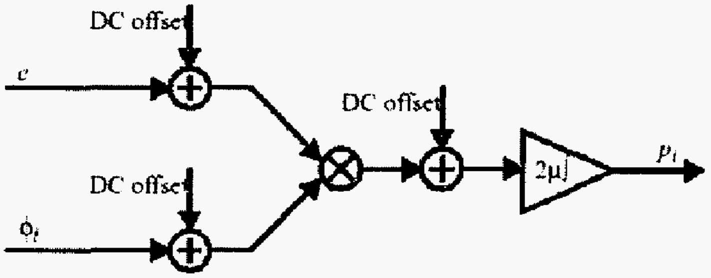
- For analog filters, analog implementation may seem natural/obvious:

$$p_i(t) = 2\mu \int_0^t e(\tau)\phi_i(\tau)d\tau$$



LMS Algorithm Implementation

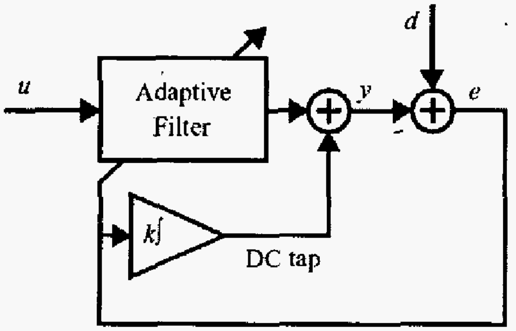
- **Problem**: dc offsets on the gradient and error signals hinder the adaptation ⇒ always present on analog signals



LMS Algorithm Implementation

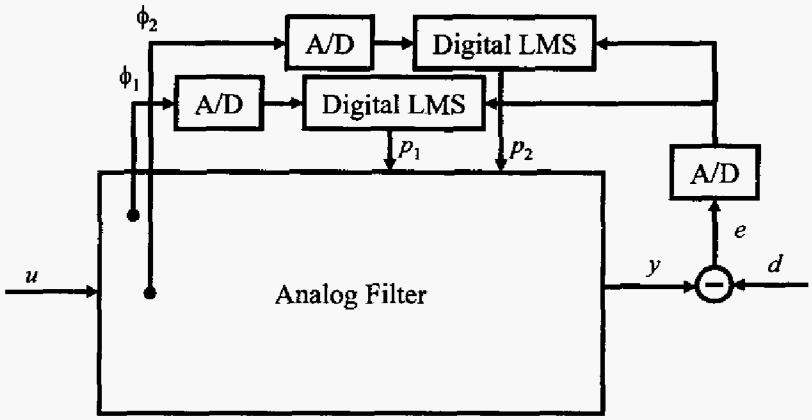
- **Solution**: dc offsets on the error signal can be eliminated by a dc tap

➢ Other sources of offset remain



LMS Algorithm Implementation

- **Solution**: use digital adaptation
- **Problem**: too much hardware!



LMS Algorithm Implementation

- **Solution**: use simplified adaptation algorithms
- Any monotonic function can be applied to e or ϕ_i and still move p_i in the correct direction

$$\text{LMS: } p_i(k+1) = p_i(k) + 2\mu \cdot e(k) \cdot \phi_i(k)$$

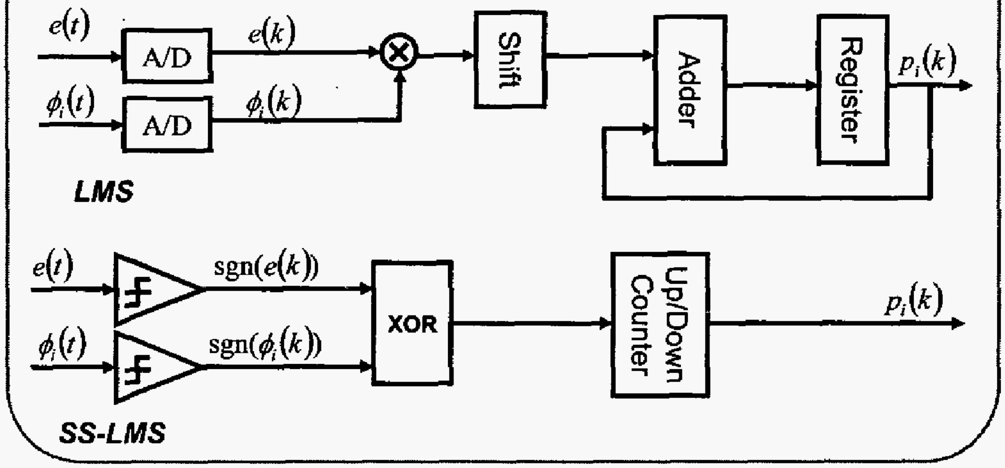
$$\text{Sign-data LMS: } p_i(k+1) = p_i(k) + 2\mu \cdot e(k) \cdot \operatorname{sgn}(\phi_i(k))$$

$$\text{Sign-error LMS: } p_i(k+1) = p_i(k) + 2\mu \cdot \operatorname{sgn}(e(k)) \cdot \phi_i(k)$$

$$\text{Sign-sign LMS: } p_i(k+1) = p_i(k) + 2\mu \cdot \operatorname{sgn}(e(k)) \cdot \operatorname{sgn}(\phi_i(k))$$

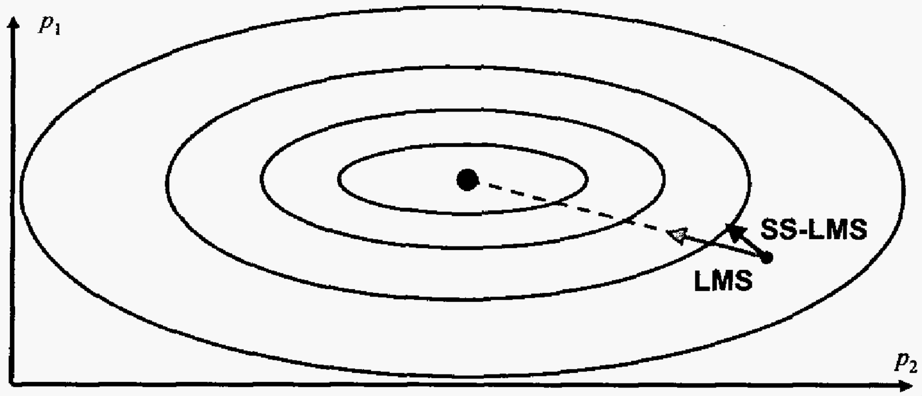
LMS Algorithm Implementation

- **Much** simpler hardware



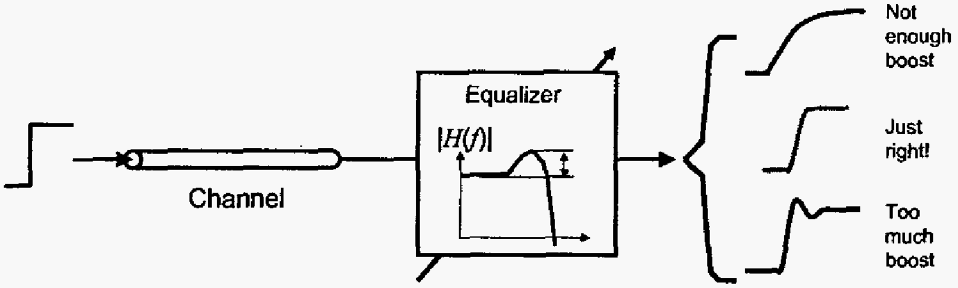
LMS Algorithm Implementation

- **Problem**: slower adaptation and risk of instability
- **Example**: Gradient estimates for LMS & SS-LMS



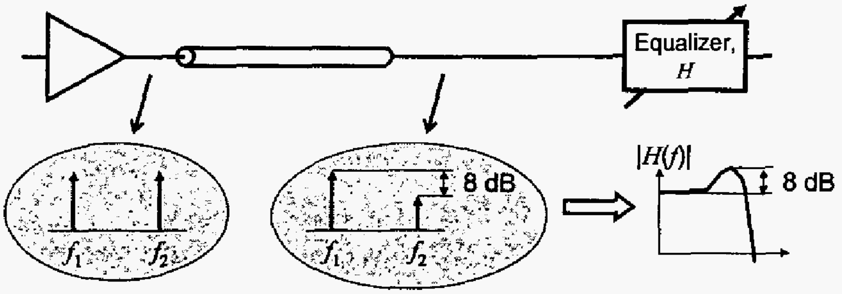
Alternate Adaptation Algorithms

- **Example #1**: look at transitions and adjust the amount of high-freq boost in the equalizer for,
  - zero overshoot/undershoot [Everitt '98]
  - certain maximum slope [Shoval '95]



Alternate Adaptation Algorithms

- Due to these difficulties, some people have implemented other adaptation approaches
- **Example #2**: Look at relative attenuation of high/low frequencies and program equalizer to compensate for the difference



Alternate Adaptation Algorithms

Example #3: “Zero-Forcing” algorithm [Lucky, ’65]

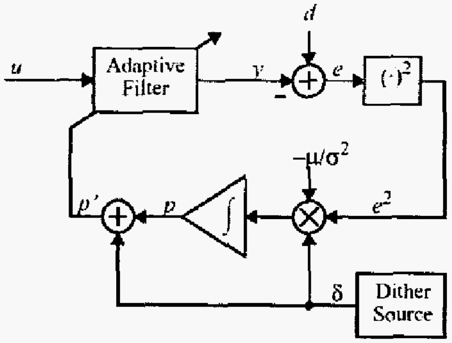
$$p(k+1) = p(k) + 2\mu e(k) y(k)$$

✓ Does not require access to gradient signals

× Does not minimize mean-squared error ⇒ causes noise enhancement

Alternate Adaptation Algorithms

- **Example #4**: Dither parameters and look for correlated changes in the output squared error [Chan Carusone, ’02]

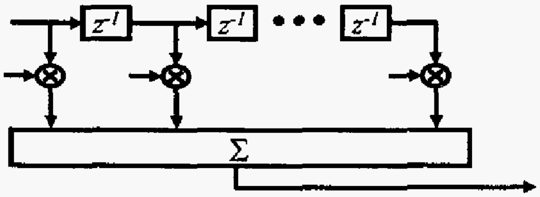


Outline

1. Motivation
2. Adaptive Filter Introduction
3. Adaptation algorithms
4. **Filter structures & implementations**
    - **a) Discrete time**
        - **i. Transversal**
        - **ii. Other**
    - **b) Continuous time**
        - **i. Transversal**
        - **ii. Cascade of biquads**
        - **iii. Ladder**
5. Applications

Discrete Time Filters

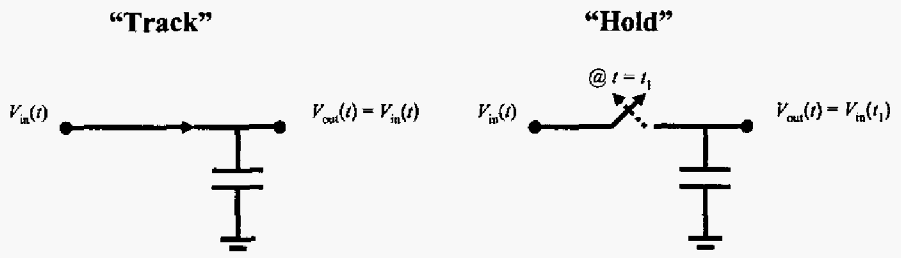
- Discrete-time basic building blocks:
    1. Delay stage
    2. Multiply or Programmable Gain
    3. Summation
- Example: Transversal Filter



Discrete Time Filters

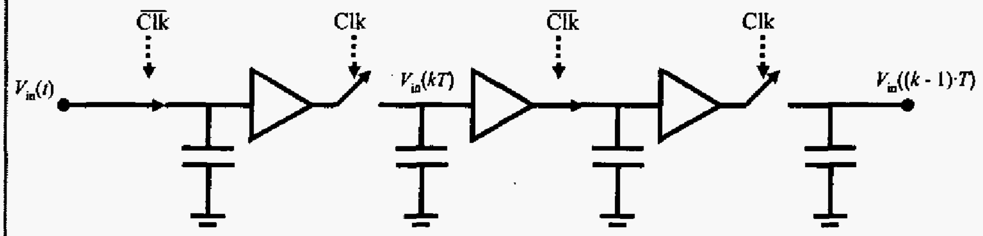
1. Delay stage: sample and hold circuits

- Based on holding a voltage on a capacitor:



Analog Filter Implementation

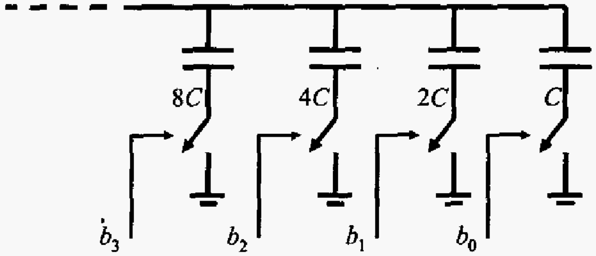
- Can create a delay-line using this approach:



Discrete Time Filters

2. Programmable Gain: can be realized by varying the size of the holding capacitors

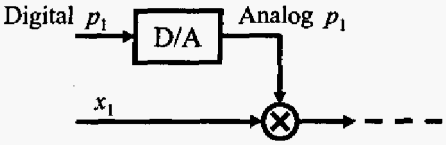
- If capacitors are digitally programmable, gain can be digitally programmable [Gomez, '93]



Discrete Time Filters

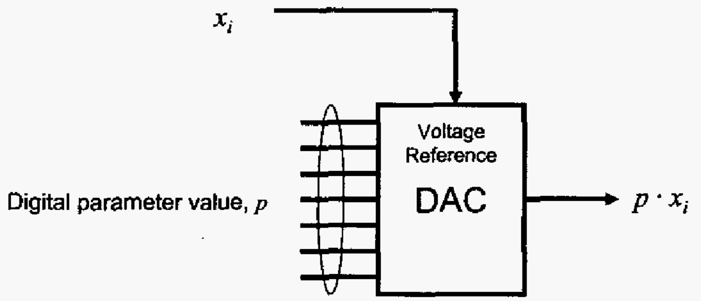
2. Programmable Gain: alternately, can multiply 2 analog quantities together via a Gilbert cell

- Requires the filter parameter value to be an analog voltage [Kiriaki, '97]



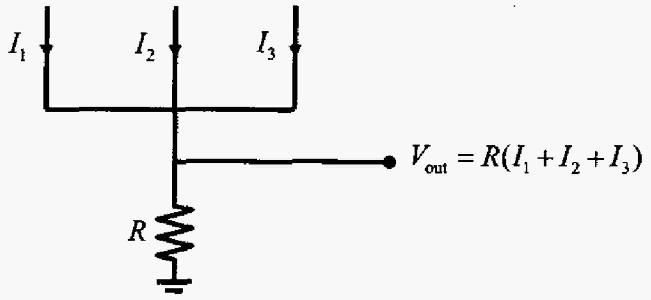
Discrete Time Filters

2. Programmable Gain: can use a multiplying digital-to-analog converter, MDAC



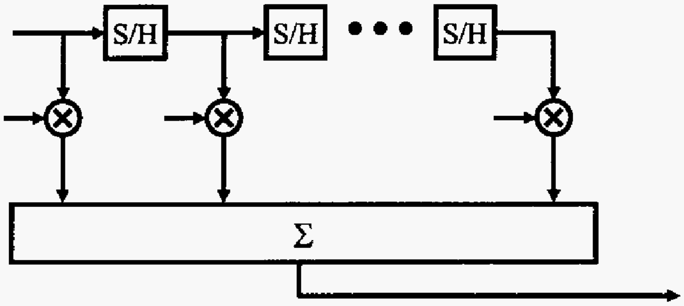
Discrete Time Filters

3. Summation: easiest and fastest way is to use current signals, then simply tie them together



Analog Transversal Filter

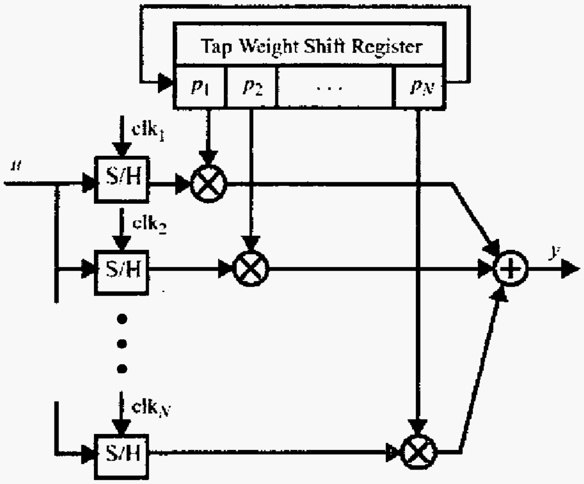
- Straightforward implementation:



- **Problems**:
  - DC offsets, noise, etc. introduced by each S/H accumulate as the signal progresses along the chain
  - Each S/H must operate at the full sampling rate

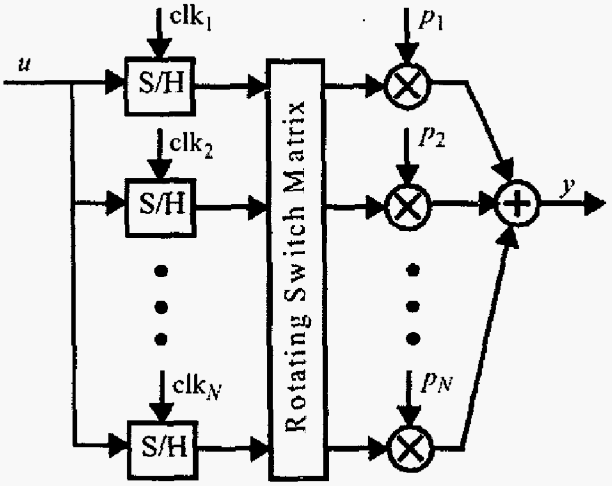
Analog Transversal Filter

- **Solution**: rotating tap weights [Haque '77], [Ahuja '79]
- **Problem**: at high speed, the dynamic power consumption of shifting the tap coefficients around becomes large



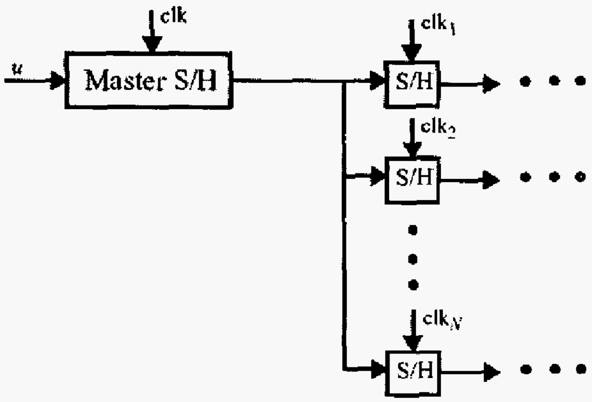
Analog Transversal Filter

- **Solution**: rotating switch matrix [Sonntag '95]
- **Problem**: mismatch between different S/Hs show up as fixed pattern noise at the output



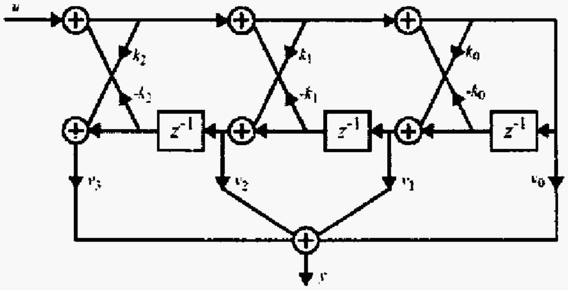
Analog Transversal Filter

- **Solution**: master S/H at front end [Kiriaki '97]
- **Problem**: S/H must operate at the full sampling rate



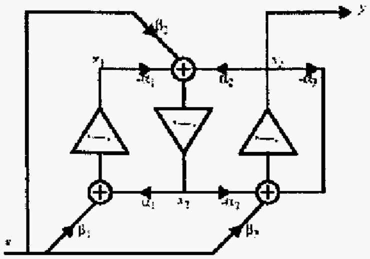
Discrete Time Filters

- Other discrete time structures possible
- **Example:** lattice filters have desirable properties for adaptive filters
- IIR, but stability can be guaranteed by ensuring $|k_i| < 1$
- Same analog building blocks
- Have not been common in practice



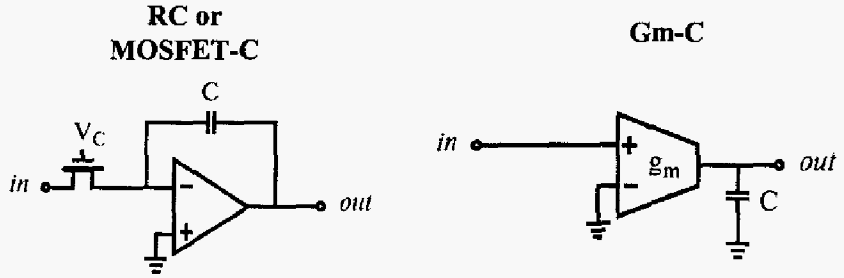
Continuous Time Filters

- Active filters preferred for integrated circuits
- Continuous time filter building blocks:
  - Integrator
  - Multiplier or Programmable Gain
  - Summation: usually use current summation for high-speed, as before



Continuous Time Filters

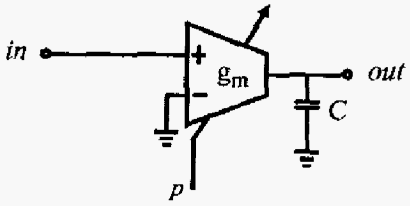
- **Integrators**: two main types
- In both cases, input voltage is converted into a current, then integrated on a capacitor



Continuous Time Filters

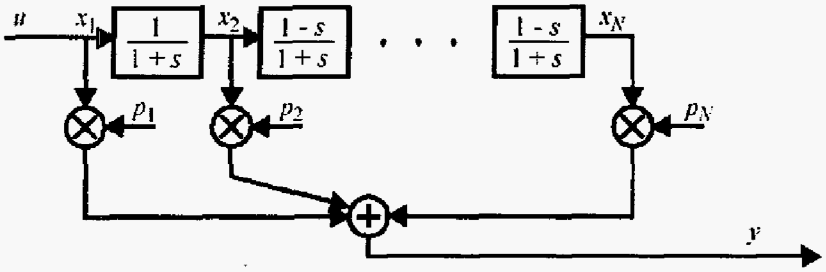
- **Multiplier / Programmable Gain**

1. Can use a Gilbert cell as before
2. Can use a programmable gain stage
   - may even combine with the integrator
   - Example: Gm-C integrator with programmable gain



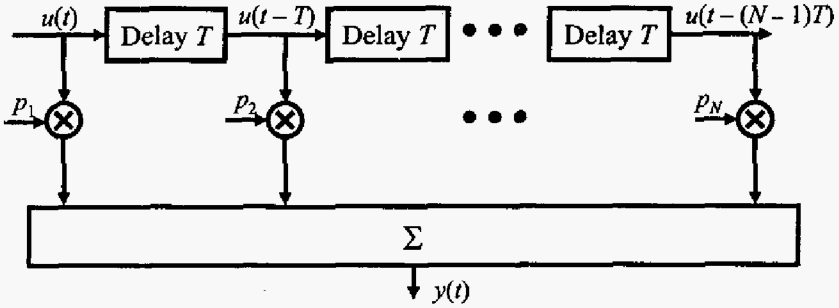
Continuous Time Filters

- Many options for filter structures
- **Example**: Laguerre Filter Structure
  - Special case of ALC (guaranteed stable, easy LMS adaptation, fixed zero locations, etc…)



Continuous Time Filters

- Essentially a continuous time tapped delay line [Sands, '96], [Parsi, '96]
- Instead of first-order allpass delay elements (as in Laguerre structure), one can also use Bessell allpass delay stages



Continuous Time Filters

- Could also use a biquad with adaptive ω_0 & Q (e.g. [Shoval '95]):

$$H(s)=\frac{a_2 s^2 + a_1 s + a_0}{s^2 + \frac{\omega_0}{Q} s + \omega_0^2}$$

✓ Easy to build

✓ Easy to adapt while maintaining stability (usually using "alternate" non-LMS algorithms)

× Limited to 2nd order transfer functions

  - ➢ Cascade adaptive biquads to get higher order transfer functions
  - × May have problems with dynamic range scaling

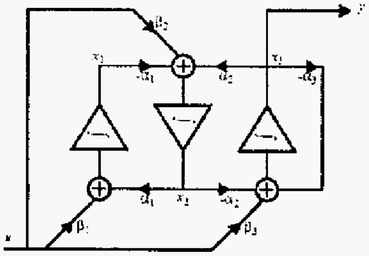
Continuous Time Filters

- Orthonormal ladder filter structure

✓Arbitrary rational transfer function

✓Good dynamic range scaling

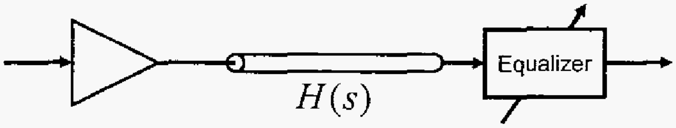
×Difficult to obtain gradient signals



Outline

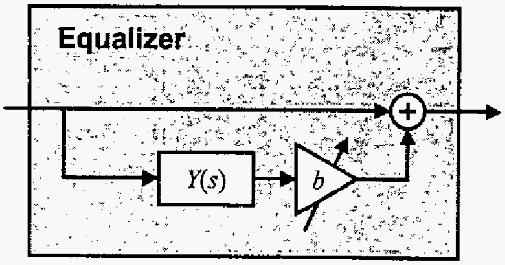
1. Motivation
2. Adaptive Filter Introduction
3. Adaptation algorithms
4. Filter structures & implementations
5. **Applications**
    a) **Co-axial cable**
    b) **Magnetic storage**
    c) **High-speed ethernet**
    d) **Chip-to-chip communications**
    e) **Optical Fibre**

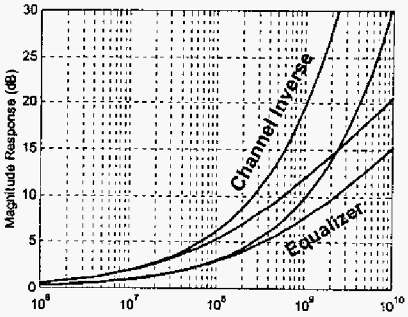
Co-axial Cable



- **Application**: high-speed point-to-point links
- Channel transfer function: $H(s)=e^{-aL\sqrt{s}}$
- ISI can be cancelled using inverse of channel response: $H^{-1}(s)=1/H(s)$
- Approximate, $H^{-1}(s)=e^{aL\sqrt{s}}\cong 1+b\cdot Y(s)$ where $Y(s)$ is fixed with +10 to +12 dB/decade slope

Co-axial Cable





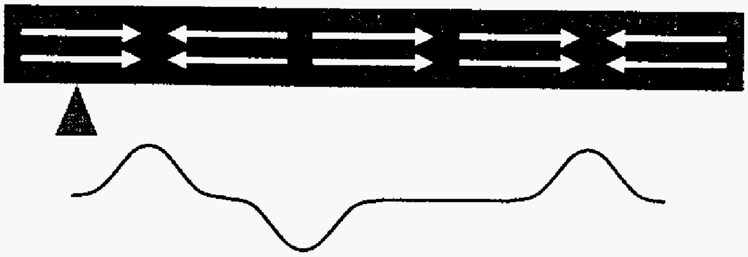
- **Equalizer examples**: [Hartman '99], [Shakiba '99]
- b is adapted via a LMS or other algorithm

⇒ An adaptive linear combiner

× Must be careful to avoid boosting noise too much

⇒ Practical implementations have a limited bandwidth

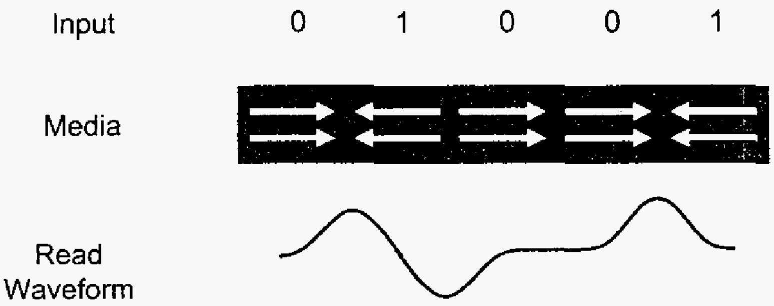
Magnetic Storage



Input	0	1	0	0	1
Media					
Read Waveform					
Output	+1	-1	0	+1	

➢ Partial Response Signal!

Magnetic Storage

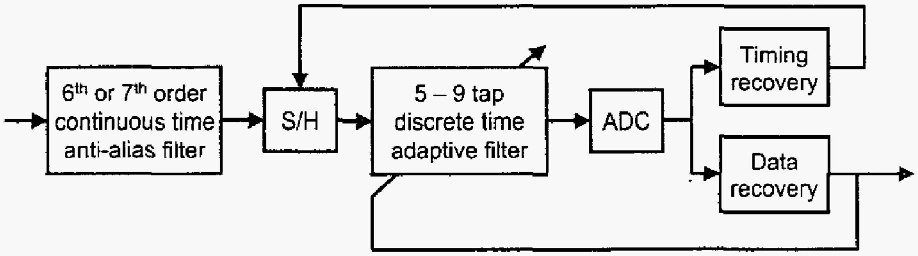


Increased Density

➢ Intersymbol Interference!

Magnetic Storage

- **Equalization #1**: Analog discrete time
- Examples: [Xu '96], [Kiriaki '97], [Wang '98]

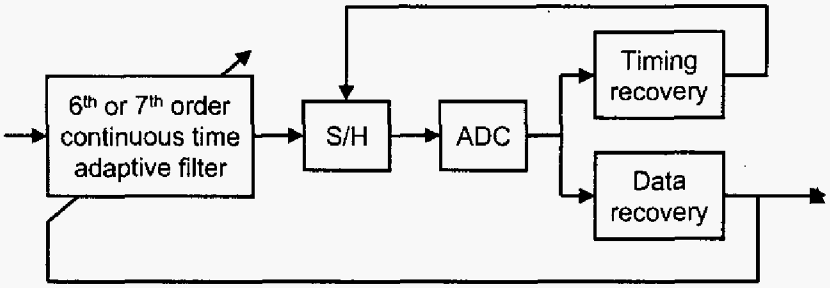


- Switched capacitor transversal filters with rotating tap weights or rotating switch matrix
- SS-LMS (digital) adaptation

✓ Robust and simple adaptation

Magnetic Storage

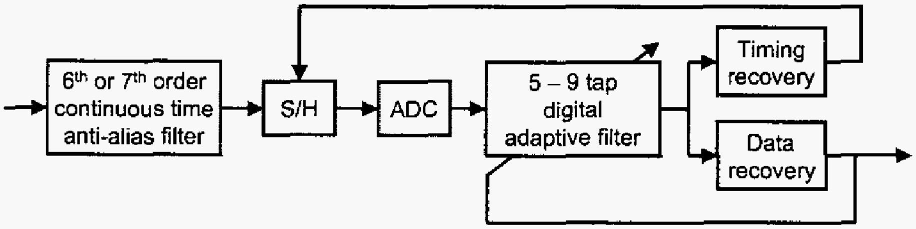
- **Equalization #2**: Analog continuous time
- Examples: [Pai '96], [Brown '99]



- Continuous time ladder structures with adapted zeros
- SS-LMS (digital) adaptation
- ✓ Generally lower power & smaller area at high speeds

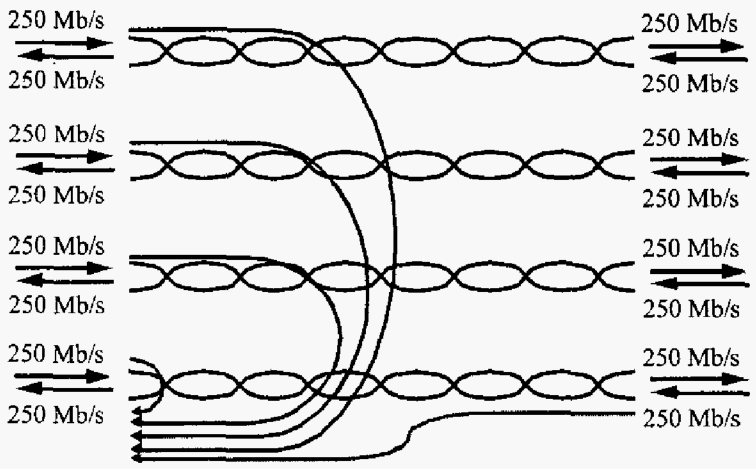
Magnetic Storage

- **Equalization #3**: Digital



- ✓ Robust and simple adaptation
- ✓ Scales well

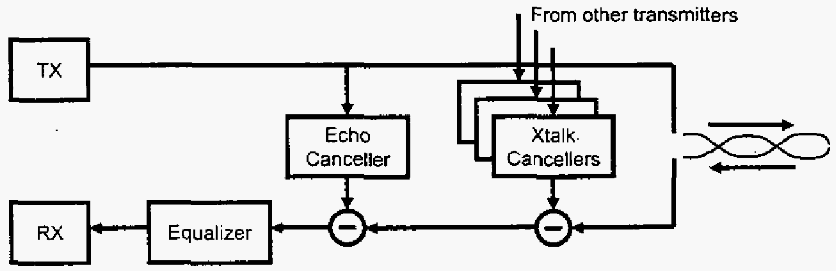
Gigabit Ethernet over Copper



Received Signal = Echo + Crosstalk + Desired

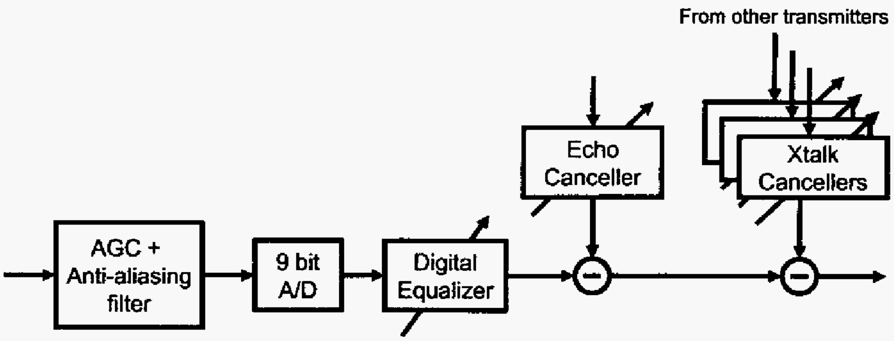
Gigabit Ethernet over Copper

- Challenges:
  - Elimination of ISI, echo, crosstalk
  - Integration of entire transceiver with reasonable power & cost



Analog vs. Digital Adaptive Filters

DSP based: [He '01], [Roo '01]

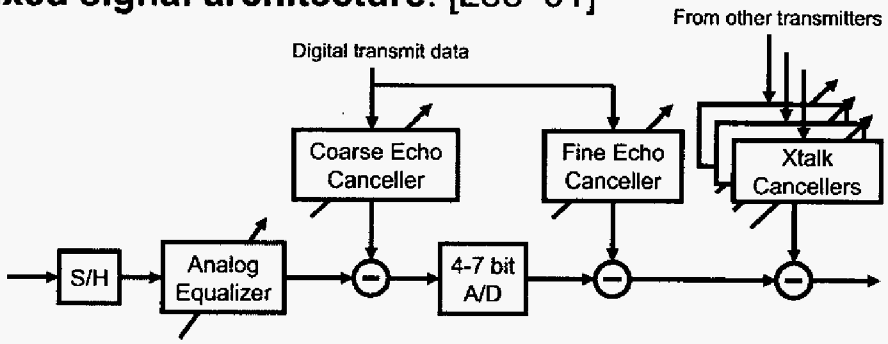


- Lots of high-speed DSP ⇒ power-hungry
- Considerable analog front end still required
- 9-bit A/D design is challenging

Analog Adaptive Filters
Prof. T. Chan Carusone, University of Toronto

Analog vs. Digital Adaptive Filters

Mixed signal architecture: [Lee '01]

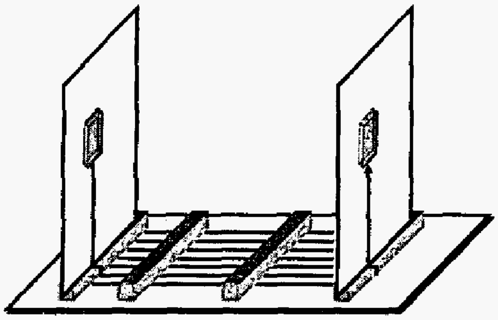


- S/H based transversal equalizer with rotating tap weights
- Digital SS-LMS adaptation using DACs for the tap weights
- Echo canceller accepts digital inputs from transmit path
- A "rover" tap in the echo canceller eliminates distant echoes

Analog Adaptive Filters
Prof. T. Chan Carusone, University of Toronto

Chip-to-chip communication

- Usually consists of sending information from one IC to another over PCB traces and (possibly) connectors



- ➢Attenuation (conductor losses)
- ➢Intersymbol interference (limited bandwidth)
- ➢Reflections/echoes (mismatches at connectors and termination)
- ➢Crosstalk (from neighbouring channels)

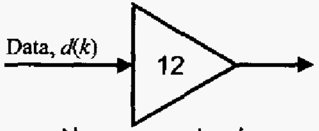
Chip-to-chip communication

- Most popular approach is adaptive pre-emphasis at transmitter

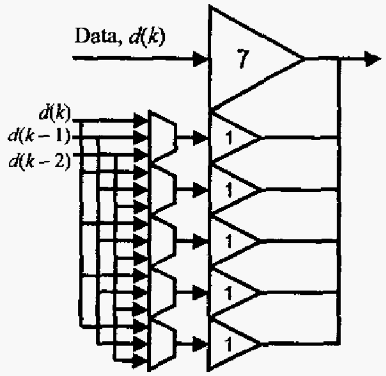
Example:

$a_0 + a_1 z^{-1} + a_2 z^{-2}$

- ➢ Minimal additional hardware



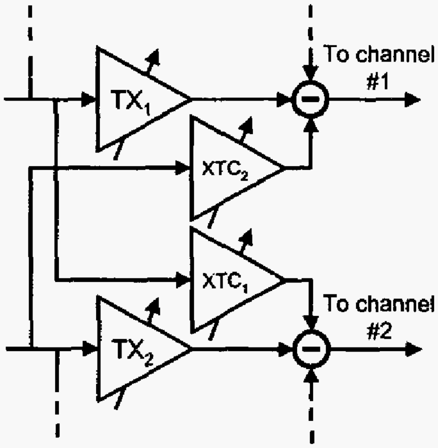
No pre-emphasis



With pre-emphasis

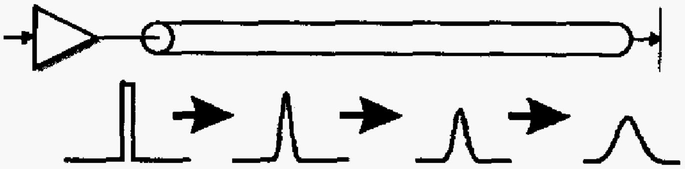
Chip-to-chip communication

- Crosstalk cancellation is similarly possible at the transmitter; e.g. [Zerbe '01]
- Challenge is performing adaptation, which requires information from the receiver [Stonick '03]
- Current research & development is directed at receive-side equalization



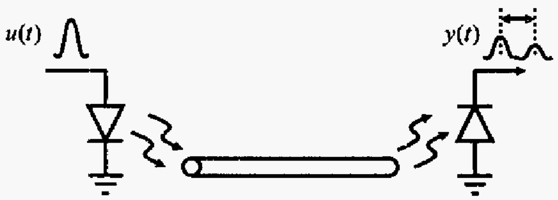
Optical Fiber

- Different colors (wavelengths) of light propagate along fiber at different velocities ⇒ "Chromatic dispersion"
- Causes pulses to spread out ⇒ Intersymbol Interference



Optical Fiber

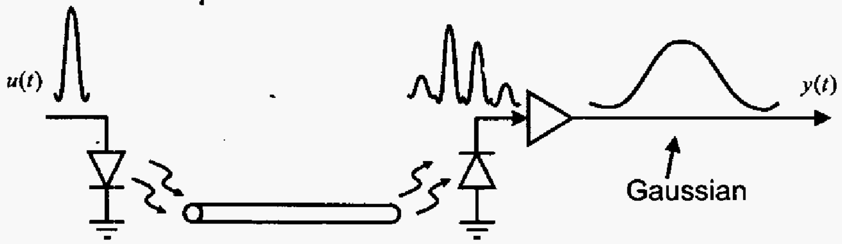
- Imperfections in the fiber cause light with different polarizations to propagate with different speeds: "Birefringence" or "Polarization mode dispersion"



DGD = Differential Group Delay

Optical Fiber

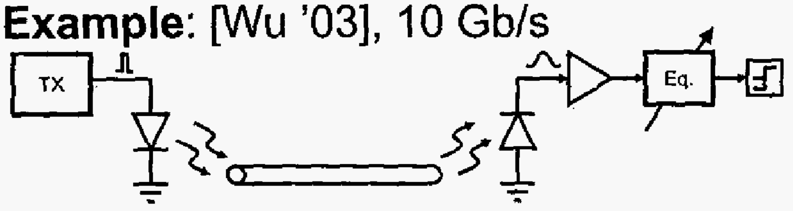
- Multimode fiber supports multiple modes of light propagation, each with a different velocity resulting in many received pulses of light with different amplitudes
- Bandwidth limitations of the receiver front-end smear together pulses into one Gaussian electrical pulse



Gaussian

Optical Fiber

- Can be cancelled using equalization (a.k.a. “Dispersion compensation”) [Azadet ’02]
- **Example**: [Wu ’03], 10 Gb/s



- Continuous time taped delay line
- Delay elements are passive LC networks

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