Adaptive Filters – Introduction

Gerhard Schmidt
Christian-Albrechts-Universität zu Kiel
Faculty of Engineering
Institute of Electrical and Information Engineering
Digital Signal Processing and System Theory
Contents of the Lecture

Today:

- Boundary conditions of the lecture
  - Contents
  - Literature hints
  - Exams
- Notation
- Example of an adaptive Filter
- Examples from speech and audio signal processing
Entire Semester:

- Introduction with examples for speech and audio processing
- Wiener Filter
- Linear Prediction
- Algorithms for adaptive filters
  - LMS und NLMS algorithm
  - Affine projection
  - RLS algorithm
- Control of adaptive filters
- Signal processing structures
- Applications of linear prediction
- Examples for speech and audio processing
**Statistical signal theory:**

**Adaptive filters:**

**Speech processing:**
Credit Points, Exams, Exercises, and Lecture Notes

Credit points:
- 4 ECTS points

Oral exam:
- About 30 minutes per student
- In the exams period

Exercises:
- Two Matlab exercises during the semester

Talks:
- Duration about 10 minutes (afterwards short discussion)
- Topics will be offered during the lectures (own suggestions are welcome)

Lecture notes:
- Printed versions will be spread at the beginning of each lecture
- In the internet via www.dss.tf.uni-kiel.de
Scalars and Vectors

**Scalars:**
- Signals: \( x(n) \)
- Impulse responses (time-variant): \( h_i(n) \)
- Example for a (real) convolution:
  \[
  y(n) = \sum_{i=0}^{N-1} x(n - i) h_i(n)
  \]

**Vectors:**
- Signal vectors:
  \[
  x(n) = \begin{bmatrix} x(n), x(n - 1), \ldots, x(n - N + 1) \end{bmatrix}^T
  \]
- Impulse response vectors (time-variant):
  \[
  h(n) = \begin{bmatrix} h_0(n), h_1(n), \ldots, h_{N-1}(n) \end{bmatrix}^T
  \]
- Example for a real convolution:
  \[
  y(n) = x^T(n) h(n) = h^T(n) x(n)
  \]

**Matrices:**
- \( A(n) = \begin{bmatrix} a_{00}(n) & a_{01}(n) & \ldots & a_{0N}(n) \\ a_{10}(n) & a_{11}(n) & \ldots & a_{1N}(n) \\ \vdots & \vdots & \ddots & \vdots \\ a_{M0}(n) & a_{M1}(n) & \ldots & a_{MN}(n) \end{bmatrix} \)
Random Variables and Processes:

- **Notation:** \( x(n), x_1(n), x_2(n) \)

- **Probability density function:** \( f_x(x, n), f_{x_{1\times2}}(x_1, x_2, n_1, n_2) \)

- **Stationary random processes:**
  \[
  f_x(x, n) = f_x(x, n + n_0) = f_x(x) \\
  f_{x_{1\times2}}(x_1, x_2, n_1, n_2) = f_{x_{1\times2}}(x_2, x_2, n_1 + n_0, n_2 + n_0) = f_{x_{1\times2}}(x_1, x_2, n_2 - n_1)
  \]

- **Expected values of stationary random processes:**
  \[
  \mathbb{E}\left\{x(n)\right\} = \int_{x=-\infty}^{\infty} x f_x(x) \, dx = \mu_x^{(1)} = \mu_x \\
  \mathbb{E}\left\{x^2(n)\right\} = \int_{x=-\infty}^{\infty} x^2 f_x(x) \, dx = \mu_x^{(2)}, \\
  \mathbb{E}\left\{g(x(n))\right\} = \int_{x=-\infty}^{\infty} g(x) f_x(x) \, dx
  \]
Auto and cross correlation for real, stationary random processes:

- **Auto-correlation function:**
  \[
  E \left\{ x(n) x(n + l) \right\} = s_{xx}(l)
  \]

- **Cross-correlation function:**
  \[
  E \left\{ x(n) y(n + l) \right\} = s_{xy}(l)
  \]

- **(Auto) power spectral density:**
  \[
  S_{xx}(\Omega) = \sum_{l=-\infty}^{\infty} E \left\{ x(n) x(n + l) \right\} e^{-j\Omega l} = \sum_{l=-\infty}^{\infty} s_{xx}(l) e^{-j\Omega l}
  \]

- **(Cross) power spectral density:**
  \[
  S_{xy}(\Omega) = \sum_{l=-\infty}^{\infty} E \left\{ x(n) y(n + l) \right\} e^{-j\Omega l} = \sum_{l=-\infty}^{\infty} s_{xy}(l) e^{-j\Omega l}
  \]
White Noise

**Stationary white noise:**

- Auto-correlation function:
  \[
  s_{xx}(l) \bigg|_{\text{white noise}} = \begin{cases} 
  \sigma_x^2, & \text{if } l = 0, \\
  0, & \text{else}.
\end{cases}
  \]

- Auto power spectral density:
  \[
  S_{xx}(\Omega) \bigg|_{\text{white noise}} = \sigma_x^2
  \]
Basic Structure

A First Example of an Adaptive Filter – Part 1

Digital Signal Processing and System Theory | Adaptive Filters | Introduction

Slide I-10
Matlab Demo
Selected Application Areas

- Speech coding (e.g. GSM, UMTS)
- Speech enhancement (hands-free systems, hearing aids, public address systems)
- Equalization (sending antennas, radar, loudspeakers)
- Anti-noise systems (cars and airplanes)
- Multi-channel signal processing (beamforming, submarine localization, layer of earth analysis)
- Missile control
- Medical applications (fetal heart rate monitoring, dialysis)
- Processing of video signals (cancellation of distortions, image analysis)
- Antenna arrays
Basis Structures of Adaptive Filters – Part 1

System Identification

$\text{Examples:}$

- Line echo cancellation
- Cancellation of acoustical echoes
Inverse Modelling

Examples:
- Equalization of amplifiers of transmission antennas
- Loudspeaker equalization
Examples:

- Speech coding in the GSM and UMTS networks
- Suppression of carrier signals after demodulation
Cancellation of Undesired Signals

\[
y(n) = F(x(n)) + b(n)
\]

**Example:**
- Automotive speech signal enhancement via cancellation of engine harmonics
Examples from Speech and Audio Processing

Contents

Part 1: Automotive hands-free telephone systems
- Basics
- Solutions
- Examples

Part 2: In-car communication systems
- Basics
- Solutions
- Examples
Automotive Hands-Free Telephone Systems
Automotive Hands-Free Telephone Systems

Basics – Electro-Acoustic Transducers

**Microphones:**
- Integrated in the rear-view mirror (example)
- Up to four microphones

**Loudspeakers:**
- Loudspeakers of the car stereo (head unit)
- coupling > 0 dB
- Volume adjustable by the passengers
Automotive Hands-Free Telephone Systems

Basics – Loudspeaker Enclosure Microphone (LEM) Systems – Part 1

Signal of the remote communication partner

\[ x(n) \]: Excitation signal

\[ d(n) \]: Echo (*desired*) signal

\[ s(n) \]: Local speech signal

\[ b(n) \]: Background noise

\[ y(n) \]: Microphone signal
Assumption:
The loudspeaker enclosure microphone system (LEM system) can be modeled as a linear system with finite memory.
**Boundary conditions:**
- Volume of a passenger compartment: $5 \ldots 15 \text{ m}^3$

**Properties:**
- Short delay
- Direct sound after $3 \ldots 4 \text{ ms}$
- Early reflections
- Diffuse sound (decays logarithmically in amplitude)
Automotive Hands-Free Telephone Systems

Basics – Background Noise and its Components

**External components:**
- Engine noise
- Wind noise
- Tire noise

**Internal components:**
- Air conditioning
- Defrost

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![Power spectral densities of background noise measured in a car at different speeds](image)

- 200 km/h
- 150 km/h
- 100 km/h
- 50 km/h
- 0 km/h (engine on)
- 0 km/h (engine off)
Automotive Hands-Free Telephone Systems

A Basic System With Two Adaptive Filters

\[ x(n) \rightarrow \hat{h}(n) \rightarrow y(n) \]

\[ x(n) \rightarrow d(n) \rightarrow s(n) \rightarrow b(n) \]

\[ \hat{s}(n) \rightarrow w(n) \rightarrow e(n) \]

- Echo cancellation filter
- Noise suppression filter
- Adaptive filters for echo and noise suppression
An Adaptive Filter for Cancellation of Acoustical Echoes

- Adaptive echo cancellation filter
- FIR model

(system parameters are unknown, only input and output signals are measurable)
Maximal Achievable Echo Reduction – Part 1

Derivation during the lecture ...
Maximal Achievable Echo Reduction – Part 2

**Boundary conditions:**

- White noise as excitation signal
- Ideal convergence, meaning that all filter coefficients of the adaptive filter are equal to the corresponding ones of the impulse response.
- Linear loudspeakers, microphones, and amplifiers

![Graph showing maximum echo attenuation in relation to the filter length](image)
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A Basic System With Two Adaptive Filters

- Noise suppression filter
- Echo cancellation filter

$x(n)$ → $\hat{h}(n)$ → $\hat{d}(n)$ → $\tilde{s}(n)$

$x(n)$ → $\hat{h}(n)$ → $\hat{d}(n)$ → $y(n)$

$s(n)$ → $d(n)$ → $b(n)$
Residual Echo and Noise Suppression

Remaining echoes....

\[(d(n) - \hat{d}(n)) + b(n)\]

... and local background noise

Local speech signal \(s(n)\)

\[e(n)\]

\[w(n)\]

\[\hat{s}(n)\]

**Approach according to Wiener (next lecture):**

Cross power spectral density of the distorted input signal \(e(n)\) and the desired output signal \(s(n)\)

\[E \left\{ \left( s(n) - \hat{s}(n) \right)^2 \right\} \rightarrow \text{min} \implies W_{\text{opt}} (e^{j\Omega}) = \frac{S_{es}(\Omega)}{S_{ee}(\Omega)}\]

Auto power spectral density of the distorted input signal \(e(n)\)
Automotive Hands-Free Telephone Systems

A Basic System With Two Adaptive Filters – Audio Examples

**Stereo signals (16 kHz):**

<table>
<thead>
<tr>
<th>Left:</th>
<th>Right:</th>
</tr>
</thead>
<tbody>
<tr>
<td>Received signal ...</td>
<td>Sent signal ...</td>
</tr>
<tr>
<td>... of the remote communication partner</td>
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Transmission to the communication partner (channel delay: about 180 ms)

- **Initial filter convergence:** Adaptation at the beginning of the call
- **Double talk:** Both partners speak simultaneously
- **Enclosure dislocations:**
  - Without Wiener filter
  - With Wiener filter
Enhanced Systems

**Improvements:**

- Improved noise suppression by adaptive combination of several microphone signals (beamforming)
- Further improvements by applying adaptive filters for different kinds of distortions
Automotive Hands-Free Telephone Systems

Microphone Array Using Four Sensors (Integrated into the Rear-View Mirror)

- Cheap realization by means of an integrated microphone module.
- A fixed steering direction can be used for the driver – the steering angle varies only in a small range (62° - 75°).
- The array can be used for the driver and for the passenger simultaneously.
- Cardioid microphones are usually applied (± 3 dB sensitivity).
Beamforming – Introduction

**Beamformer:**

- Minimizing the output power with respect to one or more constraints (signals from a desired direction must pass the structure without distortion)
- The desired direction is known in automotive applications (at least approximately)
- The performance of adaptive filtering is limited by sensor tolerances and multipath propagation within the passenger compartment
Beamforming – Adaptive Structure

\[ y_0(n) \rightarrow \text{Delay} \rightarrow \text{Summation path} \rightarrow \frac{1}{2} \rightarrow \text{Output of the so-called generalized sidelobe canceller} \]

\[ y_1(n) \rightarrow \text{Blocking path} \rightarrow \text{Adaptive filter} \rightarrow e(n) \]

"Griffith-Jim" beamformer (generalized sidelobe canceller)
Beamforming – Audio Examples

- 4-channel beamformer
- Loudspeaker on the passengers seat (undesired signal)
- Adaptive filtering of the microphone signal results in an SNR improvement of about 15 dB.

Single microphone
Fixed beamformer
Adaptive beamformer
Speech and noise were mixed artificially to obtain different signal-to-noise ratios.

About 30 command words for controlling the radio and phone system were used.

16 subjects (9 male, 7 female) participated in the test.
Automotive Hands-Free Telephone Systems

Involved Signal Processing Units – Start

- Microphone array
- Acoustic coupling from the loudspeaker to the microphone(s)
- Telephone or speech dialog system
Automotive Hands-Free Telephone Systems

Involved Signal Processing Units – Bandwidth Extension

**Bandwidth extension**

Missing frequency components were estimated and resynthesized.

Effect: The speech quality (not the intelligibility) of the received signal is improved.

Microphone array

Acoustic coupling from the loudspeaker to the microphone(s)

Telephone or speech dialog system

Bandwidth extension
Involved Signal Processing Units – Automatic Gain and Equalization Adjustment

**Volume and equalization control**

The (broadband) playback volume is adjusted automatically with respect to the noise measured in the car. In addition also the spectrum can be shaped in order to improve the perceived signal quality.
Involved Signal Processing Units – Adaptive Limiter

**Adaptive limiter**

Adaptive adjustment of the parameters of a limiter in order to avoid microphone clipping by those loudspeakers that are close to the microphones (e.g. so-called center speaker).
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Involved Signal Processing Units – Echo Cancellation

*Echo cancellation*

The signals emitted by the loudspeakers are reflected by windows, etc. These reflected signals as well as directly coupled signals are also recorded by the microphones.

To decouple the electro-acoustic system, the echo signals are estimated and subtracted from the microphone signal.
Beamforming

The microphone signals are filtered such that a predefined direction is kept open, while other directions are attenuated as much as possible.

Effect: Directional distortions can be suppressed.
Involved Signal Processing Units – Noise and Residual Echo Suppression

Background noise and residual echo suppression

Despite beamforming and echo cancellation several remaining undesired signal components are still audible.

Effect: Stationary background noise and residual echoes can be suppressed.

microphone array

Acoustic coupling from the loudspeaker to the microphone(s)

Beam-forming

Noise and echo suppression

Echo cancellation

Adaptive limiter

Adaptive volume and equalization control

Bandwidth extension

Telephone or speech dialog system
Involved Signal Processing Units – Wind Buffet Removal

Wind buffet suppression

Open windows and defrost on might cause wind buffets.

Effect: A detection optimized for those undesired signals finds wind buffets and replaces the signal with so-called comfort noise.
Involved Signal Processing Units – Removal of “Transients”

Suppression of transients

Transient signal, such as the noise of an indicator or a wind shield wiper, cause problems for voice recognitions signals (voice activity detection).

Effect: Short impulsive distortions are suppressed.
Involved Signal Processing Units – Adaptive Equalization

Adaptive equalization

For compensation of different microphone-speaker distances and room characteristics, a (blind) equalization can be performed adaptively.

Effect: The signal sounds more natural.

- Beam-forming
- Noise and echo suppression
- Wind buffet removal
- Suppression of transients
- Adaptive equalization
- Echo cancellation
- Telephone or speech dialog system
- Adaptive limiter
- Adaptive volume and equalization control
- Bandwidth extension

Acoustic coupling from the loudspeaker to the microphone(s)
## Involved Signal Processing Units – Summary

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In-Car Communication Systems
In-Car Communication Systems

Motivation

**Current situation:**
- Communication between passengers is difficult, because of the acoustic loss (especially front to rear).
- Driver turns around – road safety is reduced.
- Front passengers have to speak louder than normal – longer conversations will be tiring.

**Solutions:**
- Improve the speech quality and intelligibility by means of an intercom system.

**Application:**
- Mid and high-class automobiles, which are already equipped with the necessary audio and signal processing devices.
- Vans, etc. – systems with reduced complexity.

*Acoustic loss (referred to the ear of the driver)*
Solution:
- Improve the speech quality and intelligibility by means of an ICC system.
- The ICC system records the speech by means of microphones and improves the communication by playing back the signals via those loudspeakers that are close to the listening passengers.
In-Car Communication Systems

Results of a Comparison Mean Opinion Score (CMOS) Test

0 km/h, car parked close to a motorway

- 19.7% prefer the system to be switched off
- 29.7% have no preference
- 50.6% prefer an activated system

130 km/h, on a motorway

- 4.3% prefer the system to be switched off
- 7.1% have no preference
- 88.6% prefer an activated system

With permission from Eberhard Hänsler, Gerhard Schmidt (eds.), *Topics in Acoustic Echo and Noise Control*, Springer, 2006
On a parking area beside motorway (0 km/h):

- No significant difference (95.2% system off versus 95.0% system on).
- Due to the automatic gain adjustment the intercom system operates with only very small gain at these noise levels.

On a motorway (130 km/h):

- Significant improvement of the DRT error rate.
- Nearly 50% error reduction (85.4% correct answers increased to 92.2% correct answers).

With permission from Eberhard Hänsler, Gerhard Schmidt (eds.), *Topics in Acoustic Echo and Noise Control*, Springer, 2006
Adaptive Filters – Introduction

Summary and Outlook

This week:

- Boundary conditions of the lecture
  - Contents
  - Literature hints
  - Exams
- Notation
- Example of an adaptive Filter
- Examples from speech and audio signal processing

Next week:

- Wiener filter
- Noise suppression