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

Contents of the Lecture

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

Literature

English and German Books:

Statistical signal theory:

- A. Papoulis: ***Probability, Random Variables, and Stochastic Processes***, McGraw Hill, 1965
- E. Hänsler: ***Statistische Signale: Grundlagen und Anwendungen***, Springer, 2001 (in German)

Adaptive filters:

- E. Hänsler, G. Schmidt: ***Acoustic Echo and Noise Control***, Wiley, 2004
- S. Haykin: ***Adaptive Filter Theory, Prentice Hall***, 2002
- A. Sayed: ***Fundamentals of Adaptive Filtering***, Wiley, 2004

Speech processing:

- L. R. Rabiner, R. W. Schafer: ***Digital Processing of Speech Signals***, Prentice Hall, 1978
- P. Vary, R. Martin: ***Digital Speech Transmission***, Wiley, 2006
- L. R. Rabiner, R. W. Schafer: ***Introduction to Digital Speech Processing***, Now, 2008

Boundary Contition of the Lecture

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:

Discrete time index

- Signals: $x(n)$

Coefficient index

- 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:

Boldface and lowercase

- Signal vectors: $\boldsymbol{x}(n) = \big[x(n),\, x(n-1),\, \ldots,\, x(n-N+1)\big]^{\mathrm{T}}$
- Impulse response vectors (time-variant) : $\boldsymbol{h}(n) = \big[h_0(n),\, h_1(n),\, \ldots,\, h_{N-1}(n)\big]^{\mathrm{T}}$
- Example for a real convolution: $y(n) = \boldsymbol{x}^{\mathrm{T}}(n)\, \boldsymbol{h}(n) = \boldsymbol{h}^{\mathrm{T}}(n)\, \boldsymbol{x}(n)$

Matrices:

$$\boldsymbol{A}(n) = \begin{bmatrix} a_{00}(n) & a_{01}(n) & \ldots & a_{0N}(n) \\ a_{10}(n) & a_{11}(n) & \ldots & a_{1N}(n) \\ \vdots & \vdots & & \vdots \\ a_{M0}(n) & a_{M1}(n) & \ldots & a_{MN}(n) \end{bmatrix}$$

Boldface and uppercase

Random Processes

Random variables and processes:

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

  No differences between deterministic signals and random processes – different writing styles: $\boldsymbol{x}(\eta, n),\ x(\omega, n),\ X(n)$

- Probability density function: $f_x(x,n),\ f_{x_1 x_2}(x_1, x_2, n_1, n_2)$

- Stationary random processes:

$$
\begin{aligned}
f_x(x,\, n) &= f_x(x,\, n+n_0) \;=\; f_x(x) \\
f_{x_1 x_2}(x_1,\, x_2,\, n_1,\, n_2) &= f_{x_1 x_2}(x_2,\, x_2,\, n_1+n_0,\, n_2+n_0) \\
&= f_{x_1 x_2}(x_1,\, x_2,\, n_2-n_1)
\end{aligned}
$$

- Expected values of stationary random processes:

$$
\mathrm{E}\Big\{x(n)\Big\} \;=\; \int\limits_{x=-\infty}^{\infty} x\, f_x(x)\, dx \;=\; \mu_x^{(1)} \;=\; \mu_x
$$

$$
\mathrm{E}\Big\{x^2(n)\Big\} \;=\; \int\limits_{x=-\infty}^{\infty} x^2\, f_x(x)\, dx \;=\; \mu_x^{(2)}, \qquad \mathrm{E}\Big\{g\big(x(n)\big)\Big\} \;=\; \int\limits_{x=-\infty}^{\infty} g(x)\, f_x(x)\, dx
$$

Correlation

Auto and cross correlation for real, stationary random processes:

- Auto-correlation function:

$$\mathrm{E}\Big\{x(n)\,x(n+l)\Big\} \;=\; s_{xx}(l)$$

- Cross-correlation function:

$$\mathrm{E}\Big\{x(n)\,y(n+l)\Big\} \;=\; s_{xy}(l)$$

- (Auto) power spectral density:

$$S_{xx}(\Omega) \;=\; \sum_{l=-\infty}^{\infty} \mathrm{E}\Big\{x(n)\,x(n+l)\Big\}\,e^{-j\Omega\,l} \;=\; \sum_{l=-\infty}^{\infty} s_{xx}(l)\,e^{-j\Omega\,l}$$

- (Cross) power spectral density:

$$S_{xy}(\Omega) \;=\; \sum_{l=-\infty}^{\infty} \mathrm{E}\Big\{x(n)\,y(n+l)\Big\}\,e^{-j\Omega\,l} \;=\; \sum_{l=-\infty}^{\infty} s_{xy}(l)\,e^{-j\Omega\,l}$$

Notation – Part 4

White Noise

Stationary white noise:

- Auto-correlation function:

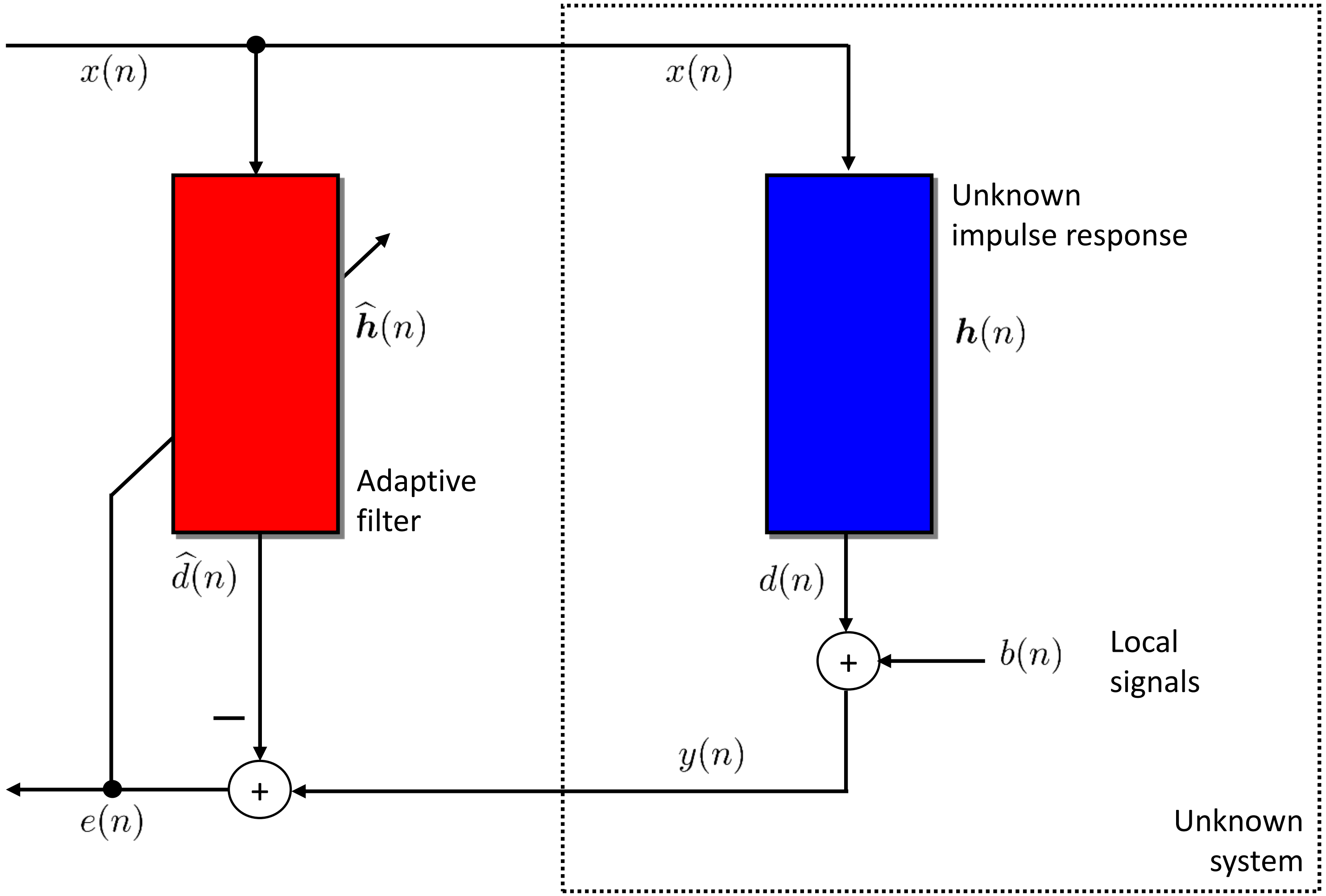
$$s_{xx}(l)\Big|_{\text{white noise}} = \begin{cases} \sigma_x^2, & \text{if } l = 0, \\ 0, & \text{else.} \end{cases}$$

- Auto power spectral density:

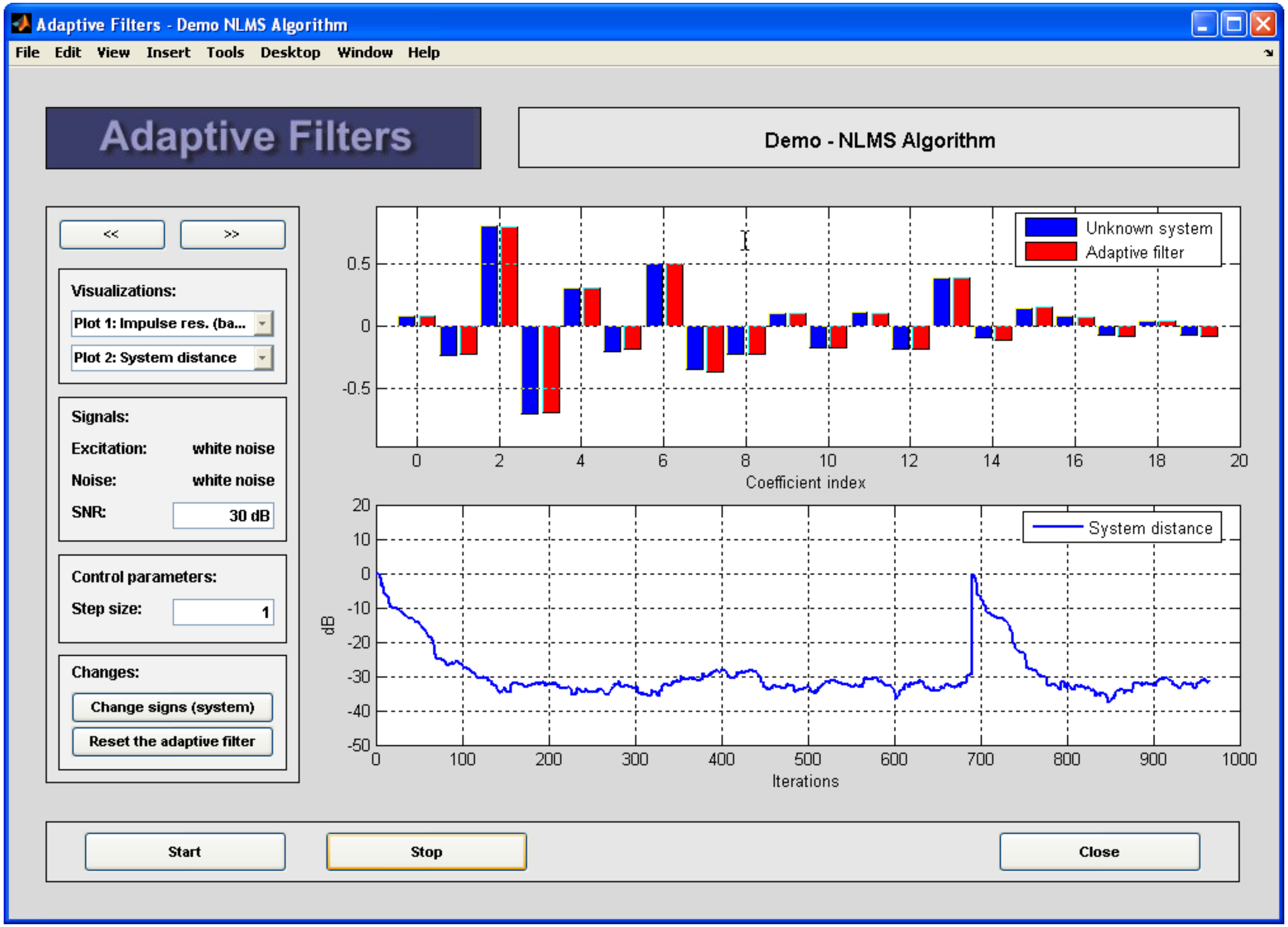
$$S_{xx}(\Omega)\Big|_{\text{white noise}} = \sigma_x^2$$

A First Example of an Adaptive Filter – Part 1

Basic Structure



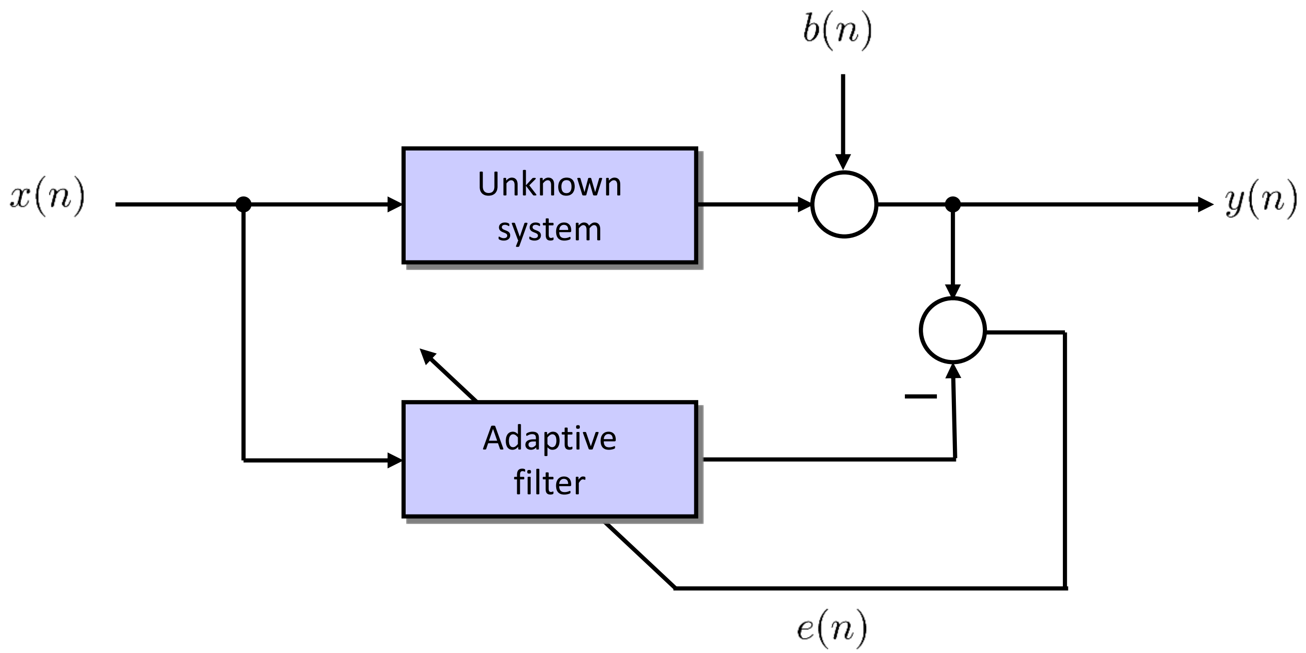
Matlab Demo


Adaptive Filters - Demo NLMS Algorithm
File Edit View Insert Tools Desktop Window Help
Adaptive Filters
Demo - NLMS Algorithm
<<
>>
Visualizations:
Plot 1: Impulse res. (ba...
Plot 2: System distance
Signals:
Excitation: white noise
Noise: white noise
SNR: 30 dB
Control parameters:
Step size: 1
Changes:
Change signs (system)
Reset the adaptive filter
Unknown system
Adaptive filter
Coefficient index
System distance
dB
Iterations
Start
Stop
Close

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

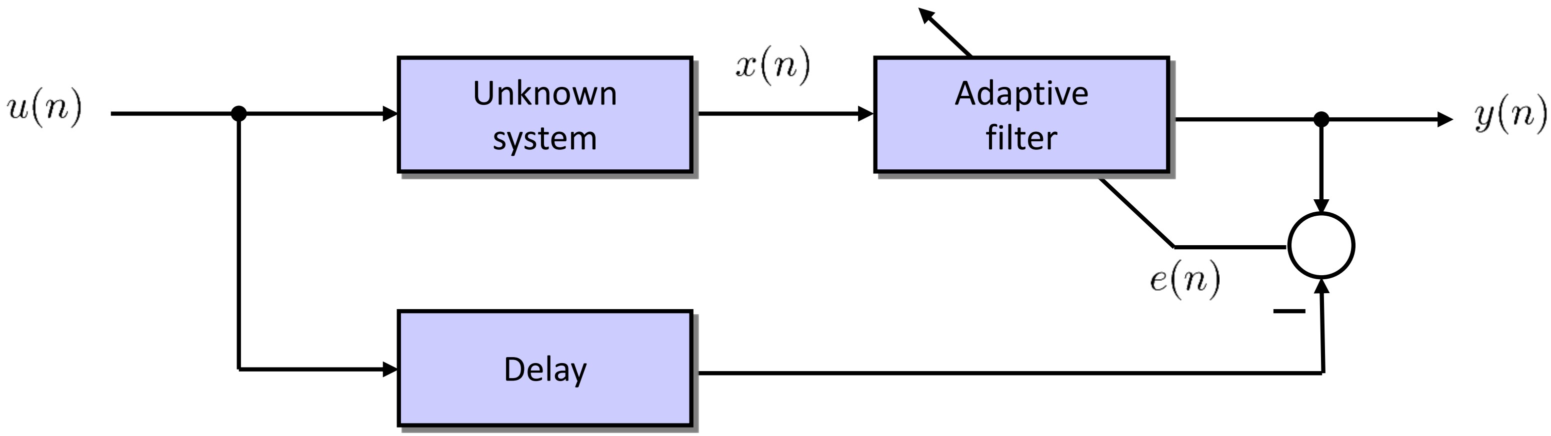
System Identification



***Examples*:**

- ❑ Line echo cancellation
- ❑ Cancellation of acoustical echoes

Inverse Modelling

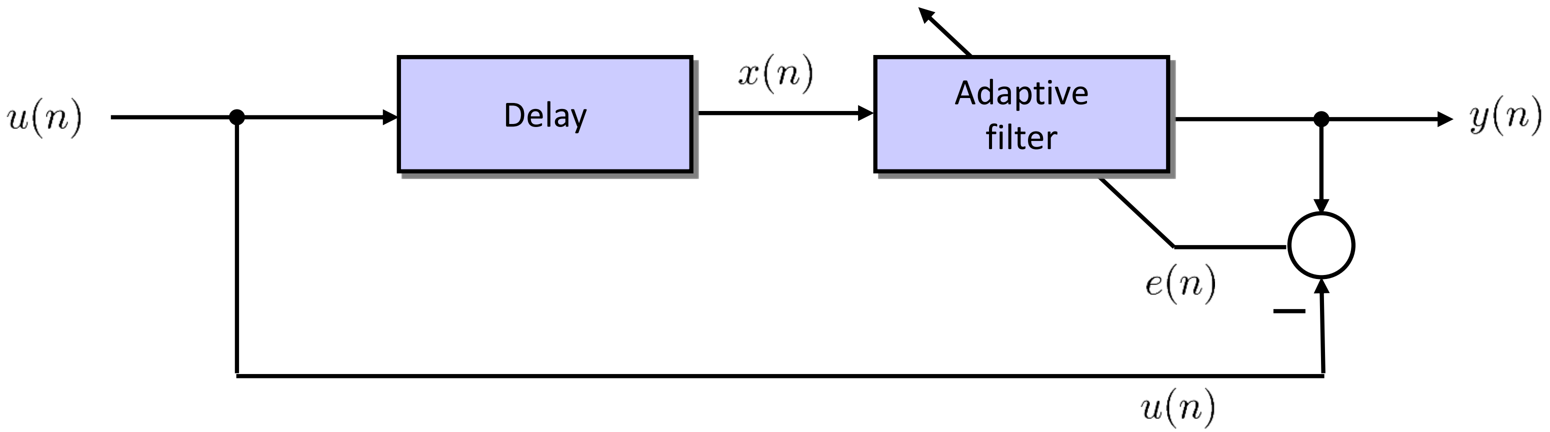


Distortions are not depicted!

Examples:

- Equalization of amplifiers of transmission antennas
- Loudspeaker equalization

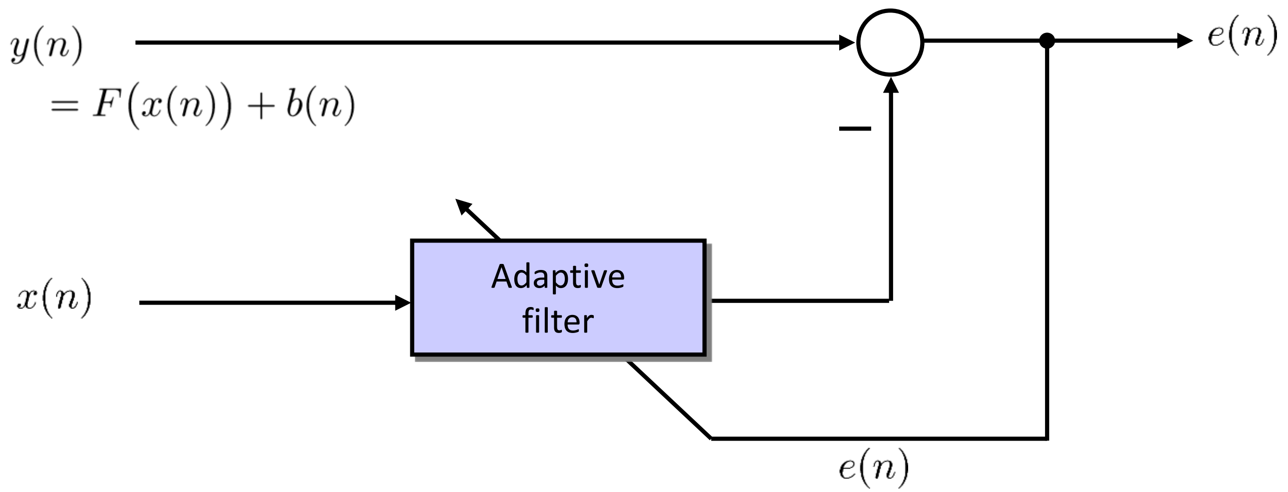
Prediction



***Examples*:**

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

Cancellation of Undesired Signals



***Example*:**

- Automotive speech signal enhancement via cancellation of engine harmonics

Contents

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

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

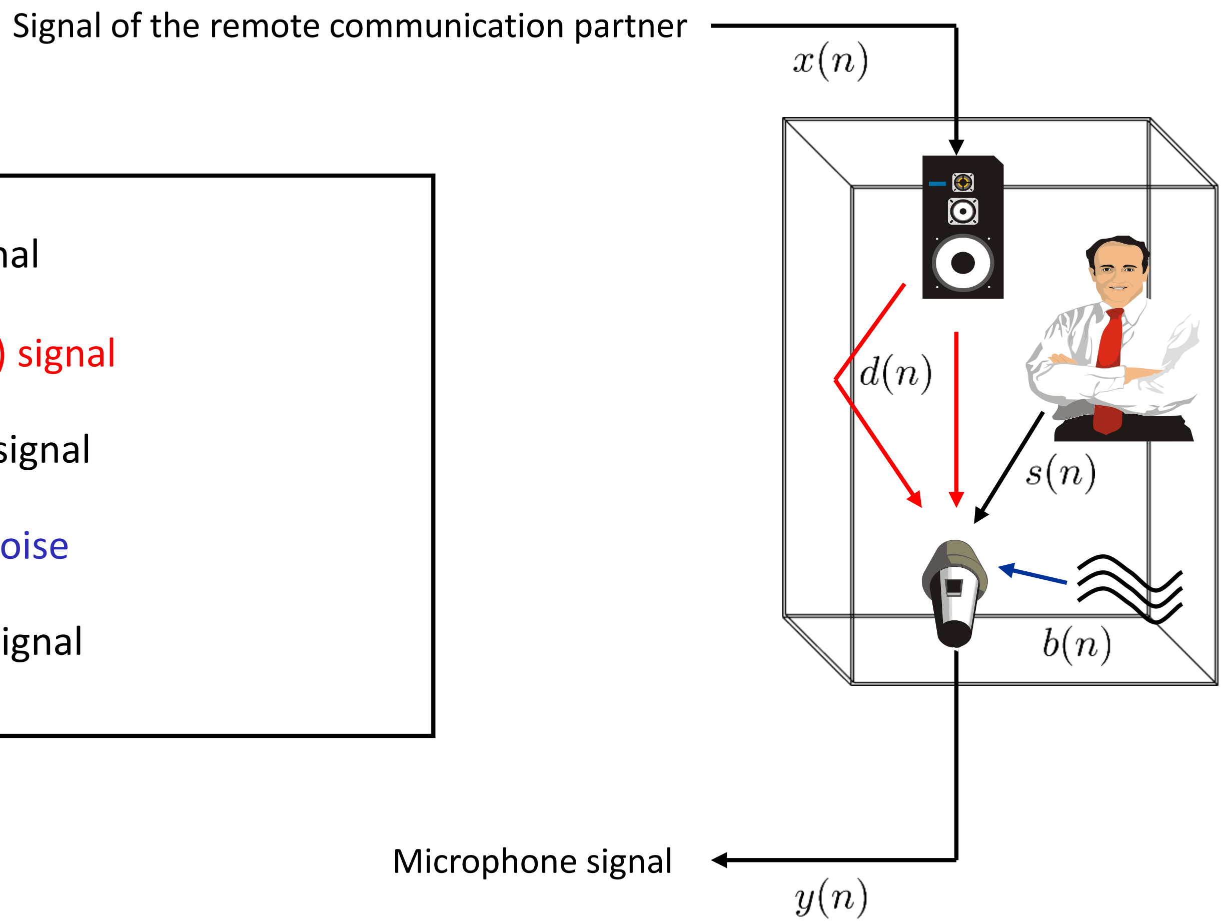
$x(n)$: Excitation signal

$d(n)$: Echo (*desired*) signal

$s(n)$: Local speech signal

$b(n)$: Background noise

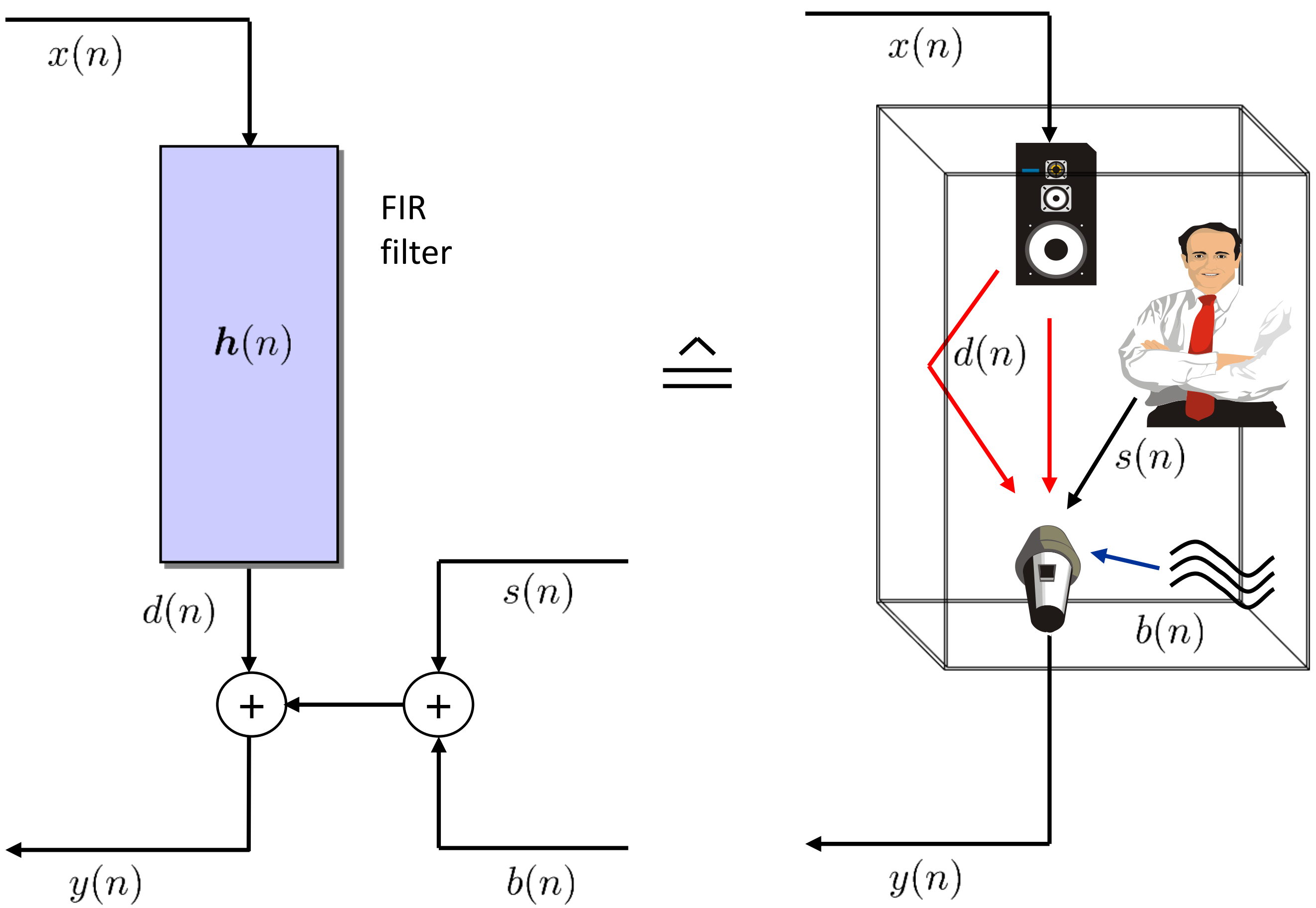
$y(n)$: Microphone signal



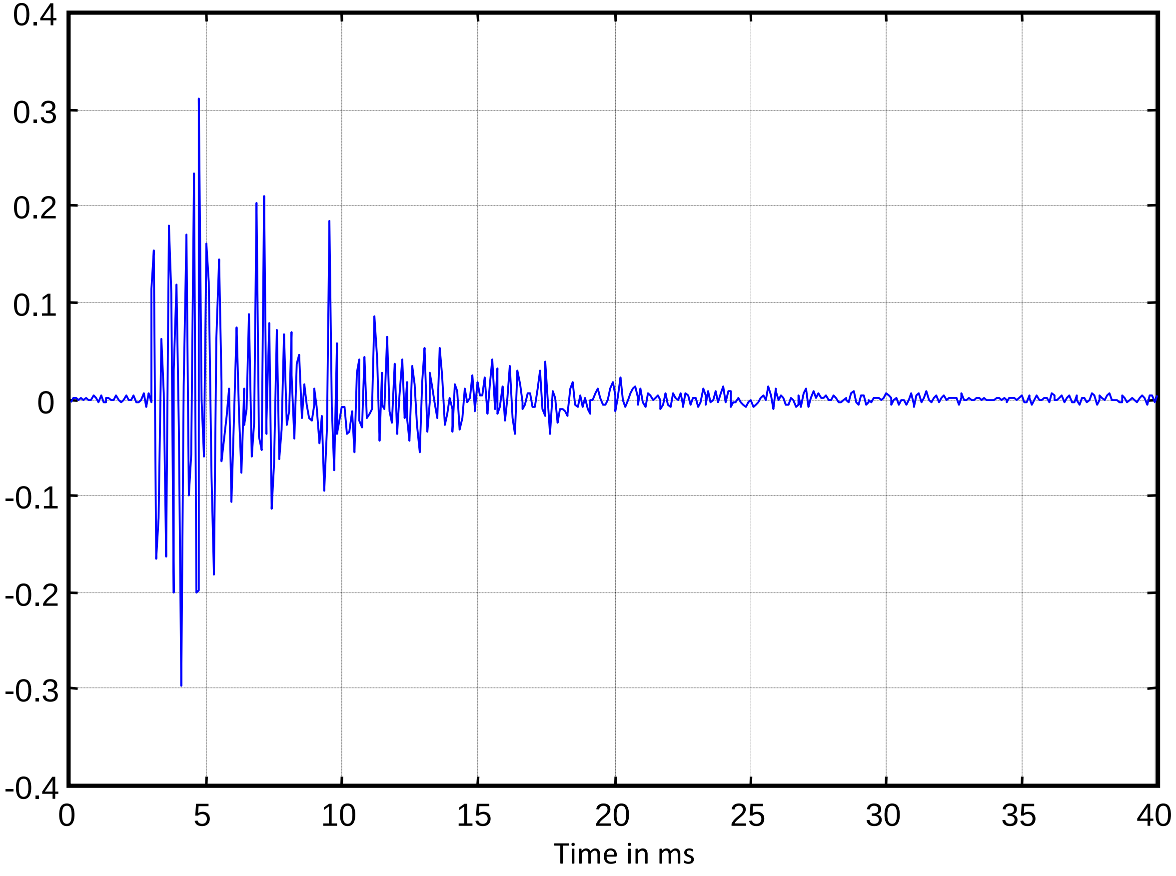
Basics – Loudspeaker Enclosure Microphone (LEM) Systems – Part 2

Assumption:

The loudspeaker enclosure microphone system (LEM system) can be modeled as a linear system with finite memory.



Basics – Loudspeaker Enclosure Microphone (LEM) Systems – Part 3



Boundary conditions:

- Volume of a passenger compartment: 5 … 15 m³

Properties:

- Short delay
- Direct sound after 3 … 4 ms
- Early reflections
- Diffuse sound (decays logarithmically in amplitude)

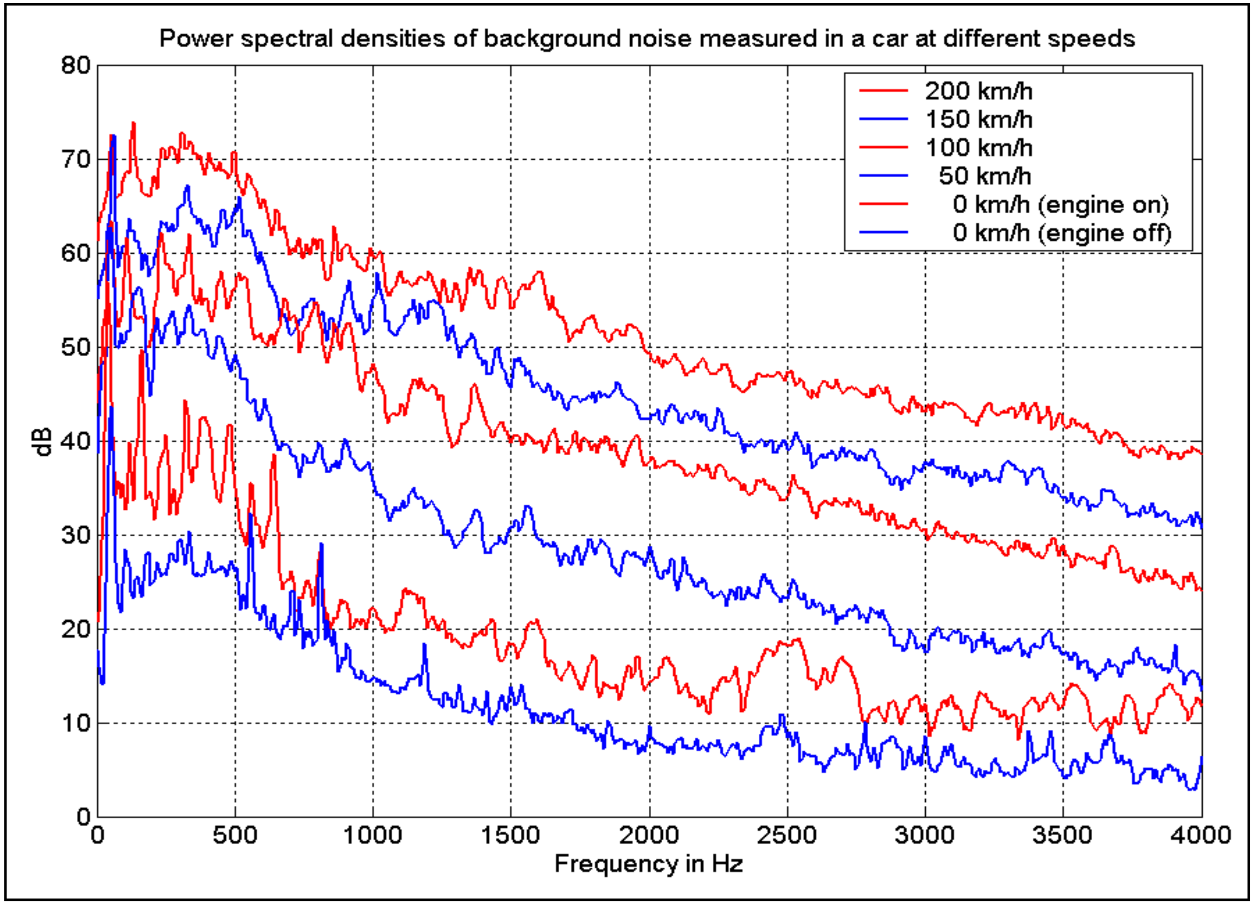
Basics – Background Noise and its Components

External components:

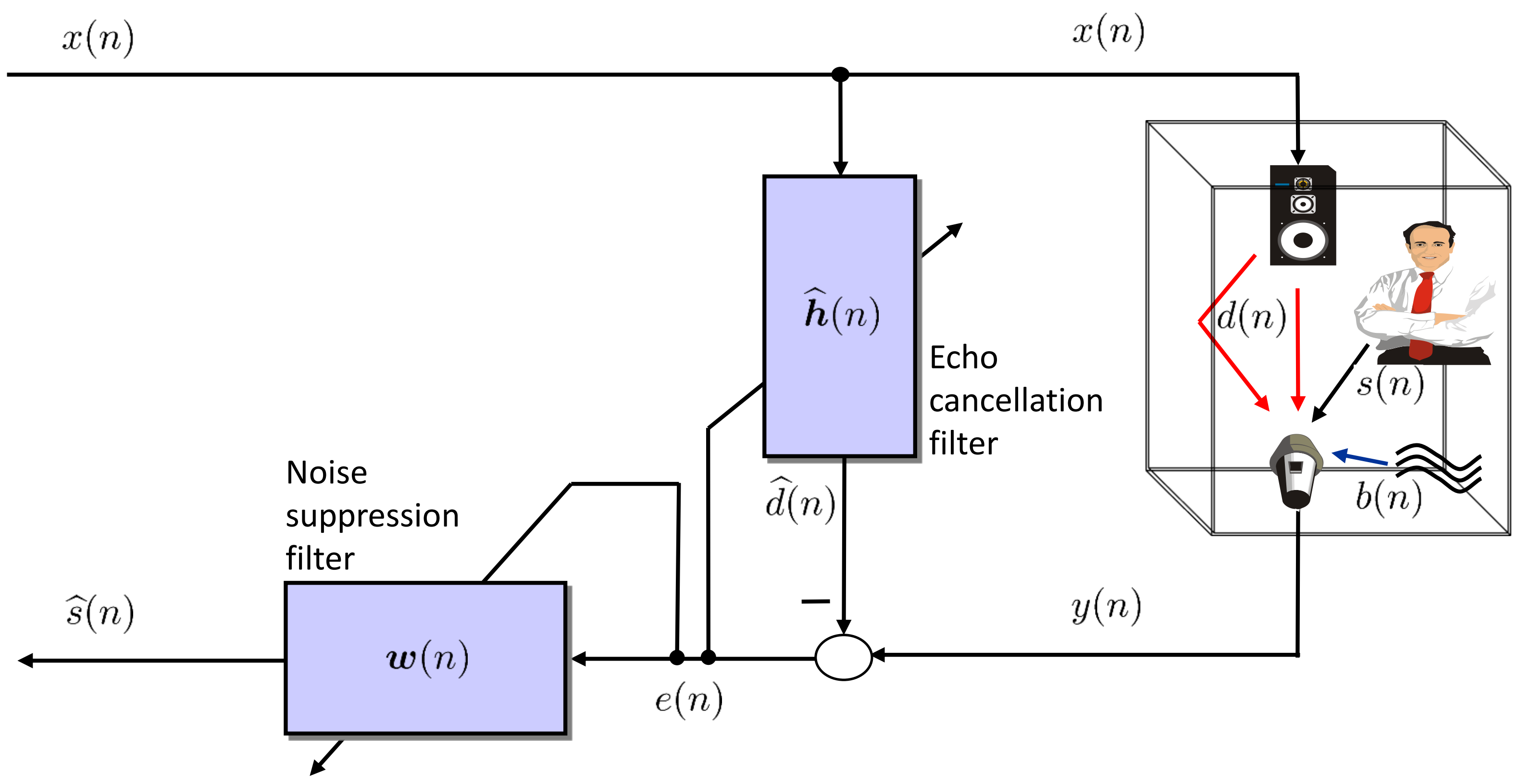
- Engine noise
- Wind noise
- Tire noise

Internal components:

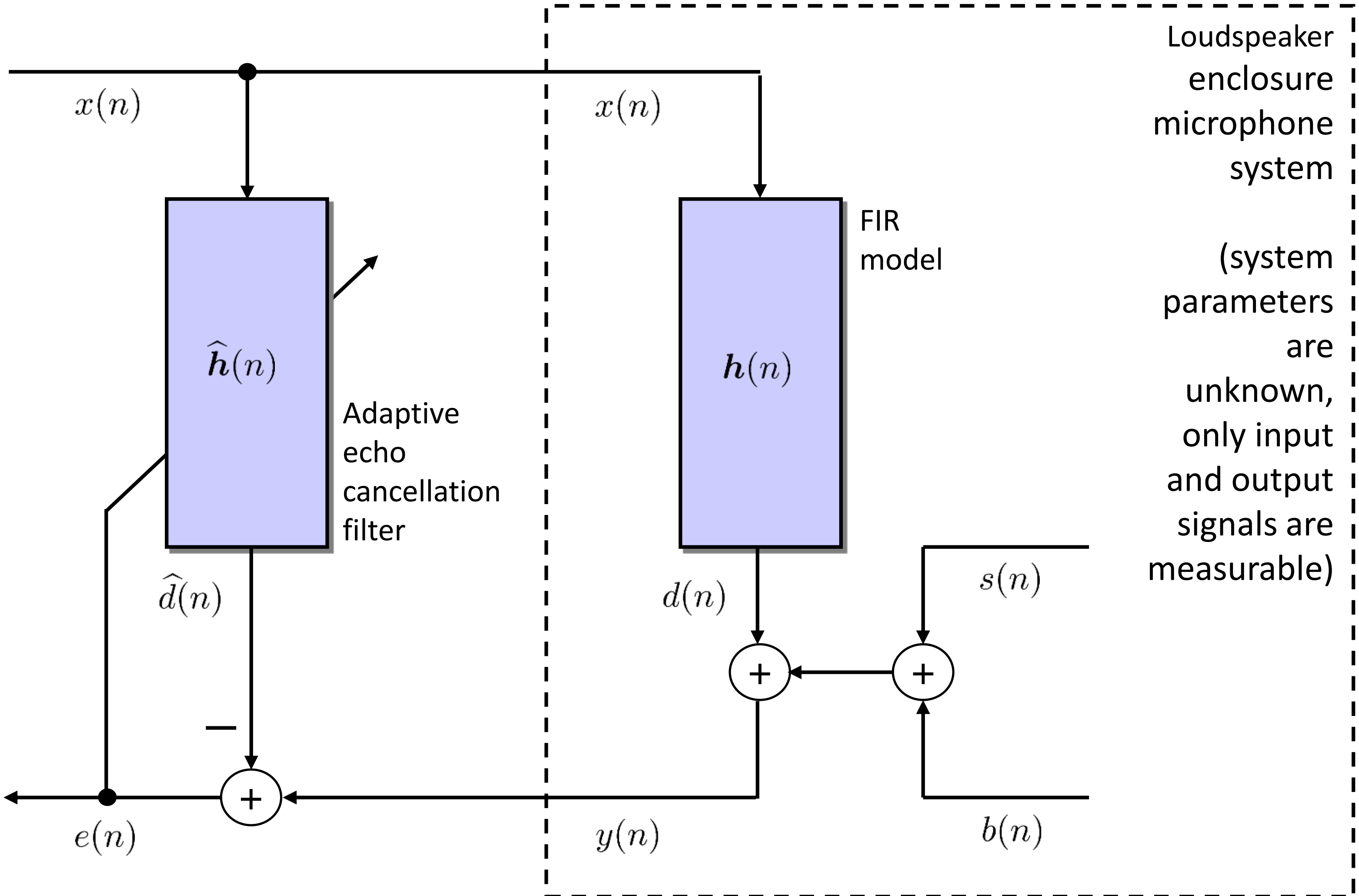
- Air conditioning
- Defrost



A Basic System With Two Adaptive Filters



An Adaptive Filter for Cancellation of Acoustical Echoes



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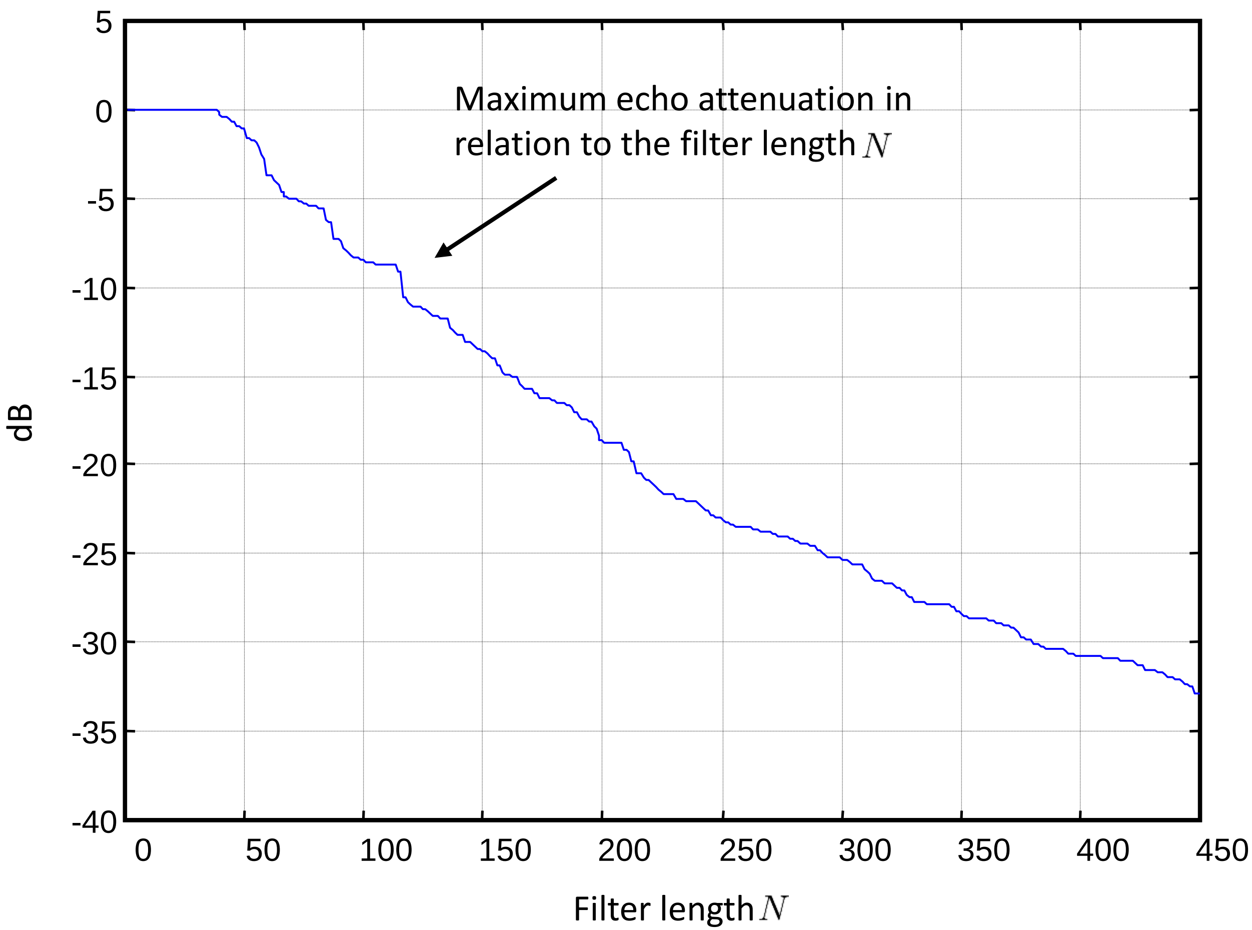
Maximal Achievable Echo Reduction – Part 1

Derivation during the lecture …

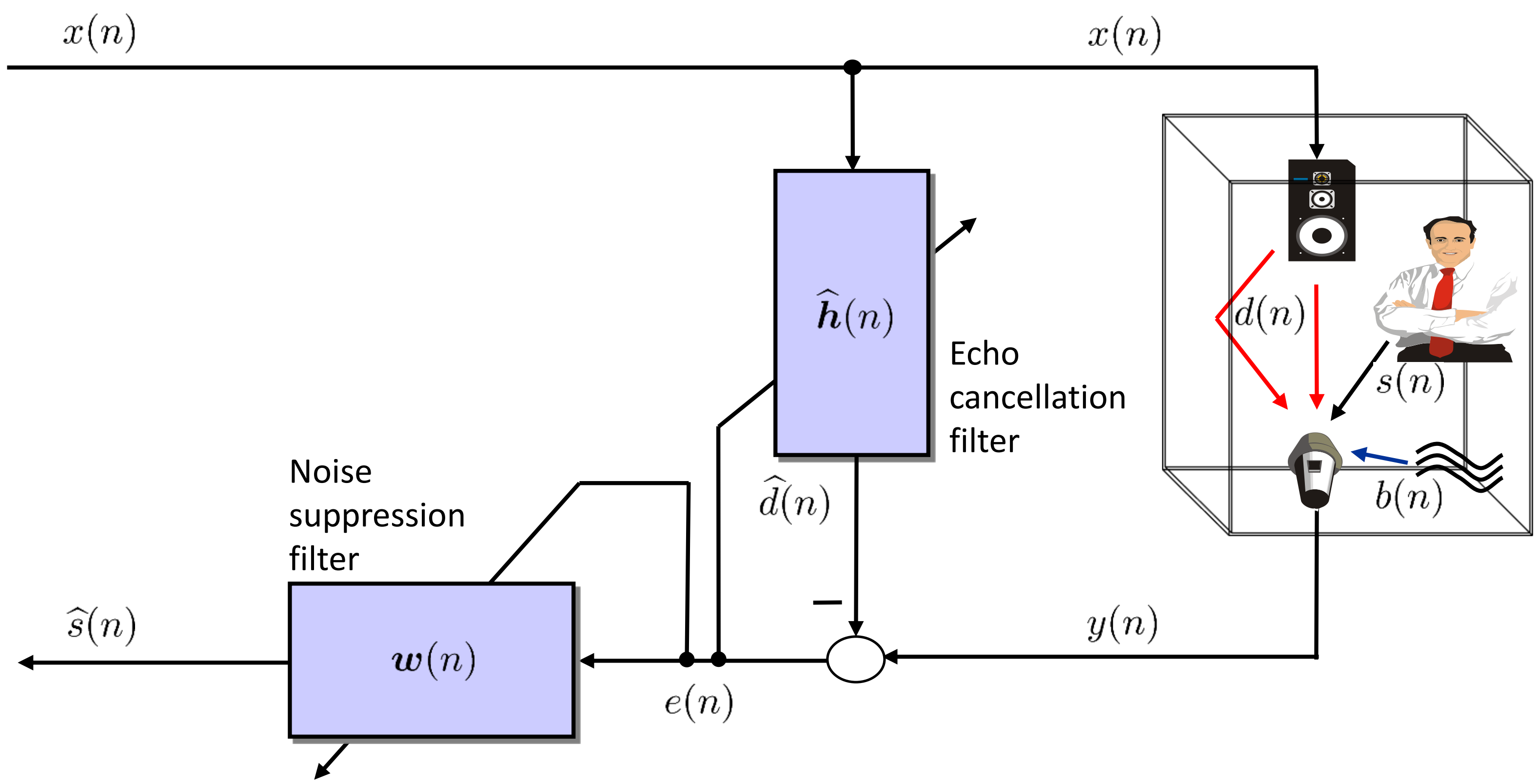
Maximal Achievable Echo Reduction – Part 2



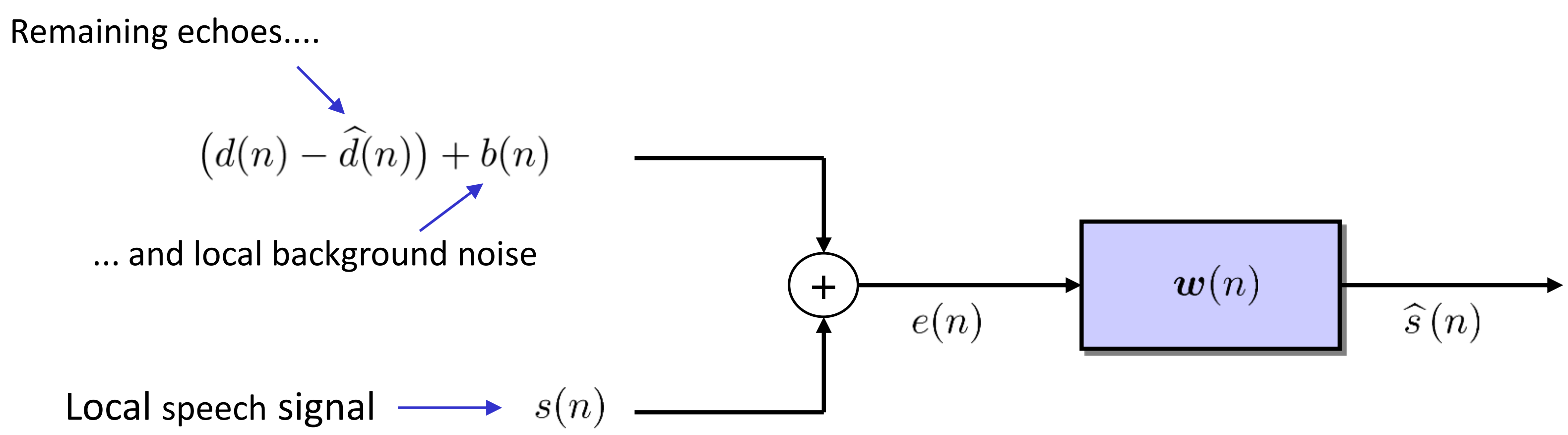
Boundary conditions:

- 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

A Basic System With Two Adaptive Filters



Residual Echo and Noise Suppression



Approach according to Wiener (next lecture):

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

$$\mathrm{E}\left\{\left(s(n) - \widehat{s}\,(n)\right)^2\right\} \rightarrow \min \qquad \Longrightarrow \qquad W_{\text{opt}}\left(e^{j\Omega}\right) = \frac{S_{es}(\Omega)}{S_{ee}(\Omega)}$$

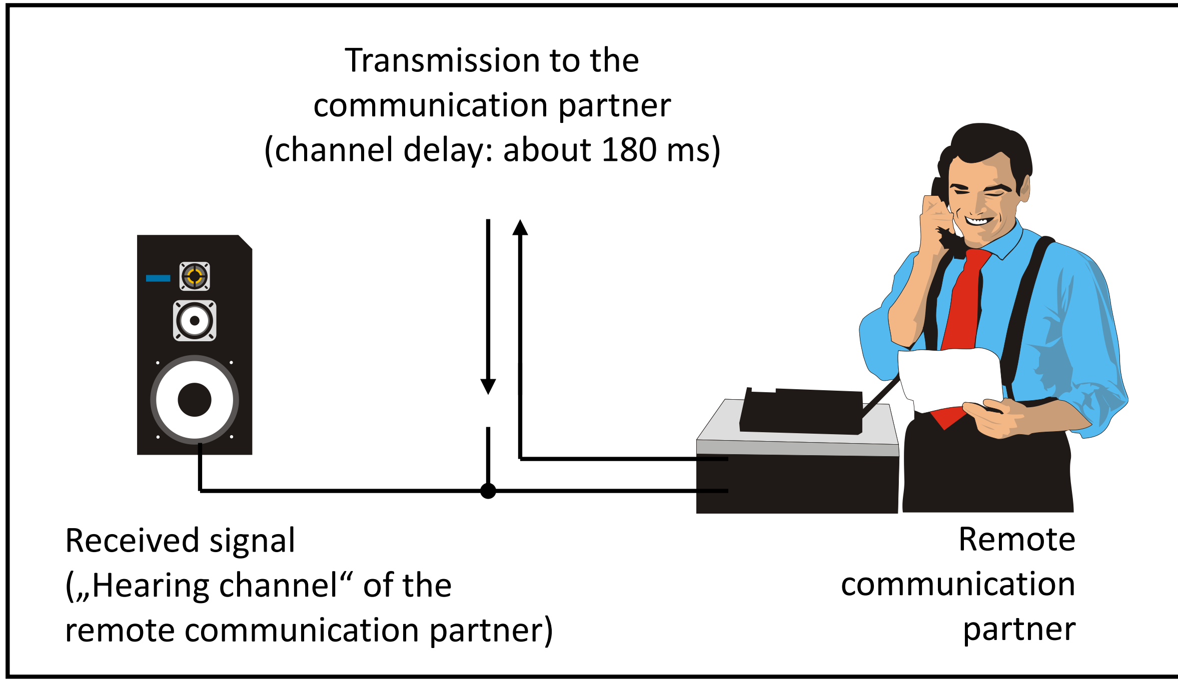
Auto power spectral density of the distorted input signal $e(n)$

A Basic System With Two Adaptive Filters – Audio Examples

Stereo signals (16 kHz):

Left:	Right:
Received signal ...	Sent signal ...

... of the remote communication partner



Initial filter convergence:

Adaptation at the beginning of the call

Double talk:

Both partners speak simultaneously

Enclosure dislocations:

Without Wiener filter

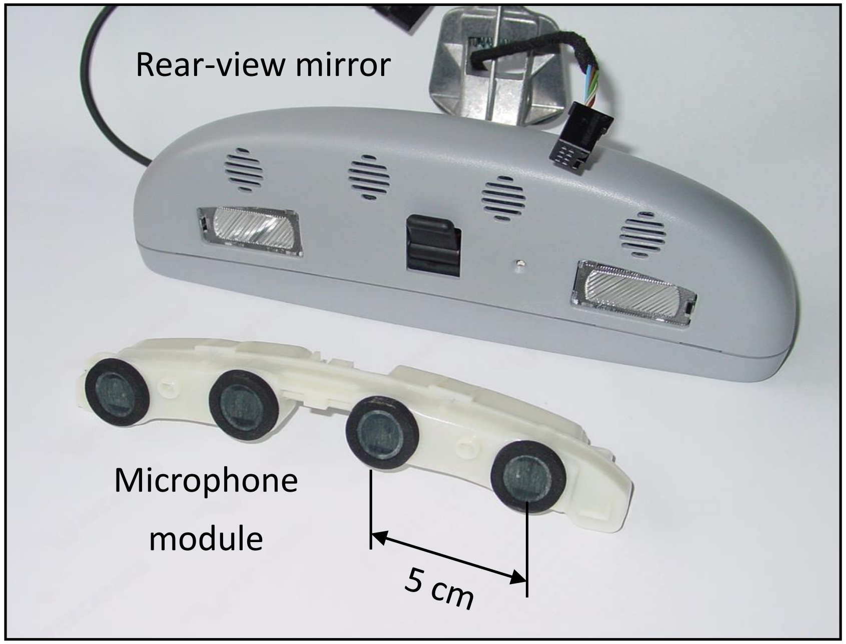
With Wiener filter

Enhanced Systems

Improvements:

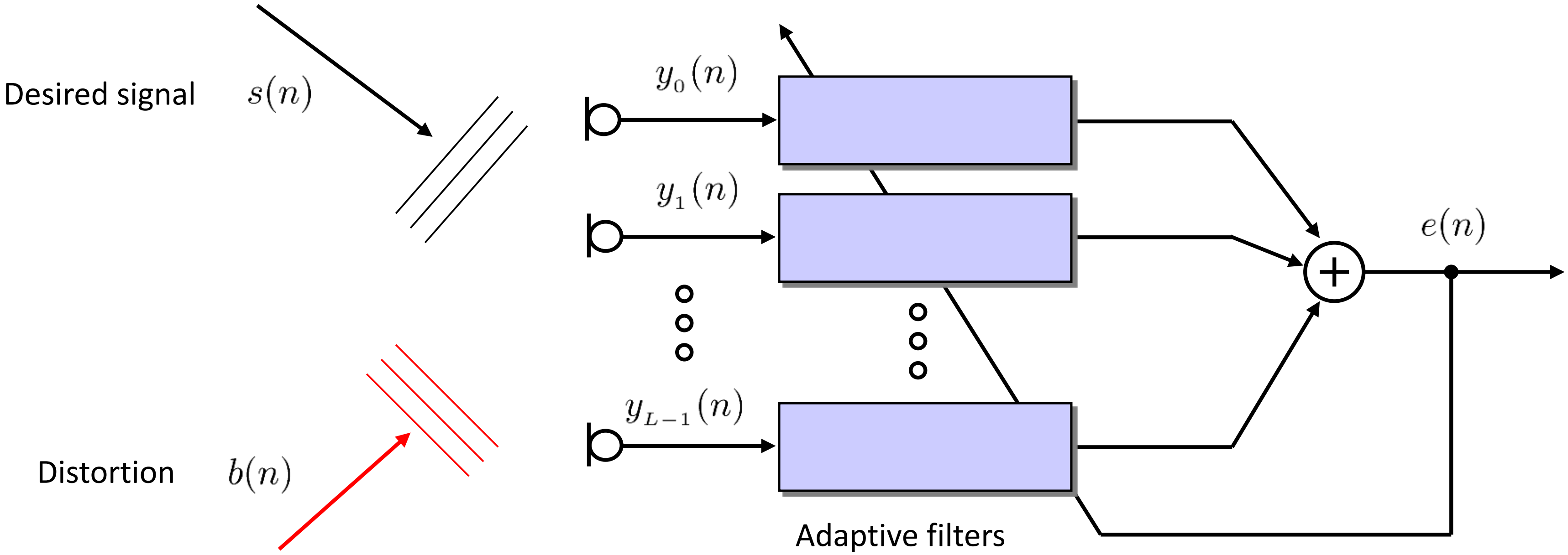
- Improved noise suppression by adaptive combination of several microphone signals (beamforming)
- Further improvements by applying adaptive filters for different kinds of distortions

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).

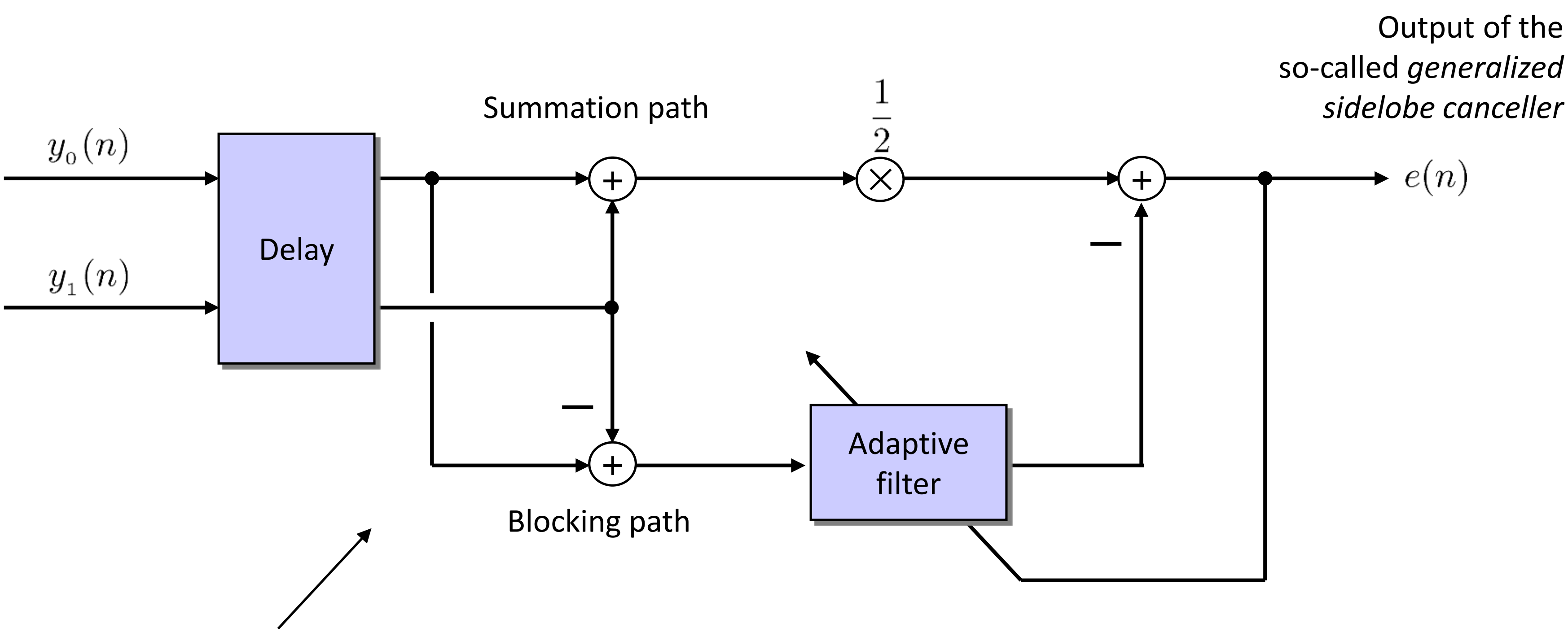
Beamforming – Introduction



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



„Griffith-Jim" beamformer (generalized sidelobe canceller)

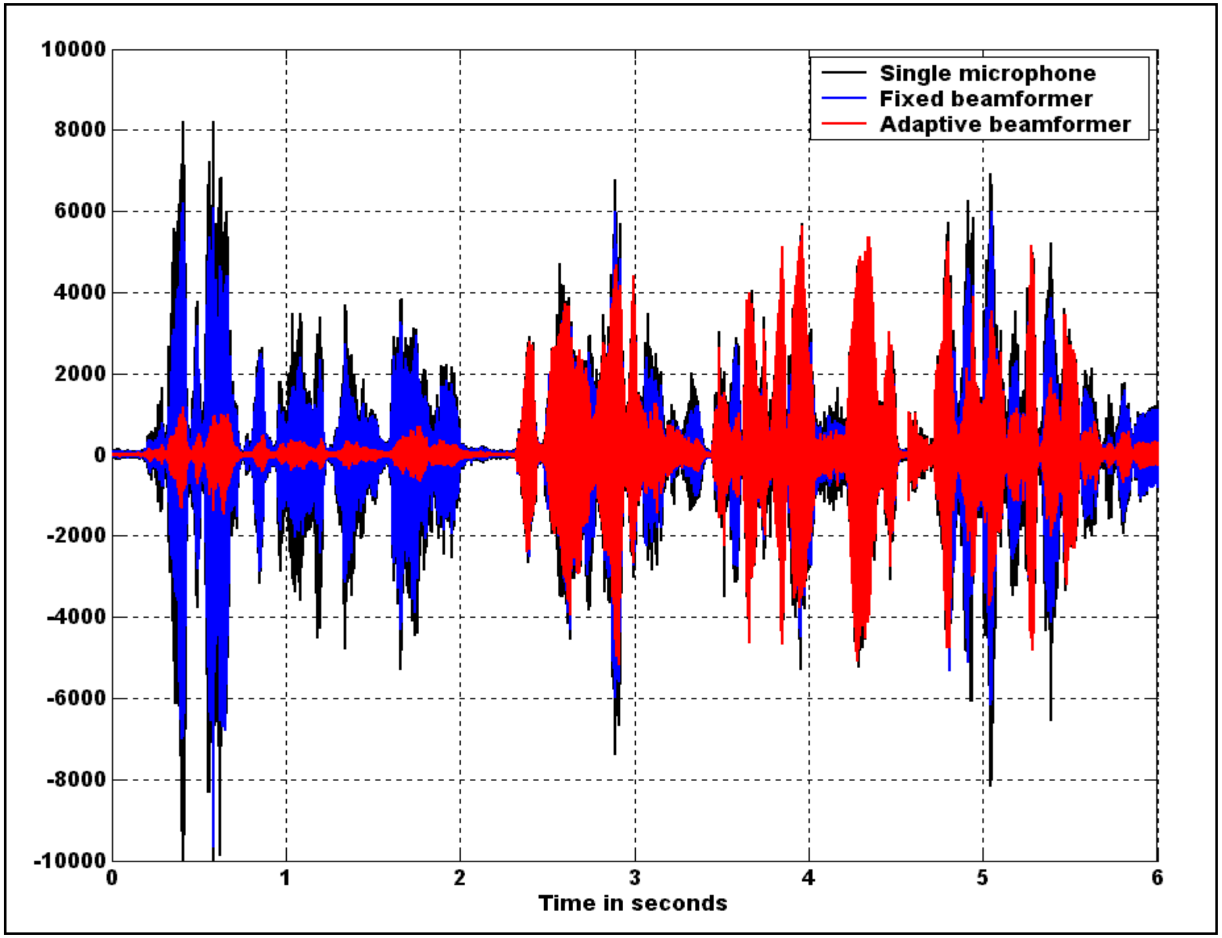
Beamforming – Audio Examples

- 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.

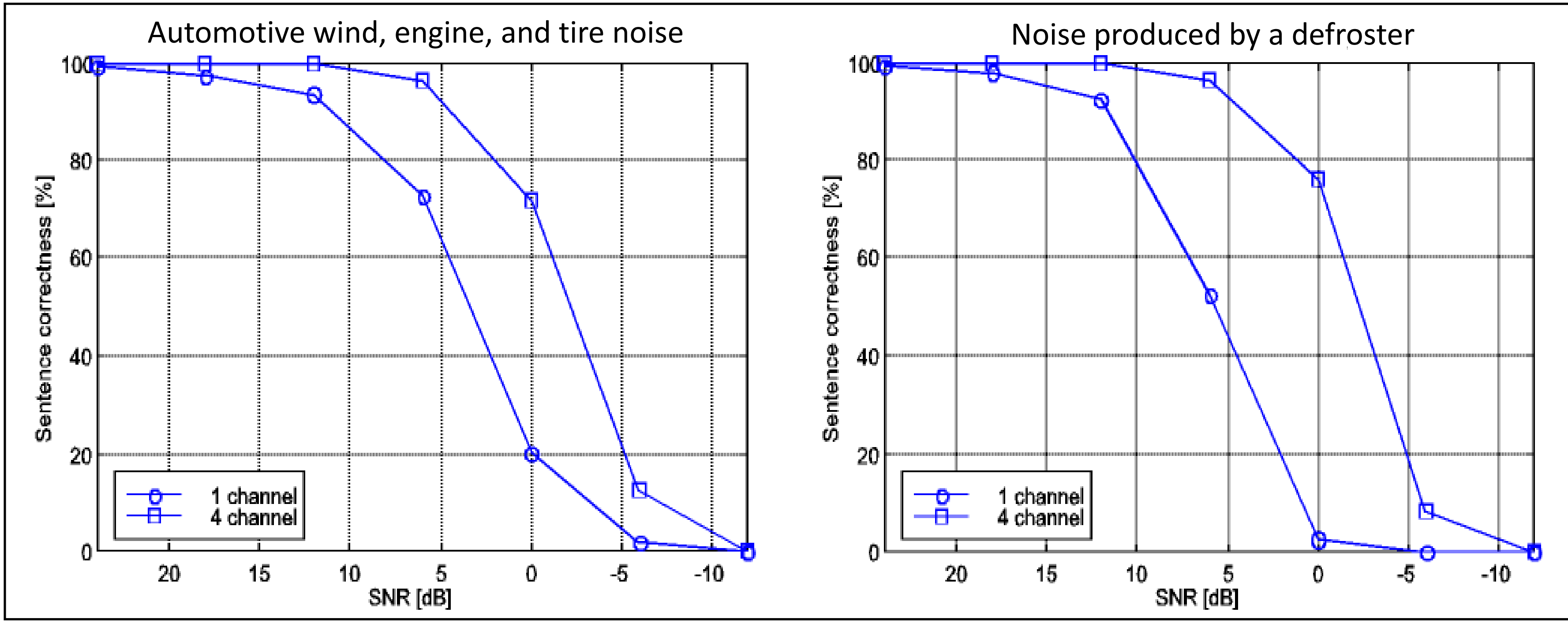
Single microphone

Fixed beamformer

Adaptive beamformer



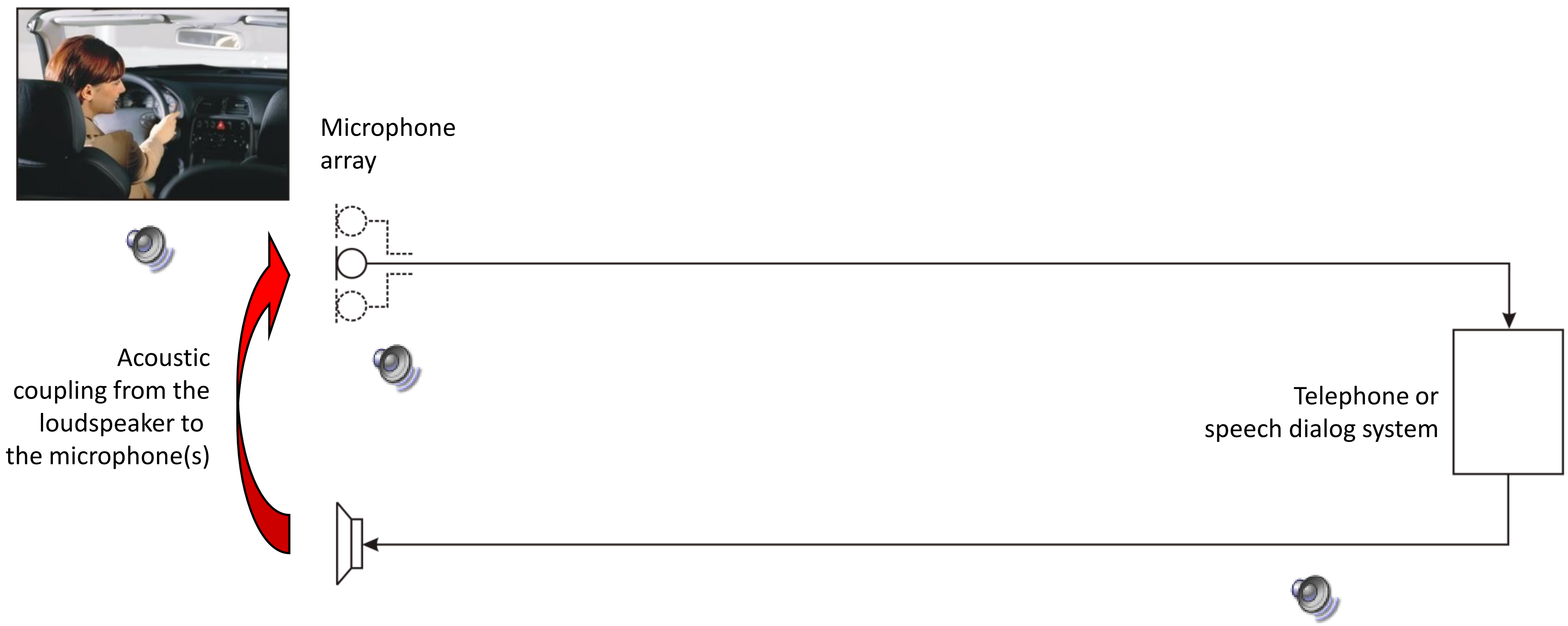
Beamforming – Performance of Speech Recognition Systems



With permission from Eberhard Hänsler, Gerhard Schmidt, *Acoustic Echo and Noise Control*, Wiley, 2004

- 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.

Involved Signal Processing Units – Start



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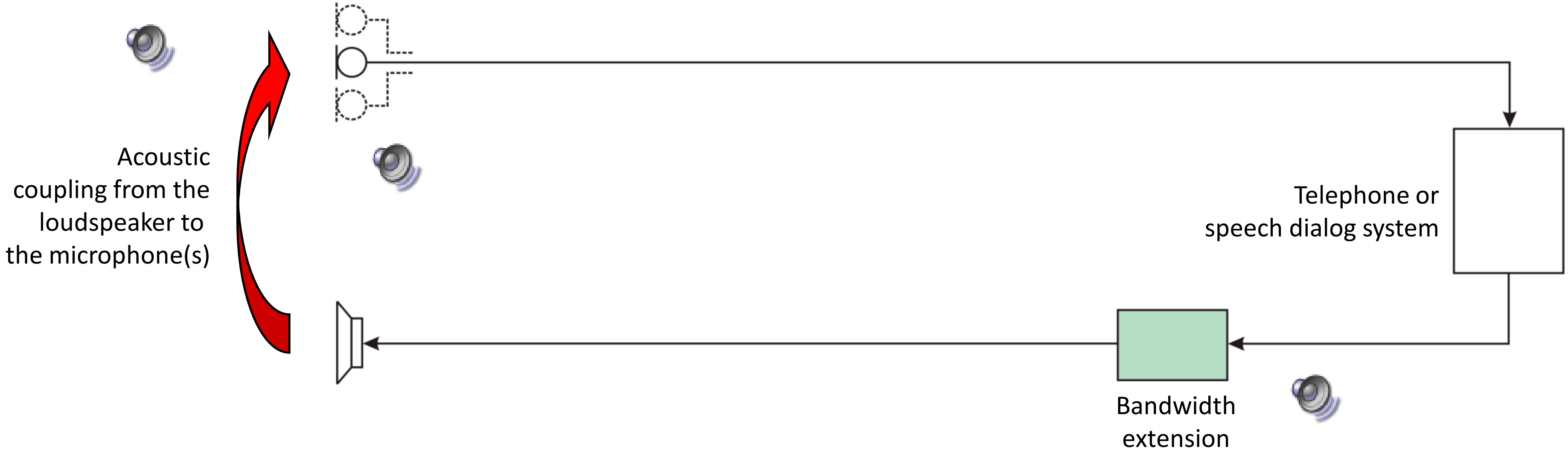
Involved Signal Processing Units – Bandwidth Extension

Bandwidth extension

Missing frequency components were estimated and resynthesized.

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

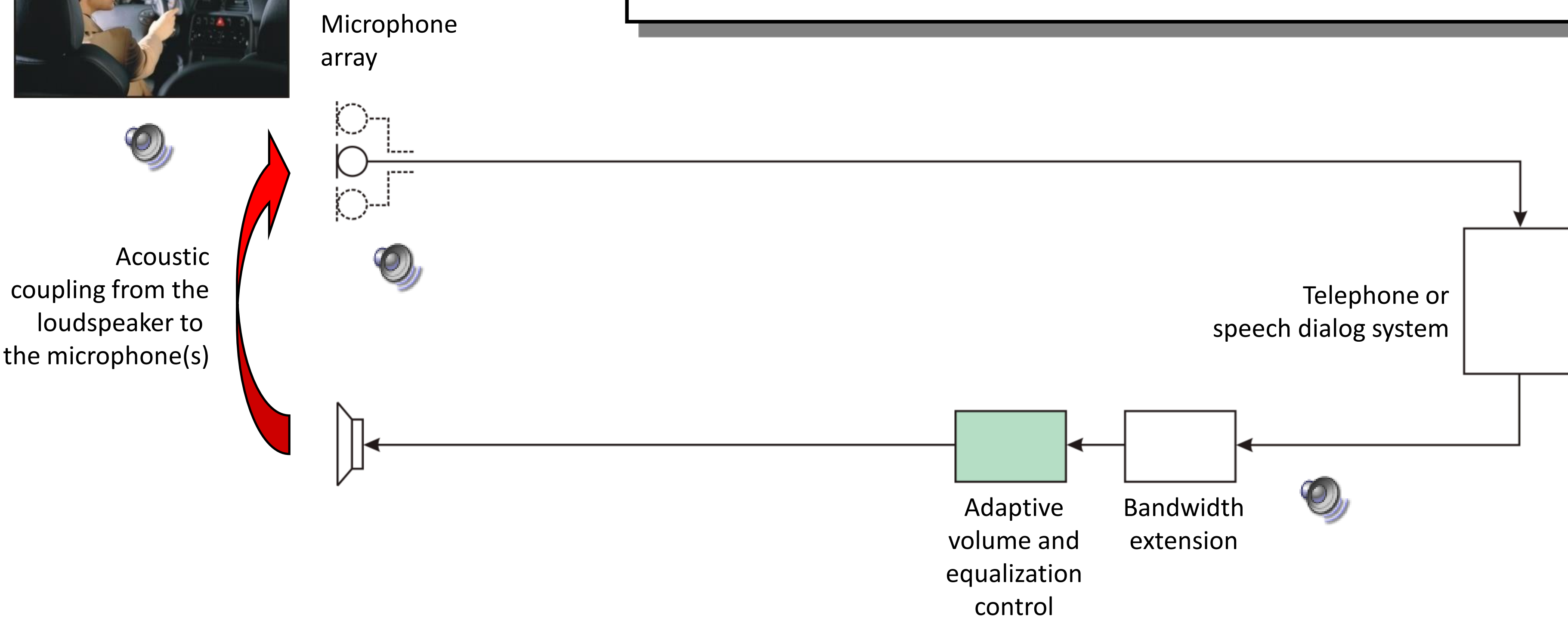
Microphone
array



Involved Signal Processing Units – Automatic Gain and Equalization Adjustment

Volume and equalization control

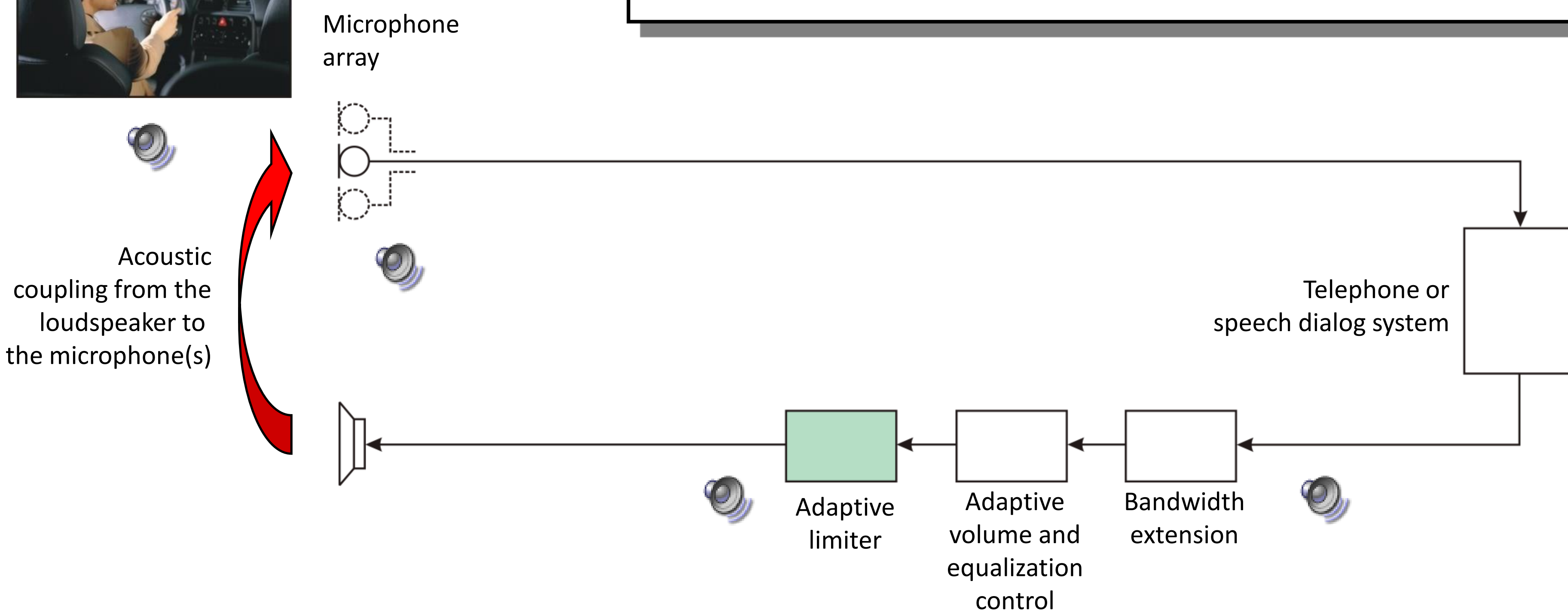
The (broadband) playback volume is adjusted automatically with respect to the noise measured in the car. In addition also the spectrum can be shaped in order to improve the perceived signal quality.



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).

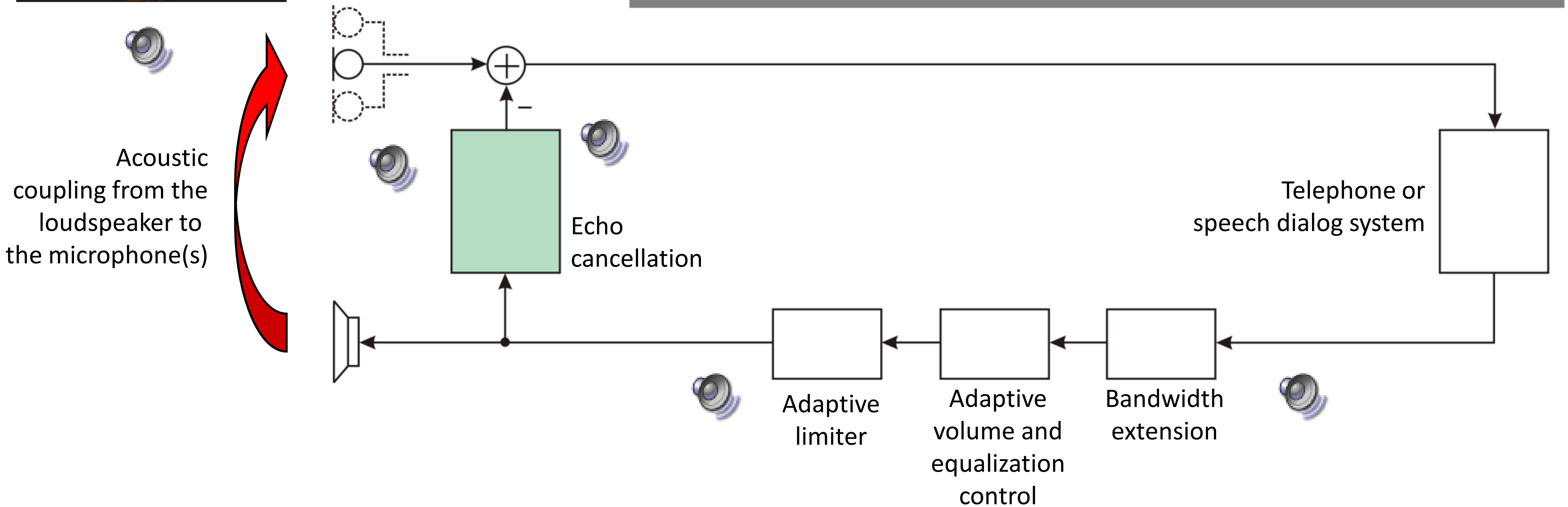


Involved Signal Processing Units – Echo Cancellation

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.

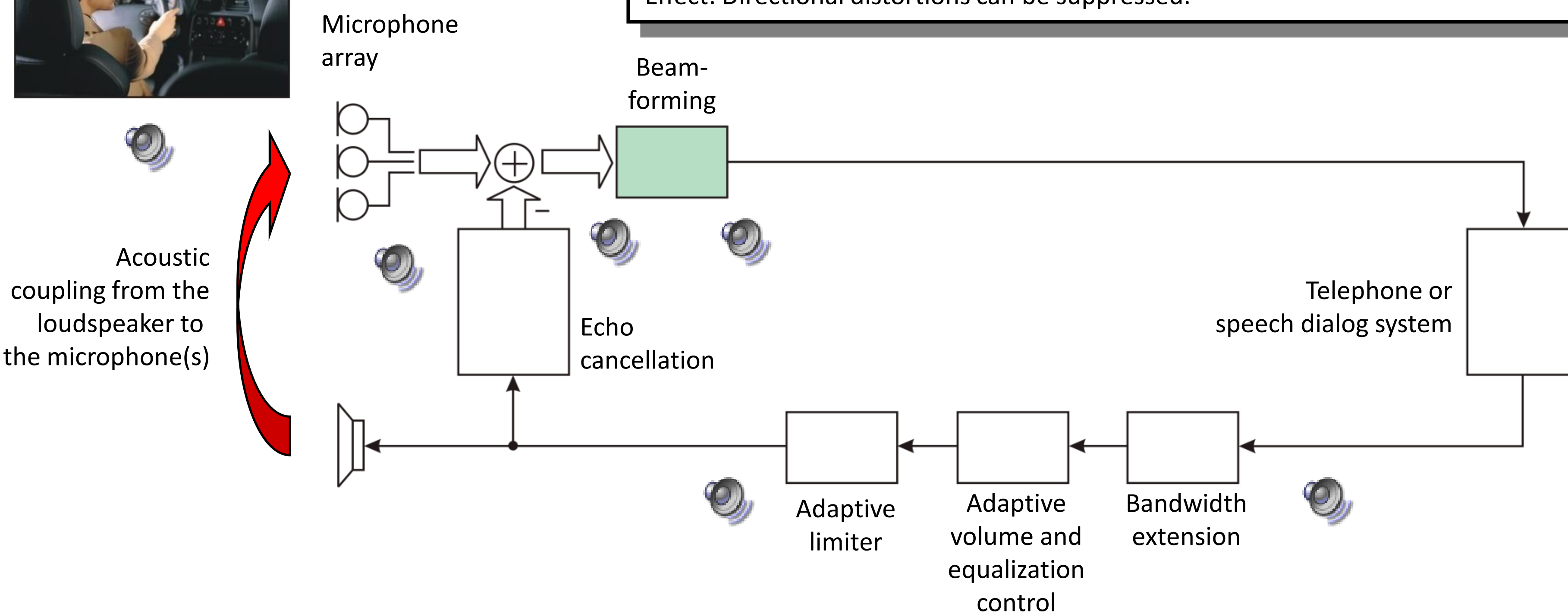


Involved Signal Processing Units – Beamforming

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.

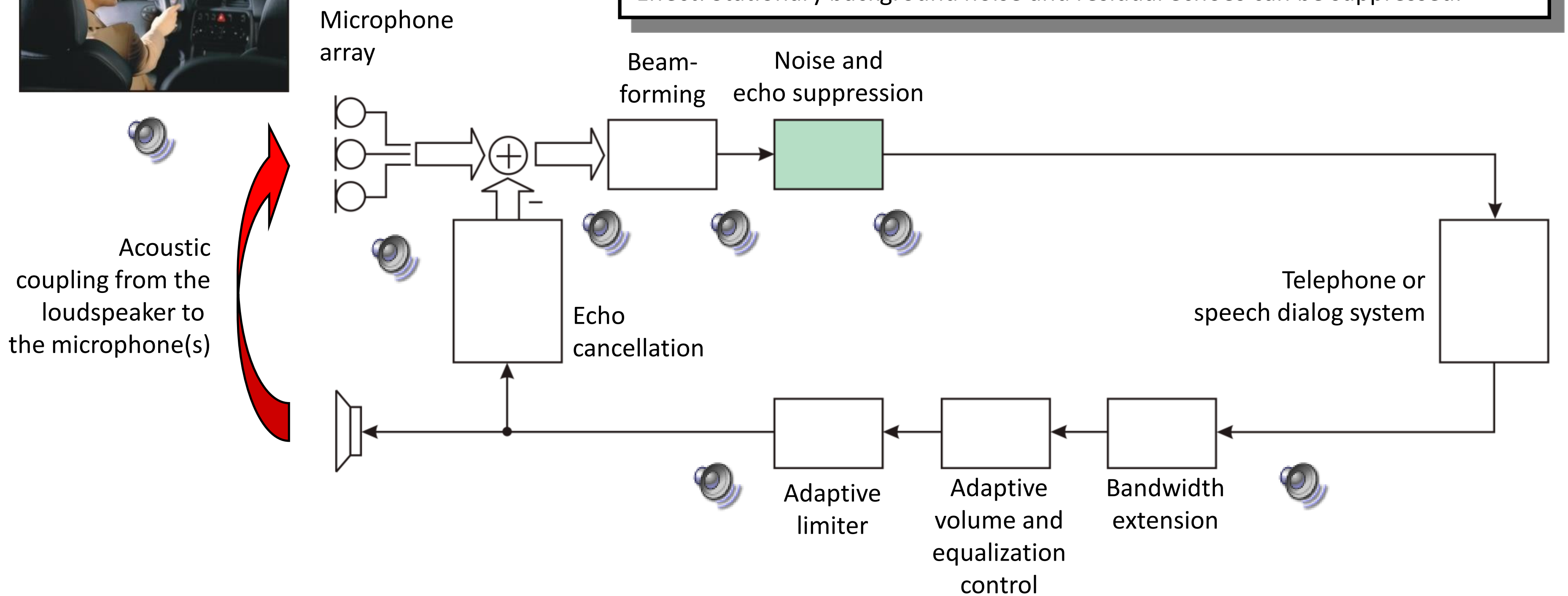


Involved Signal Processing Units – Noise and Residual Echo Suppression

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.

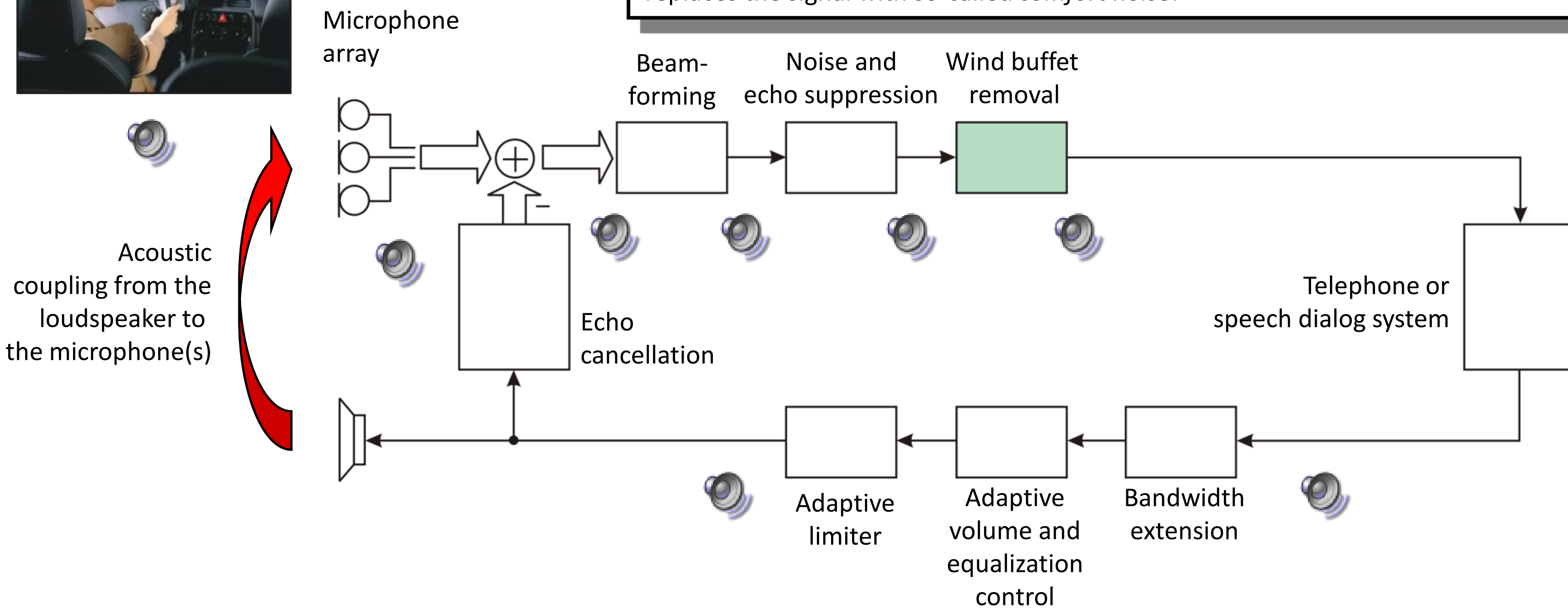


Involved Signal Processing Units – Wind Buffet Removal

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*.

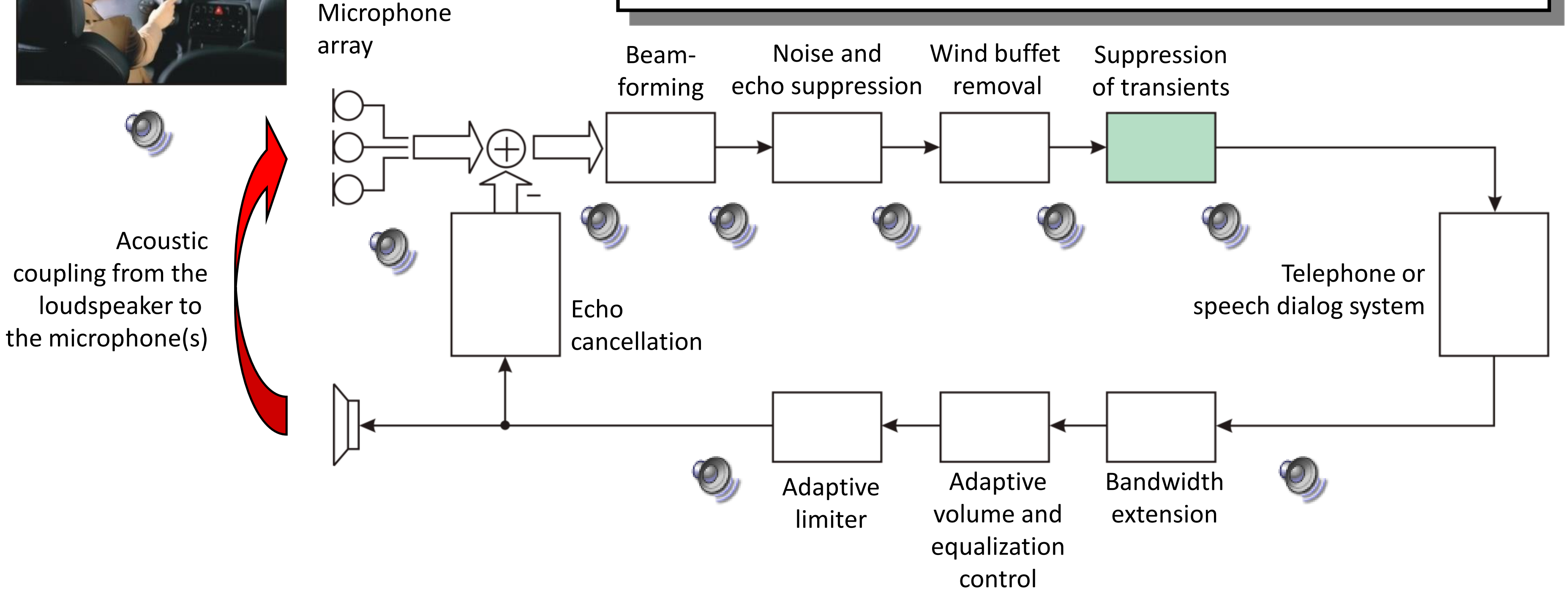


Involved Signal Processing Units – Removal of "Transients"

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.

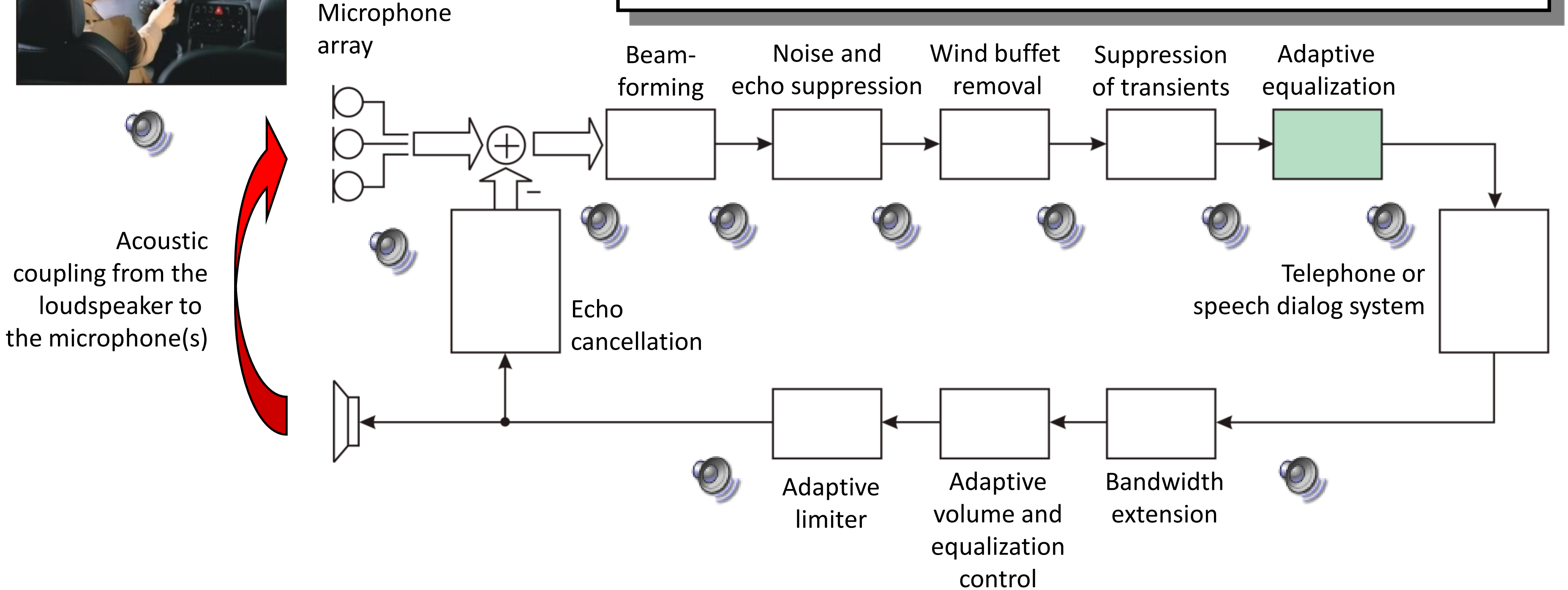


Involved Signal Processing Units – Adaptive Equalization

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.



Involved Signal Processing Units – Summary

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Missing frequency components were estimated and resynthesized.

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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.

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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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Wind buffet suppression

Open windows and defrost on cause might cause wind buffets.

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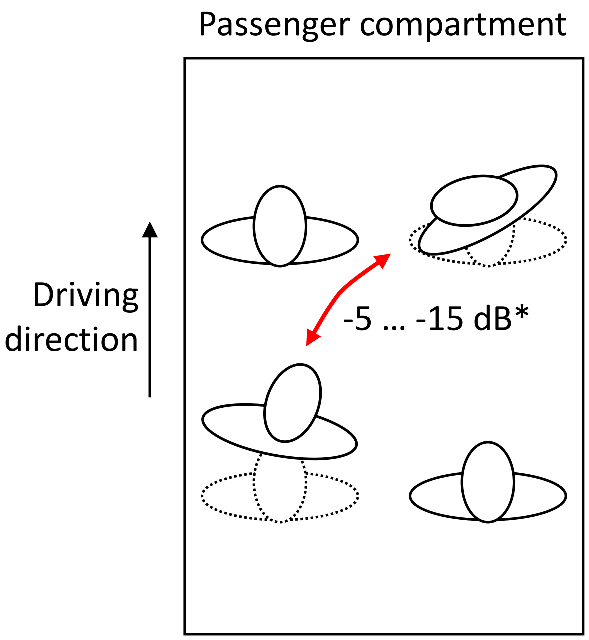
Part 2

In-Car Communication Systems

Motivation



*Acoustic loss (referred to the ear of the driver)

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.

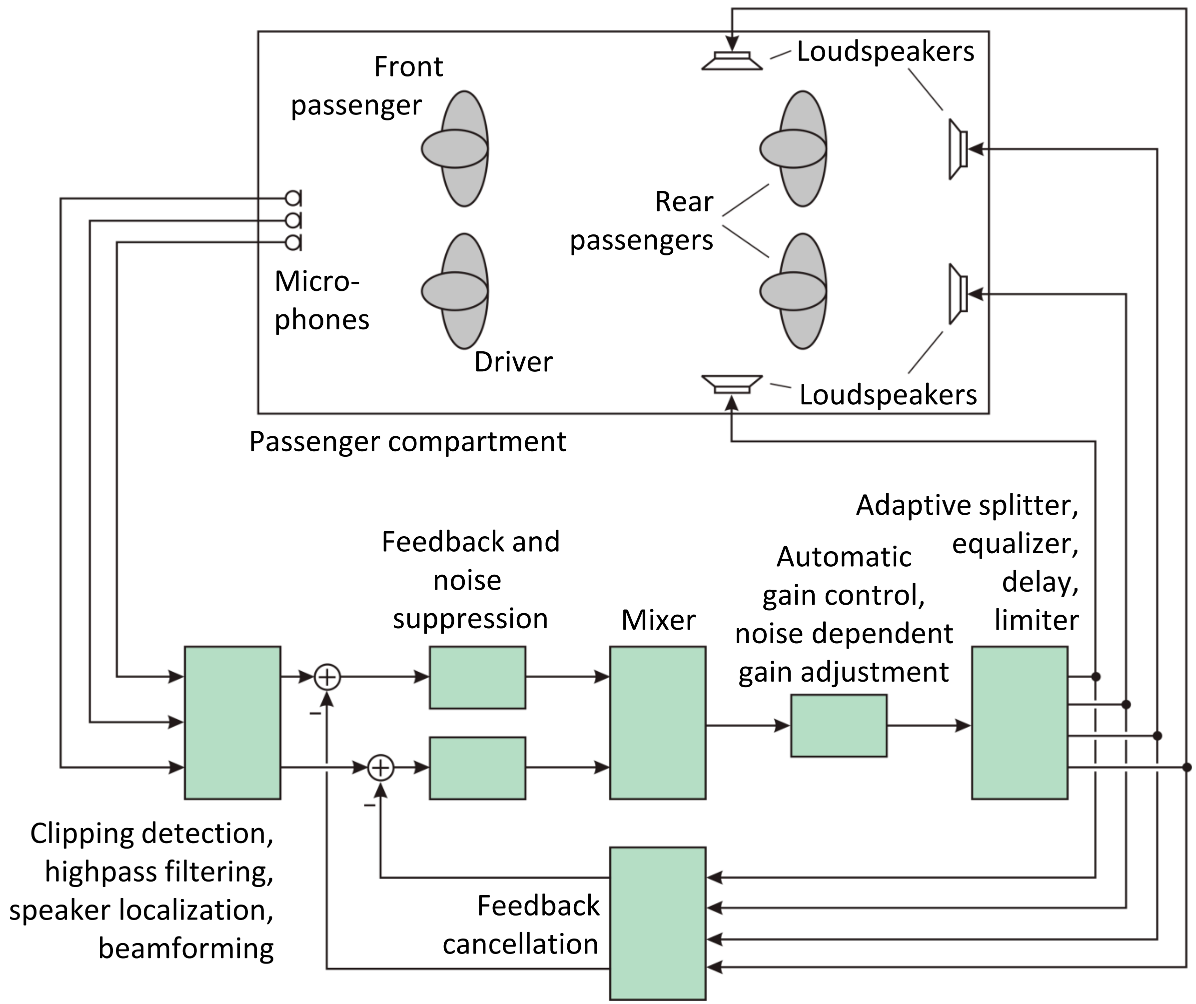
Application:

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

Algorithmic Overview

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.



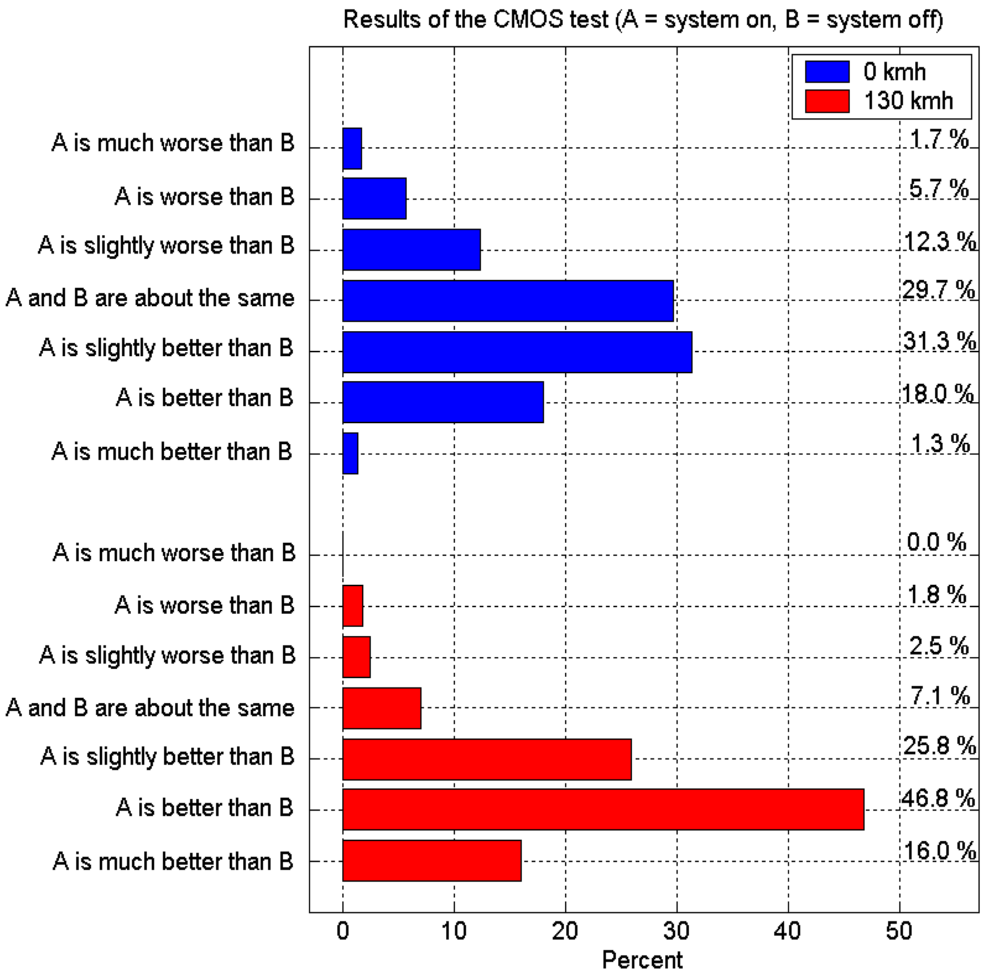
Results of a Comparison Mean Opinion Score (CMOS) Test

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



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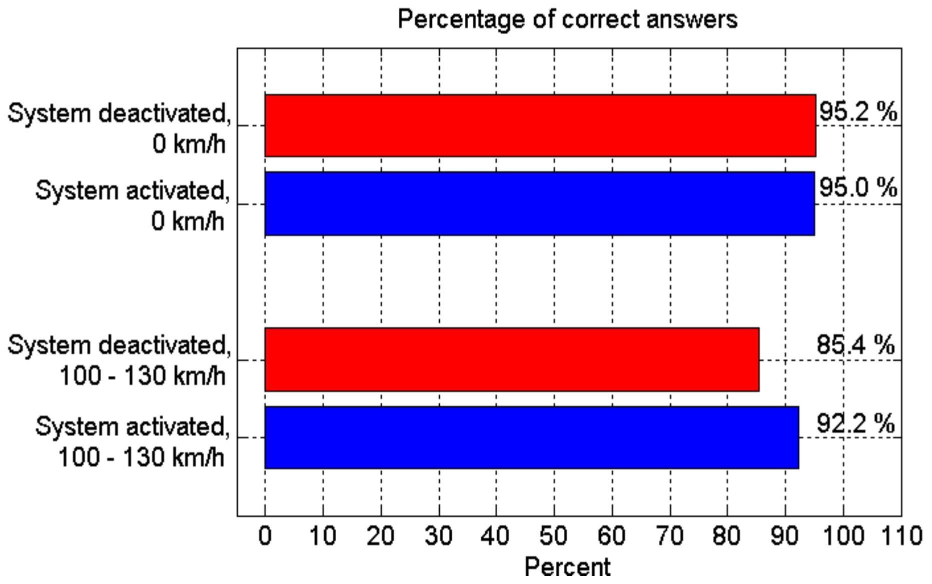
Diagnostic Rhyme Tests (DRT) and Modified Rhyme Tests (MRT)

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).



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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