

# Adaptive Filters – Wiener Filter

**Gerhard Schmidt**

Christian-Albrechts-Universität zu Kiel  
Faculty of Engineering  
Institute of Electrical and Information Engineering  
Digital Signal Processing and System Theory



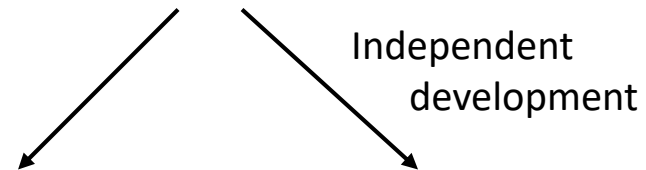
Today

## *Contents of the Lecture:*

- ❑ Introduction and motivation
- ❑ Principle of orthogonality
- ❑ Time-domain solution
- ❑ Frequency-domain solution
- ❑ Application example: noise suppression



## History and Assumptions

***Filter design by means of minimizing the squared error (according to Gauß)***

1941: A. Kolmogoroff: *Interpolation und Extrapolation von stationären zufälligen Folgen*,  
Izv. Akad. Nauk SSSR Ser. Mat. 5, pp. 3 – 14, 1941  
(in Russian)

1942: N. Wiener: *The Extrapolation, Interpolation, and Smoothing of Stationary Time Series with Engineering Applications*,  
J. Wiley, New York, USA, 1949 (originally published in  
1942 as MIT Radiation Laboratory Report)

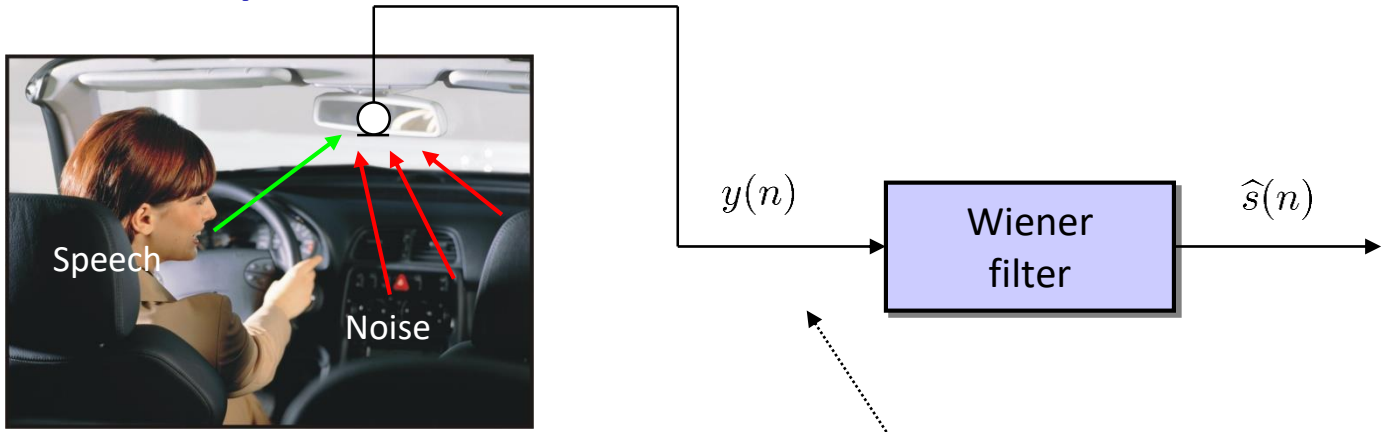
***Assumptions / design criteria:***

- ❑ Design of a filter that separates a desired signal optimally from additive noise
- ❑ Both signals are described as stationary random processes
- ❑ Knowledge about the statistical properties up to second order is necessary

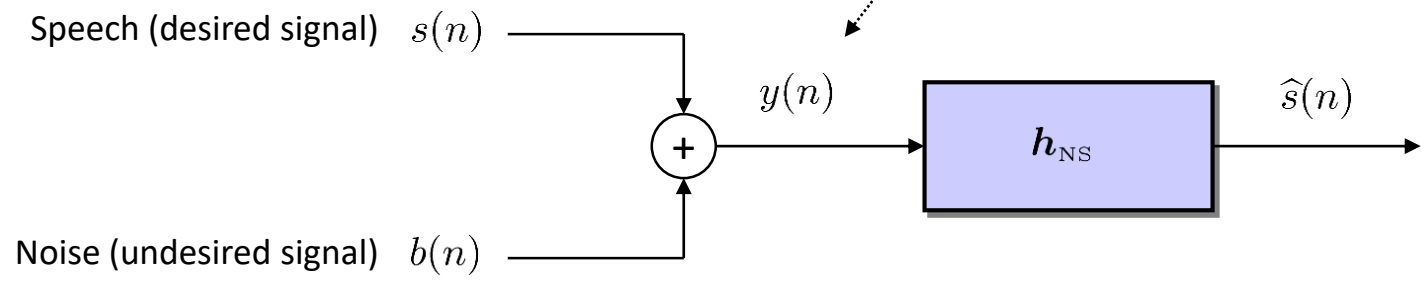
# Application Examples – Part 1

## Noise Suppression

### Application example:



### Model:

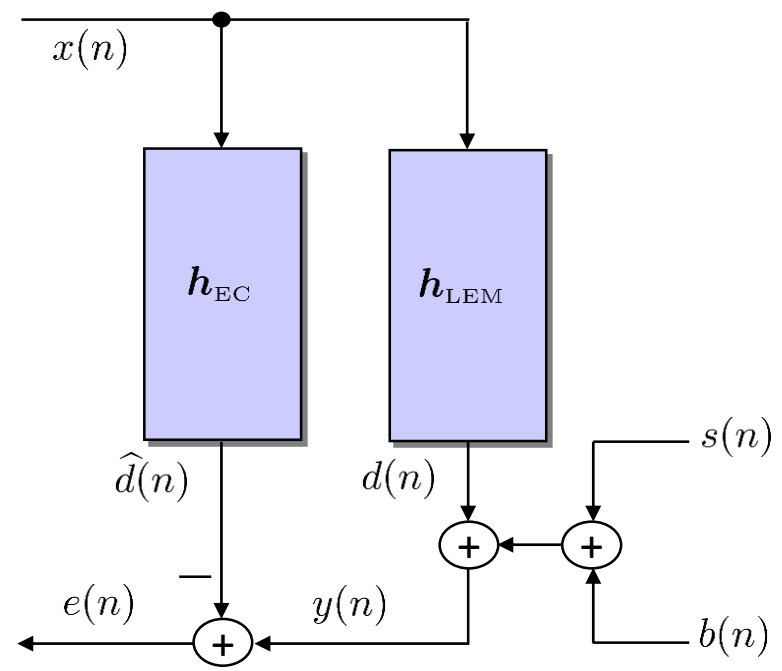


*The Wiener solution is often applied in a “block-based fashion”.*

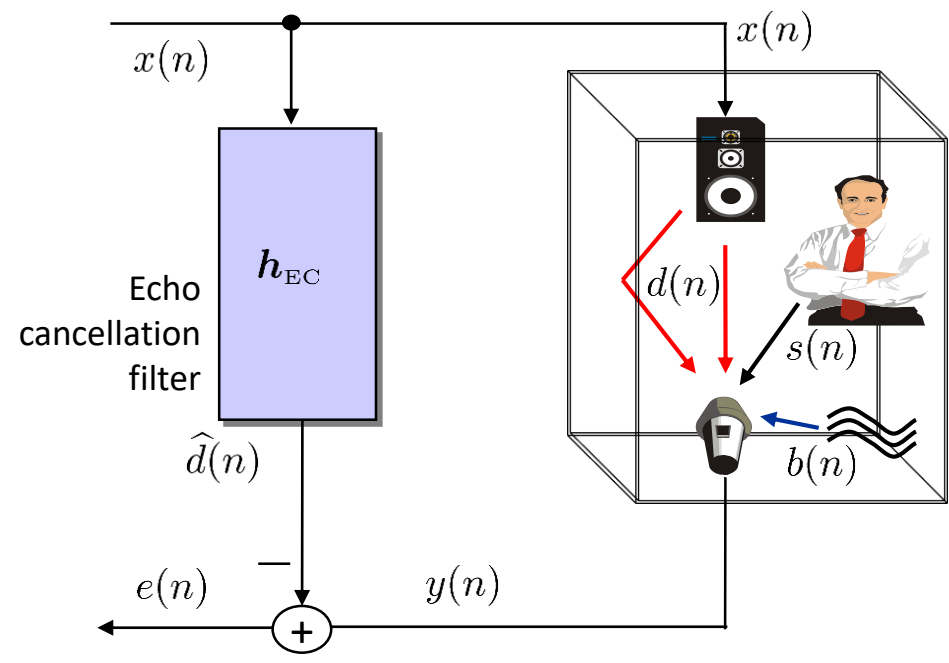
# Application Examples – Part 2

## Echo Cancellation

**Model:**



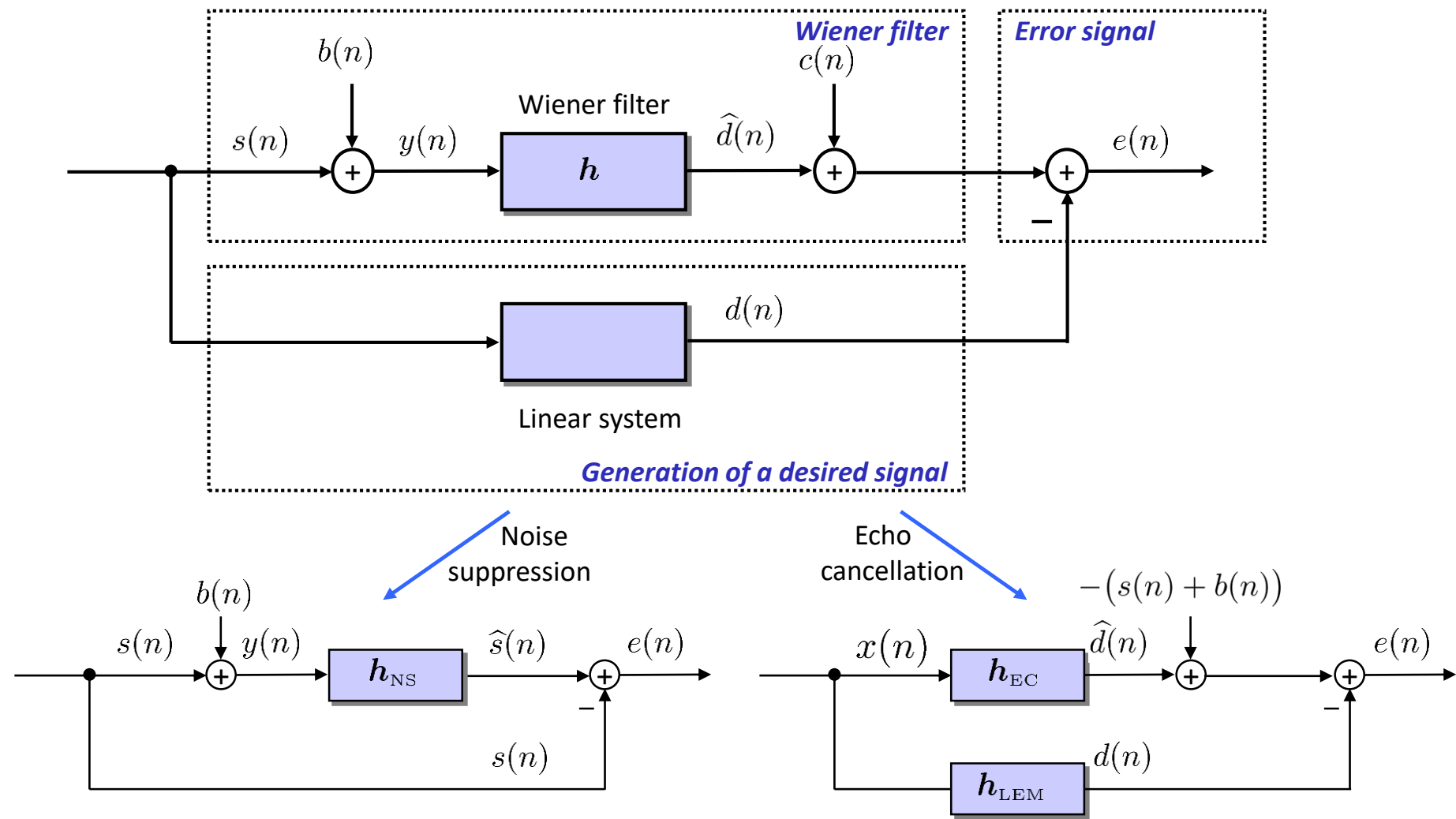
**Application example:**



*The echo cancellation filter has to converge in an iterative manner (new = old + correction) towards the Wiener solution.*

# Generic Structure

## Noise Reduction and System Identification



## Books

### **Main text:**

- E. Hänsler / G. Schmidt: *Acoustic Echo and Noise Control – Chapter 5 (Wiener Filter)*, Wiley, 2004

### **Additional texts:**

- E. Hänsler: *Statistische Signale: Grundlagen und Anwendungen – Chapter 8 (Optimalfilter nach Wiener und Kolmogoroff)*, Springer, 2001 (in German)
- M. S.Hayes: *Statistical Digital Signal Processing and Modeling – Chapter 7 (Wiener Filtering)*, Wiley, 1996
- S. Haykin: *Adaptive Filter Theory – Chapter 2 (Wiener Filters)*, Prentice Hall, 2002

### **Noise suppression:**

- U. Heute: *Noise Suppression*, in E. Hänsler, G. Schmidt (eds.), *Topics in Acoustic Echo and Noise Control*, Springer, 2006

## Derivation

Derivation during the lecture ...



## A Deterministic Example

Derivation during the lecture ...

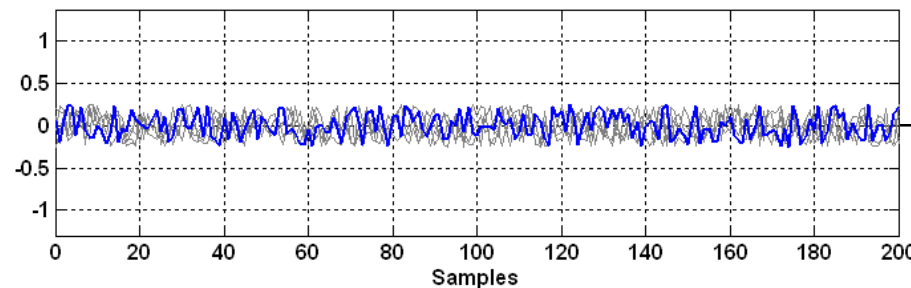
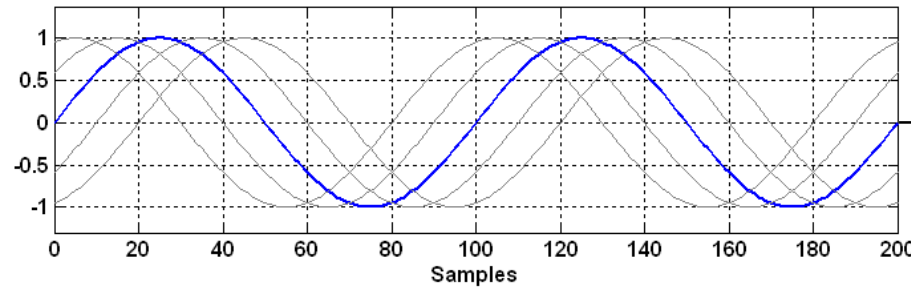
## Time-Domain Solution

Derivation during the lecture ...

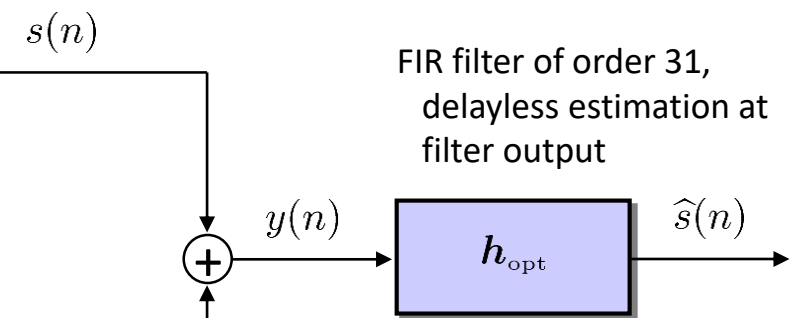
# Time-Domain Solution

## Example – Part 1

Desired signal: Sine wave with known frequency but with unknown phase, not correlated with noise



Noise: White noise with zero mean, not correlated with desired signal



FIR filter of order 31, delayless estimation at filter output

## Example – Part 2

**Wiener solution:**

$$\mathbf{h}_{\text{opt}} = \mathbf{R}_{yy}^{-1} \mathbf{r}_{yd}(0)$$

**Desired signal and noise are not correlated and have zero mean:**

$$E\{s(n) \cdot b(n+l)\} = E\{s(n)\} \cdot E\{b(n+l)\} = m_s \cdot m_b$$

$$E\{s(n)\} = E\{b(n)\} = 0$$

$$\implies E\{s(n) \cdot b(n+l)\} = 0$$

**Simplification according to the assumptions above:**

$$\mathbf{R}_{yy} = \mathbf{R}_{ss} + \mathbf{R}_{bb}$$

$$\mathbf{r}_{yd}(0) = \mathbf{r}_{ys}(0) = \mathbf{r}_{ss}(0) + \underbrace{\mathbf{r}_{bs}(0)}_0 = \mathbf{r}_{ss}(0)$$

**Wiener solution (modified):**

$$\mathbf{h}_{\text{opt}} = (\mathbf{R}_{ss} + \mathbf{R}_{bb})^{-1} \mathbf{r}_{ss}(0)$$

# Time-Domain Solution

## Example – Part 3

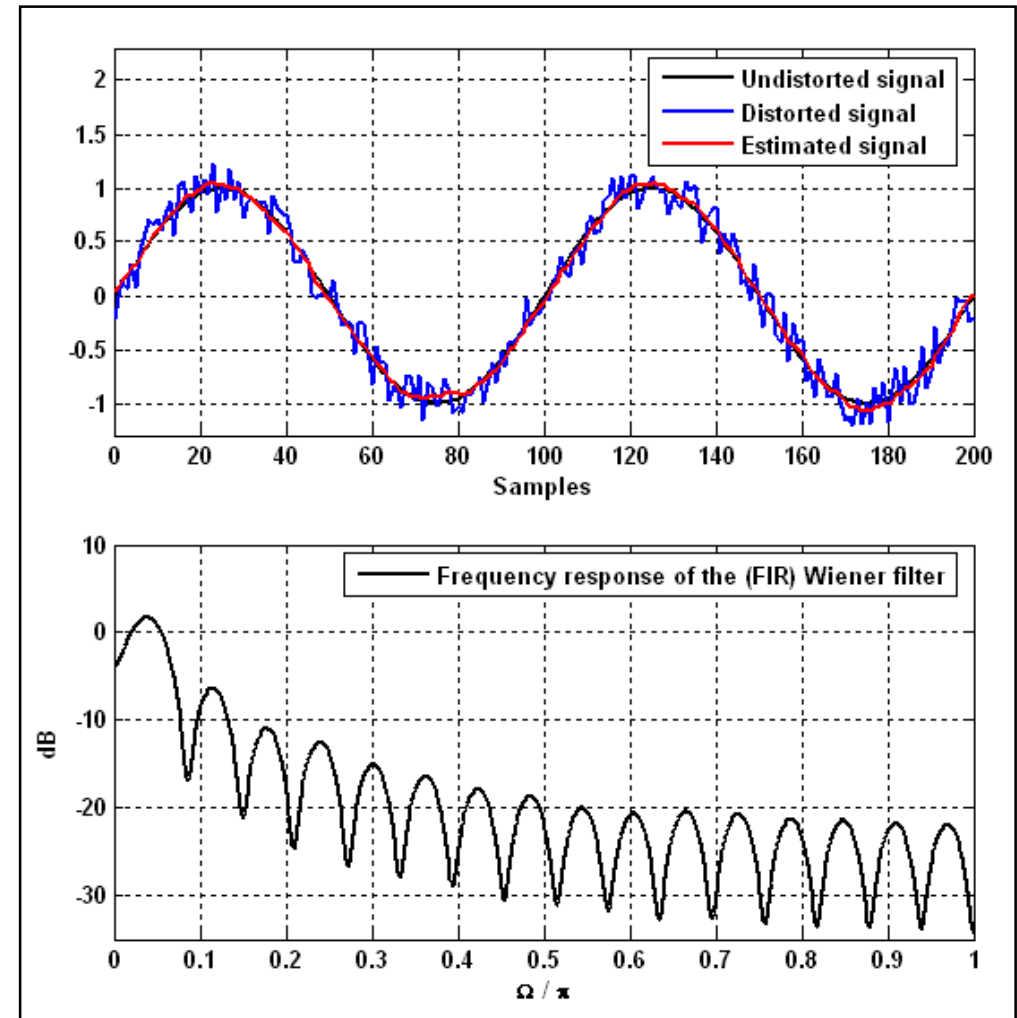
### Input signals:

Excitation: sine wave

Noise: white noise

### Assumptions:

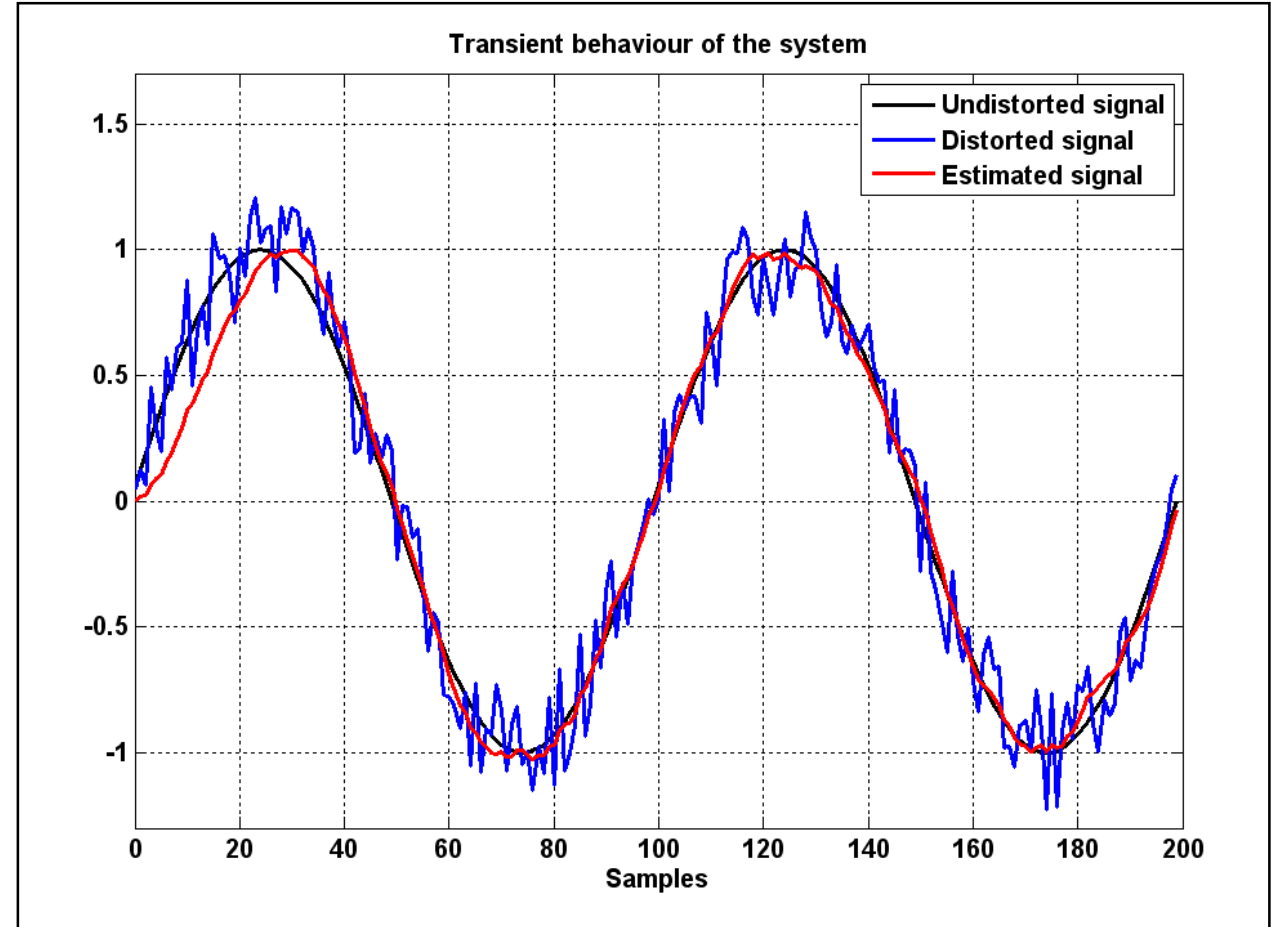
- Knowledge of the mean values and of the autocorrelation functions of the desired and of the undesired signal
- Desired signal and noise are not correlated
- Desired signal and noise have zero mean
- 32 FIR coefficients should be used by the filter



# Time-Domain Solution

## Example – Part 4

- After a short initialization time the noise suppression performs well (and does not introduce a delay!)



## Derivation – Part 1

Derivation during the lecture ...

# Error Surface

## Derivation – Part 2

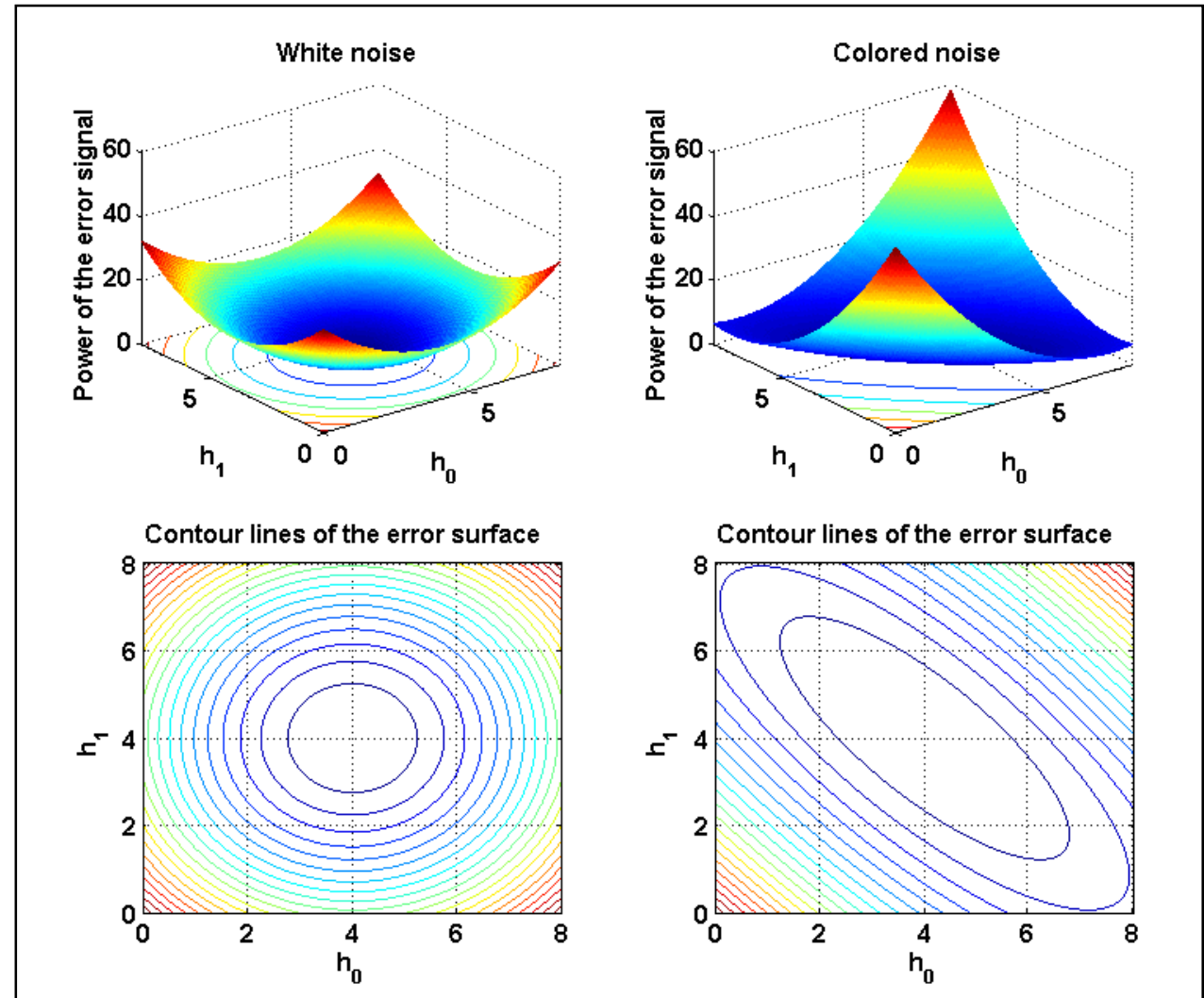
### Error surface for:

□  $R_{yy} = \begin{bmatrix} 1 & 0 \\ 0 & 1 \end{bmatrix}$

□  $R_{yy} = \begin{bmatrix} 1 & 0.8 \\ 0.8 & 1 \end{bmatrix}$

### Properties:

- Unique minimum (no local minima)
- Error surface depends on the correlation properties of the input signal





## Derivation

Derivation during the lecture ...

## Noise Suppression – Part 1

### *Frequency-domain Wiener solution (non-causal):*

$$H_{\text{opt}}(e^{j\Omega}) = \frac{S_{yd}(\Omega)}{S_{yy}(\Omega)}$$

### *Desired signal = speech signal:*

$$d(n) = s(n) \quad \implies \quad H_{\text{opt}}(e^{j\Omega}) = \frac{S_{ys}(\Omega)}{S_{yy}(\Omega)}$$

### *Desired signal and noise are orthogonal:*

$$E\{s(n_1)b(n_2)\} = 0, \text{ for all } n_1 \text{ and } n_2$$

$$\begin{aligned} \implies H_{\text{opt}}(e^{j\Omega}) &= \frac{S_{ys}(\Omega)}{S_{yy}(\Omega)} = \frac{S_{ss}(\Omega) + S_{bs}(\Omega)}{S_{yy}(\Omega)} \\ &= \frac{S_{ss}(\Omega)}{S_{yy}(\Omega)} = \frac{S_{yy}(\Omega) - S_{bb}(\Omega)}{S_{yy}(\Omega)} \\ &= 1 - \frac{S_{bb}(\Omega)}{S_{yy}(\Omega)} \end{aligned}$$

## Noise Suppression – Part 2

**Frequency-domain solution:**

$$H_{\text{opt}}(e^{j\Omega}) = 1 - \frac{S_{bb}(\Omega)}{S_{yy}(\Omega)}$$

**Approximation using short-term estimations:**

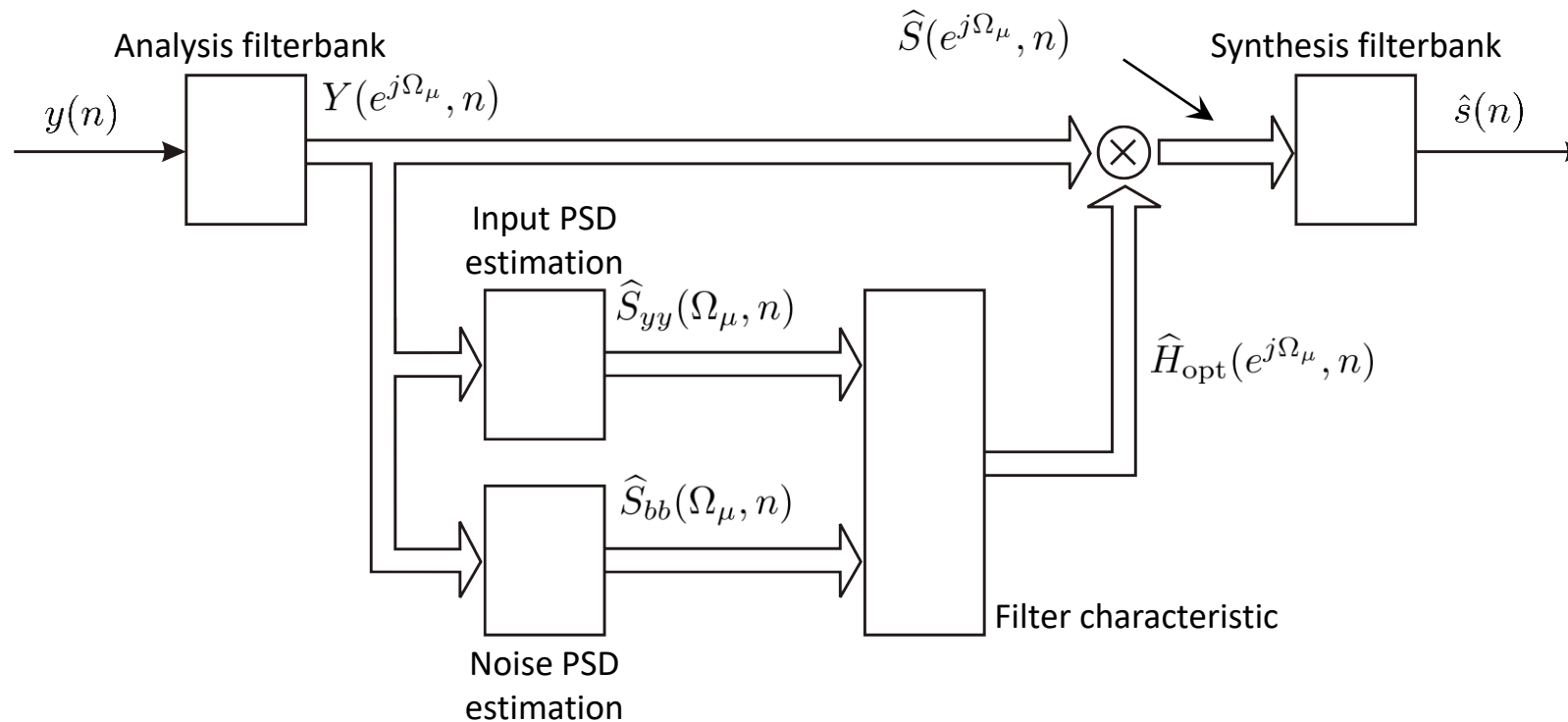
$$\hat{H}_{\text{opt}}(e^{j\Omega}, n) = \max \left\{ 0, 1 - \frac{\hat{S}_{bb}(\Omega, n)}{\hat{S}_{yy}(\Omega, n)} \right\}$$

**Practical approaches:**

- ❑ Realization using a filterbank system (time-variant attenuation of subband signals)
- ❑ Analysis filters with length of about 15 to 100 ms
- ❑ Frame-based processing with frame shifts between 1 and 20 ms
- ❑ The basic Wiener characteristic is usually „enriched“ with several extensions (overestimation, limitation of the attenuation, etc.)

## Noise Suppression – Part 3

### Processing structure:



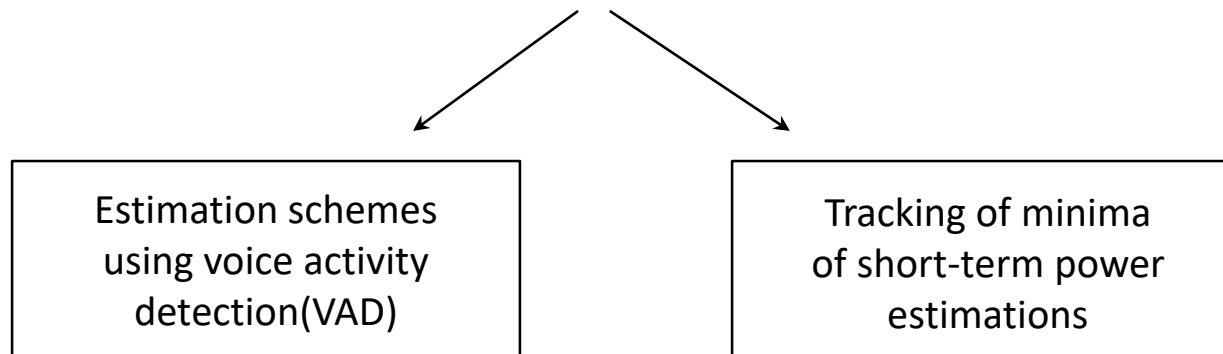
PSD = power spectral density

## Noise Suppression – Part 4

**Power spectral density estimation for the input signal:**

$$\hat{S}_{yy}(\Omega_\mu, n) = |Y(e^{j\Omega_\mu}, n)|^2$$

**Power spectral density estimation for the noise:**



## Noise Suppression – Part 5

### Schemes with voice activity detection:

$$\hat{S}_{bb}(\Omega_\mu, n) = \begin{cases} \beta \hat{S}_{bb}(\Omega_\mu, n-1) + (1-\beta) \hat{S}_{yy}(\Omega_\mu, n), & \text{during speech pauses,} \\ \hat{S}_{bb}(\Omega_\mu, n-1), & \text{else.} \end{cases}$$

### Tracking of minima of the short-term power:

$$\overline{S_{yy}}(\Omega_\mu, n) = \beta \overline{S_{yy}}(\Omega_\mu, n-1) + (1-\beta) \hat{S}_{yy}(\Omega_\mu, n)$$

*Bias correction* →

$$\hat{S}_{bb}(\Omega_\mu, n) = K \begin{cases} \max \{ S_{\min}, \hat{S}_{bb}(\Omega_\mu, n-1) \} \Delta_{\text{inc}}, & \text{if } \overline{S_{yy}}(\Omega_\mu, n) > \hat{S}_{bb}(\Omega_\mu, n-1), \\ \max \{ S_{\min}, \hat{S}_{bb}(\Omega_\mu, n-1) \} \Delta_{\text{dec}}, & \text{else.} \end{cases}$$

*Constant slightly larger than 1* ←

*Constant slightly smaller than 1* ↗

## Noise Suppression – Part 6

**Problem:**

- The short-term power of the input signal usually fluctuates faster than the noise estimate – also during speech pauses. As a result the filter characteristic opens and closes in a randomized manner, with results in tonal residual noise (so-called musical noise).

**Simple solution:**

- By inserting a fixed overestimation

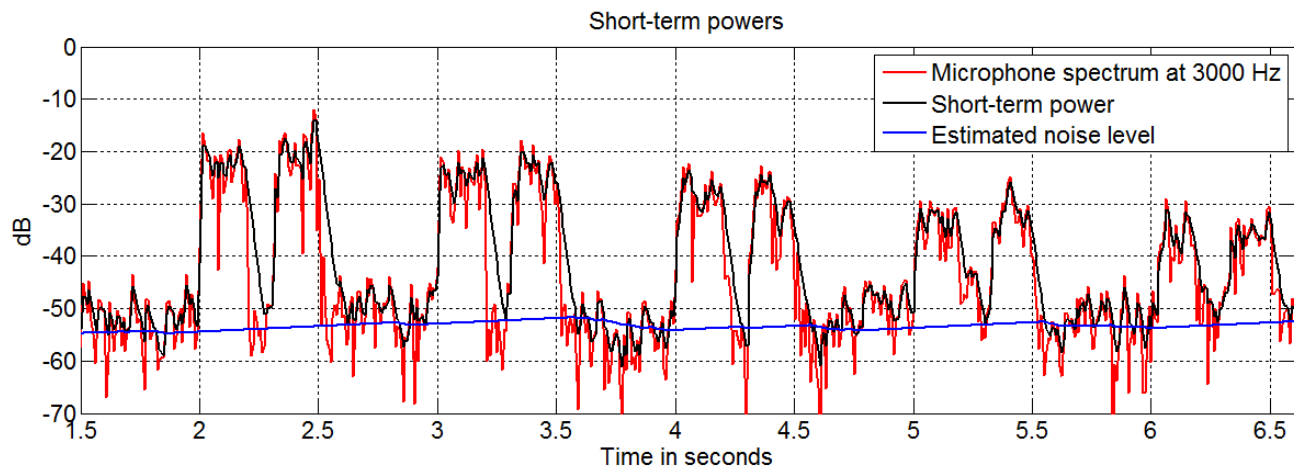
$$\hat{S}_{bb}(\Omega_{\mu}, n) \longrightarrow K_{\text{over}} \hat{S}_{bb}(\Omega_{\mu}, n)$$




the randomized opening of the filter can be avoided. This comes, however, with a more aggressive attenuation characteristic that attenuates also parts of the speech signal.

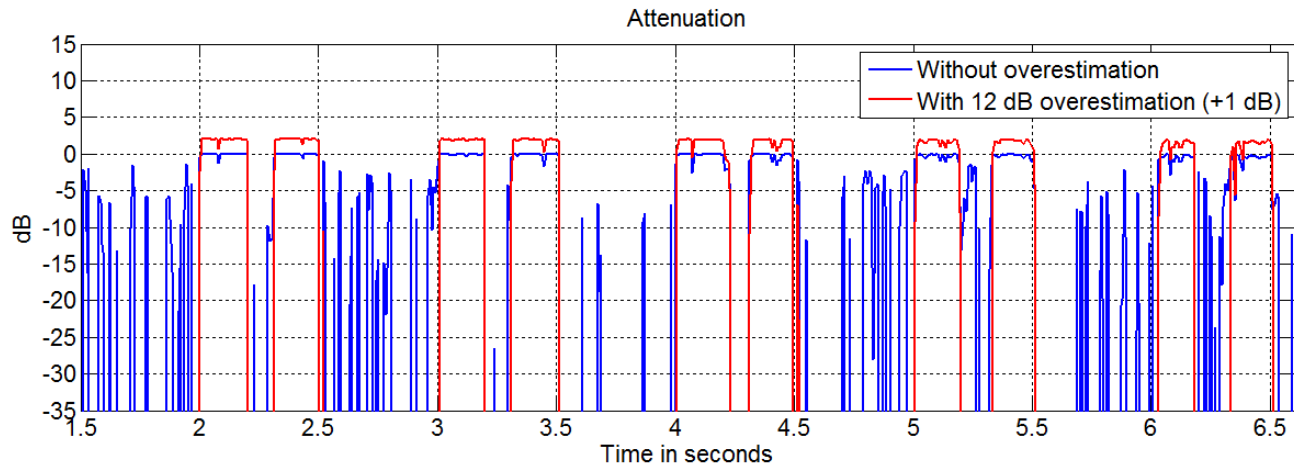
**Enhanced solutions:**

- More enhanced solutions will be presented in the lecture “Speech and Audio Processing – Audio Effects and Recognition” (offered next term by the “Digital Signal Processing and System Theory” team).

## Noise Suppression – Part 7



-  : Microphone signal
-  : Output without overestimation
-  : Output with 12 dB overestimation





## Noise Suppression – Part 8

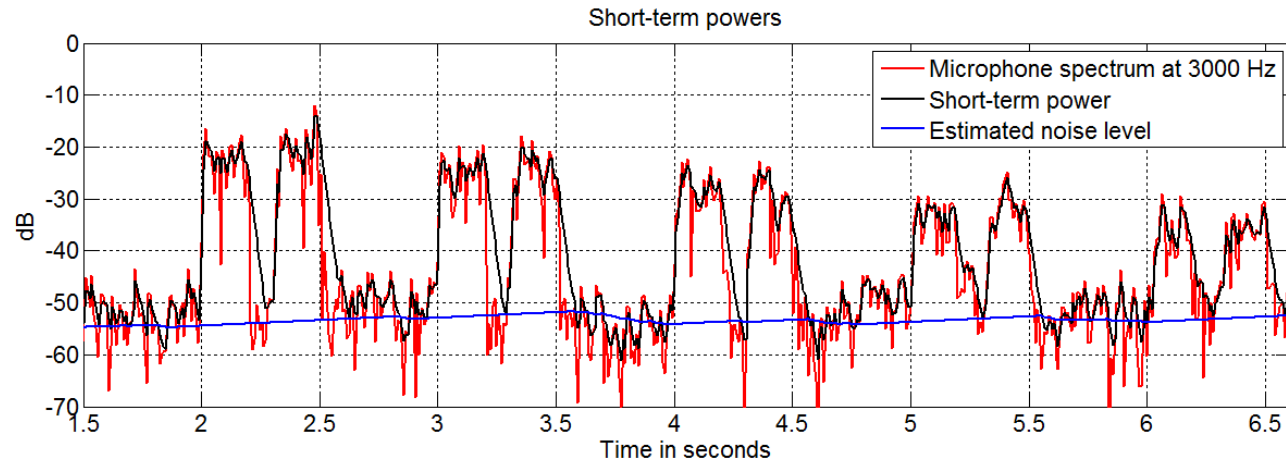
### *Limiting the maximum attenuation:*

- For several application the original shape of the noise should be preserved (the noise should only be attenuated but not completely removed). This can be achieved by inserting a maximum attenuation:


$$H_{\min}(e^{j\Omega\mu}, n) = H_{\min}.$$


- In addition, this attenuation limits can be varied slowly over time (slightly more attenuation during speech pauses, less attenuation during speech activity).

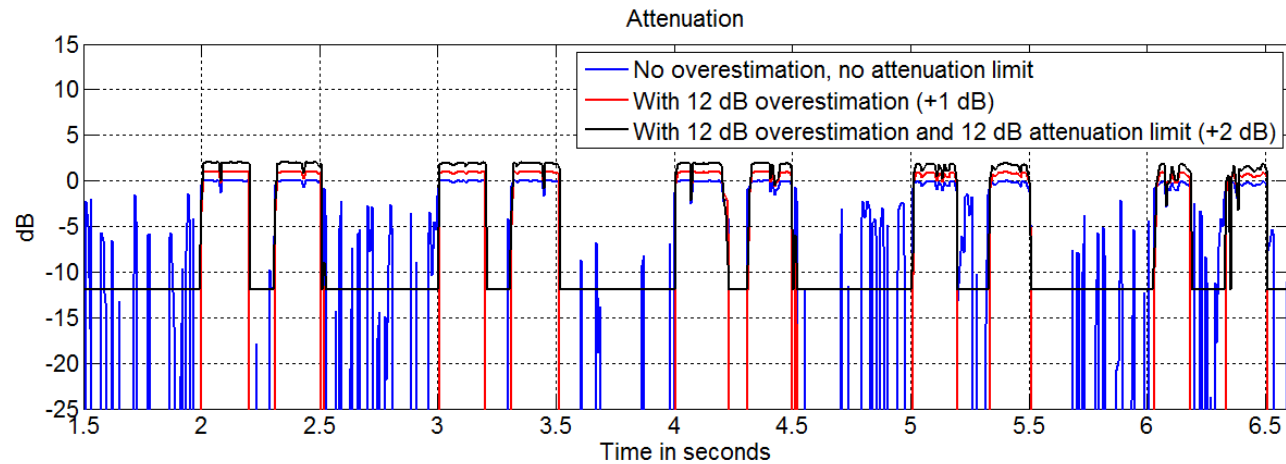
## Noise Suppression – Part 9



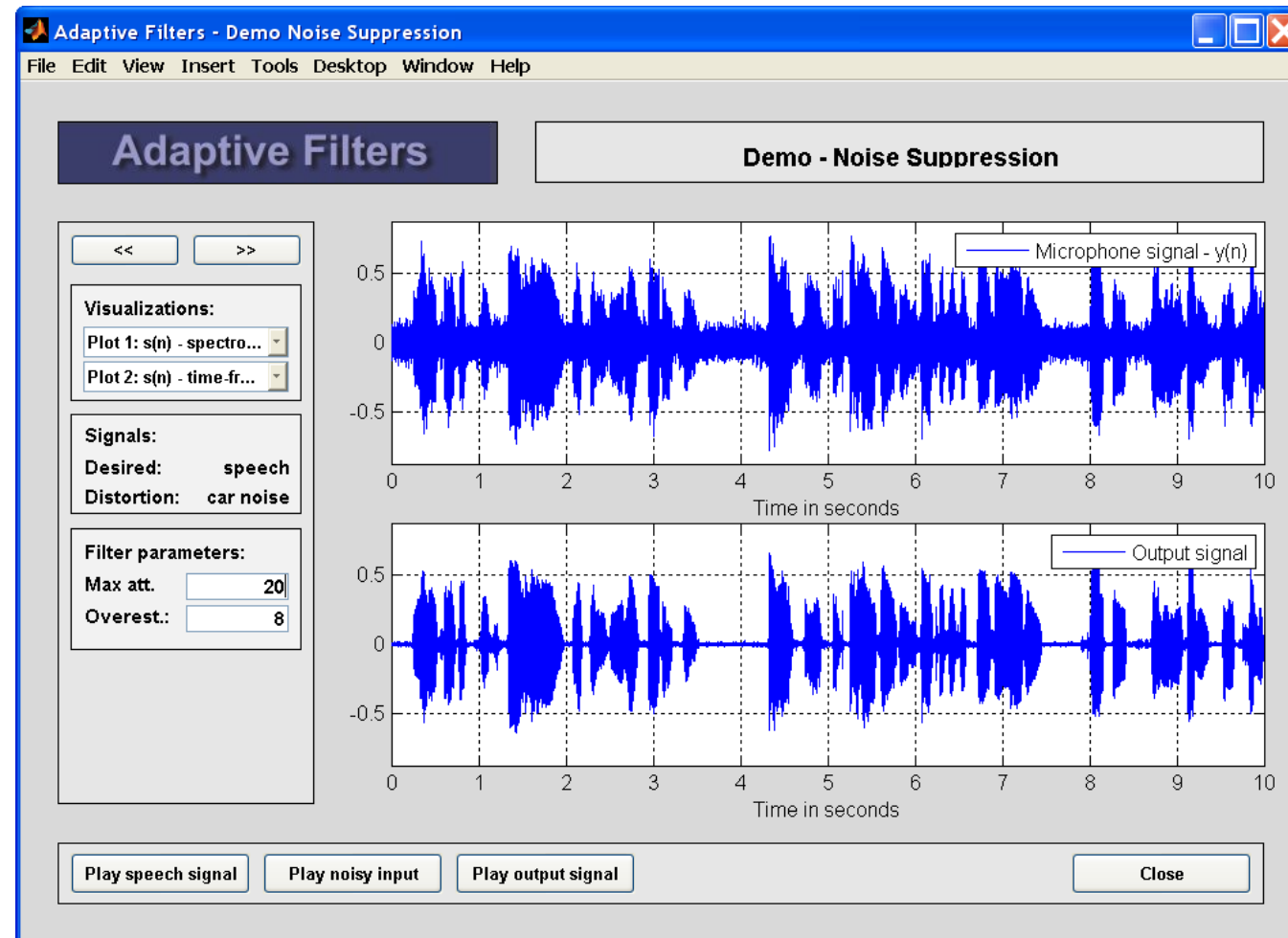
 : Microphone signal

 : Output without attenuation limit

 : Output with attenuation limit



## Noise Suppression – Part 10



## Summary and Outlook

### *This week:*

- Introduction and motivation
- Principle of orthogonality
- Time-domain solution
- Frequency-domain solution
- Application example: noise suppression

### *Next week:*

- Linear Prediction