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A Study on Combining Acoustic Echo Cancellers with Impulse Response Shortening

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Outline

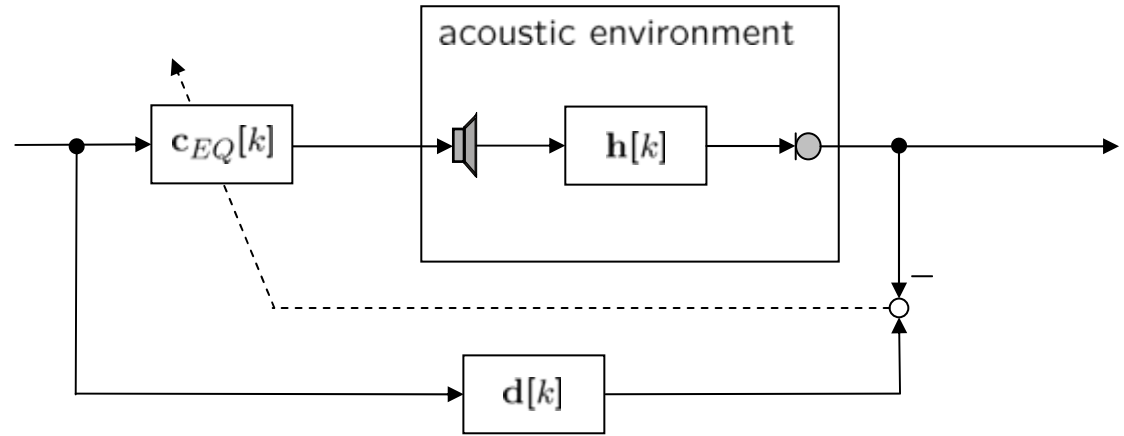
- Motivation
- Listening Room Compensation / Impulse Response Shortening
- Acoustic Echo Cancellation (AEC)
- Tail Effect for Acoustic Echo Cancellers
- Mutual influences of AEC and LRC
- Simulation Results / Conclusions

Motivation

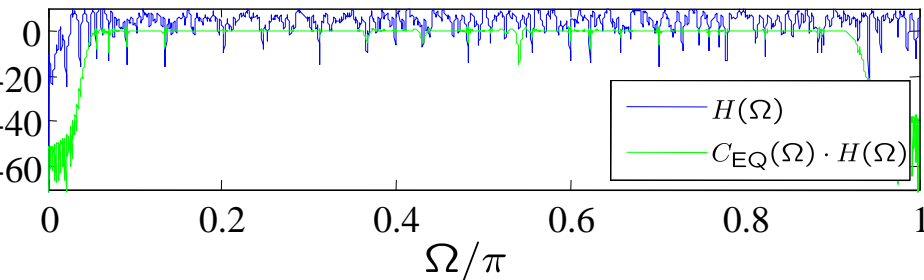
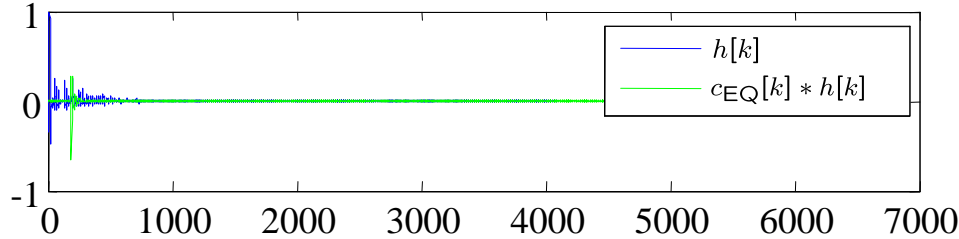
- Hands-free systems should employ subsystems for
Noise Reduction (NR)
Acoustic Echo Cancellation (AEC) and
Dereverberation / Listening Room Compensation
- Listening Room Compensation / Impulse Response Shortening can increase speech intelligibility but needs an estimate of the RIR
- Acoustic Echo Canceller have to cope with very high order impulse responses
System identification may be insufficient
- Combination of both subsystems: Inner AEC or outer AEC
- What are the mutual influences of the two subsystems?

Listening Room Compensation

- An equalizer precedes the acoustic channel
- Common design method: Least Squares Equalizer
 $c_{EQ} = H^+ d$
- Problem: Channel $h[k]$ is needed!



- The desired system $d[k]$ is approximated by the overall system of $c_{EQ}[k] * h[k]$



$$\text{var} \{H(200..3700\text{Hz})\} = 14.365$$

$$\text{var} \{C_{EQ}(200..3700\text{Hz}) \cdot H(200..3700\text{Hz})\} = 1.0379$$

Impulse Response Shortening (I)

- The goal is not spectral flatness of the overall system but a concentration of the energy at the beginning (desired area d_d).

$$d_d = \text{diag}\{w_d\} H c_{EQ}$$

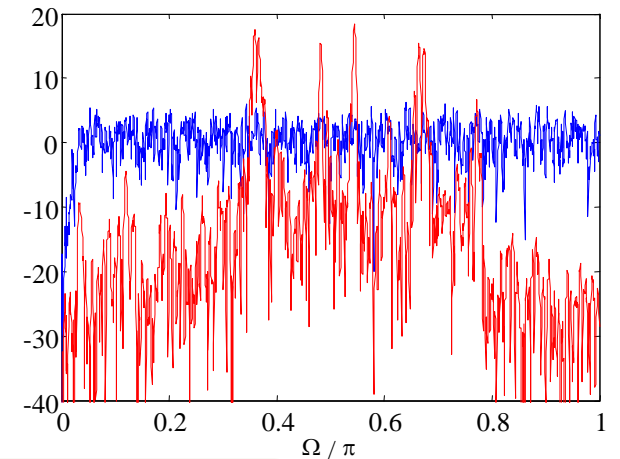
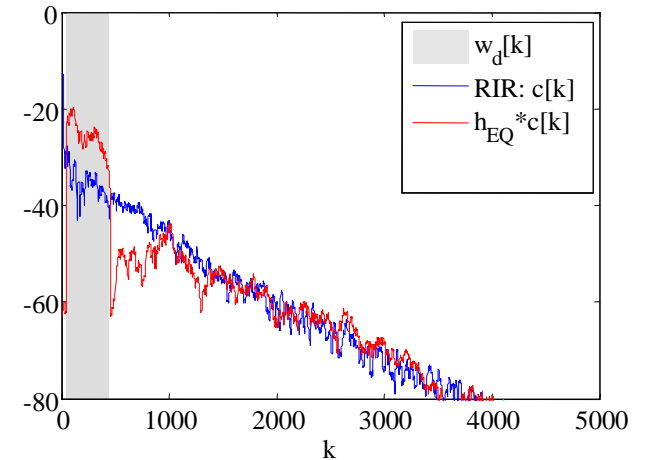
$$d_u = \text{diag}\{1 - w_d\} H c_{EQ}$$

- Maximization the energy of d_d while keeping the energy of d_u constant leads to the impulse response shortener after Melsa:

$$B_{BP} \cdot c_{EQ,opt} = A \cdot c_{EQ,opt} \cdot \lambda_{max}$$

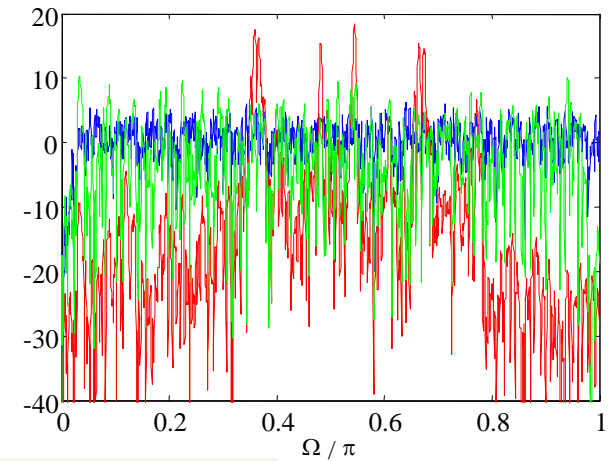
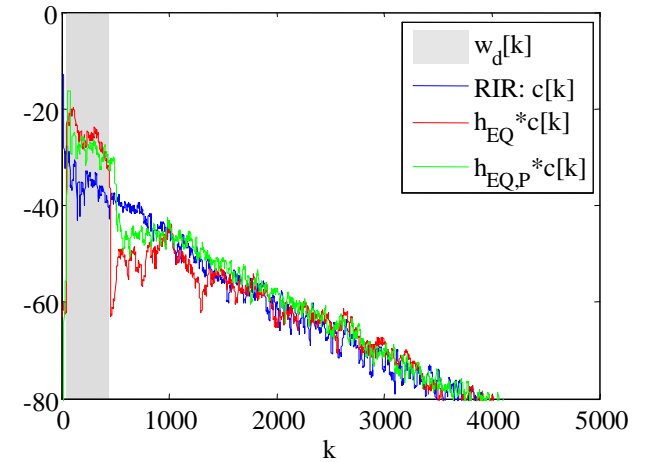
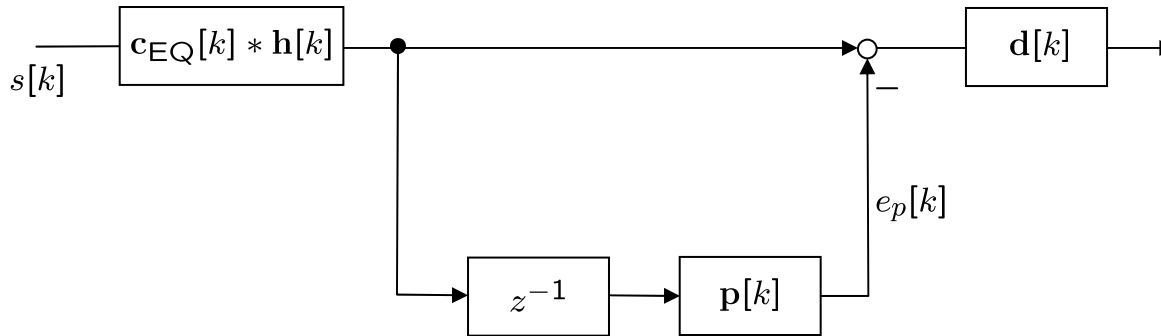
$$A = H^H \text{diag}\{w_{BP,d}\}^2 H$$

$$B_{BP} = H_{BP}^H \text{diag}\{w_{BP,d}\}^H \text{diag}\{w_{BP,d}\} H_{BP}$$



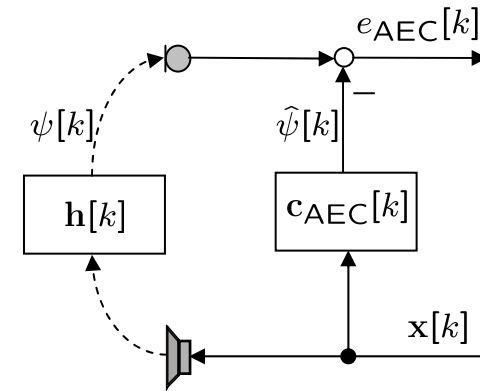
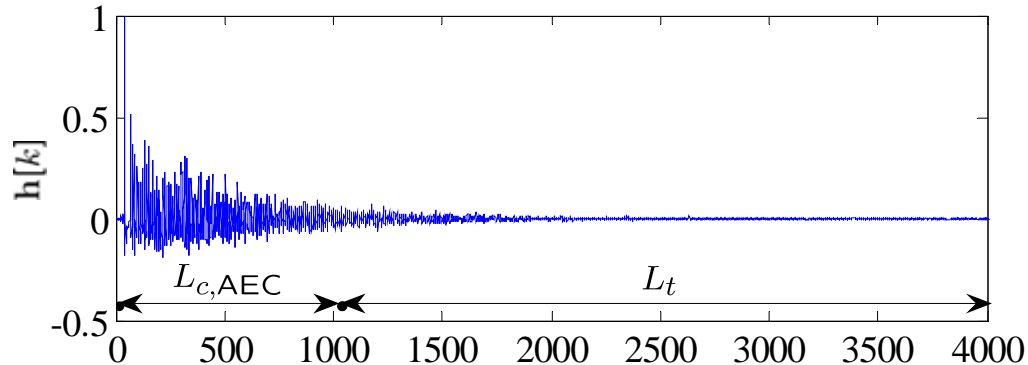
Impulse Response Shortening (II)

- Post processing by a linear prediction filter can reduce the spectral overshoots.



Tail Effect of Acoustic Echo Compensation (I)

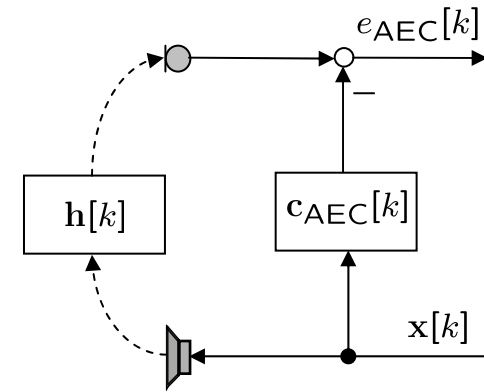
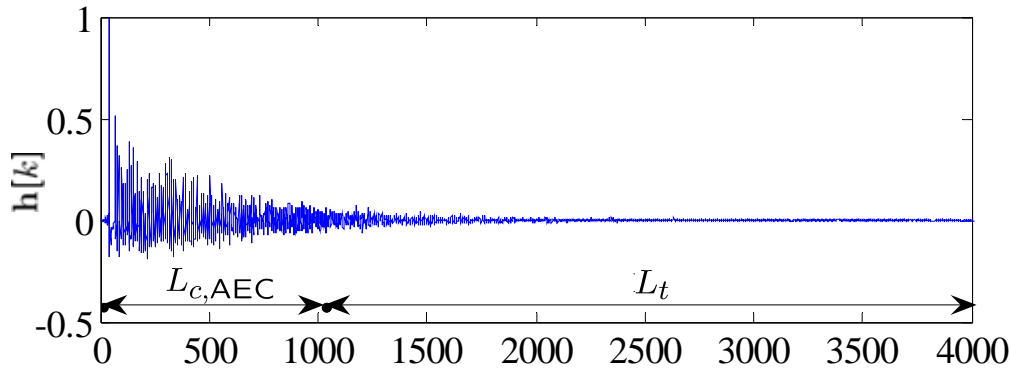
- Echo cancellation is done by system identification



- The room impulse response (RIR) $h[k]$ can be split up in two parts
- The first part ($L_{c,AEC}$ samples) can be modelled by the AEC while the *tail* (L_t samples) can not.

Tail Effect of Acoustic Echo Compensation (II)

- Echo cancellation is done by system identification



$$e_{AEC}[k] = \mathbf{h}_c^T[k] \mathbf{x}_c[k] - \mathbf{c}_{AEC}^T[k] \mathbf{x}_c[k] + \mathbf{h}_t^T[k] \mathbf{x}_t[k]$$

$$\mathbf{h}[k] = [\mathbf{h}_c^T[k], \mathbf{h}_t^T[k]]^T$$

$$\mathbf{h}_c[k] = [h_0[k], h_1[k], \dots, h_{L_{c,AEC}-1}[k]]^T$$

$$\mathbf{h}_t[k] = [h_{L_{c,AEC}}[k], h_{L_{c,AEC}+1}[k], \dots, h_{L_h-1}[k]]^T$$

$$\mathbf{x}_c[k] = [x[k], x[k-1], \dots, x[k-L_{c,AEC}+1]]^T$$

$$\mathbf{x}_t[k] = [x[k-L_{c,AEC}], \dots, x[k-L_{c,AEC}-L_t+1]]^T$$

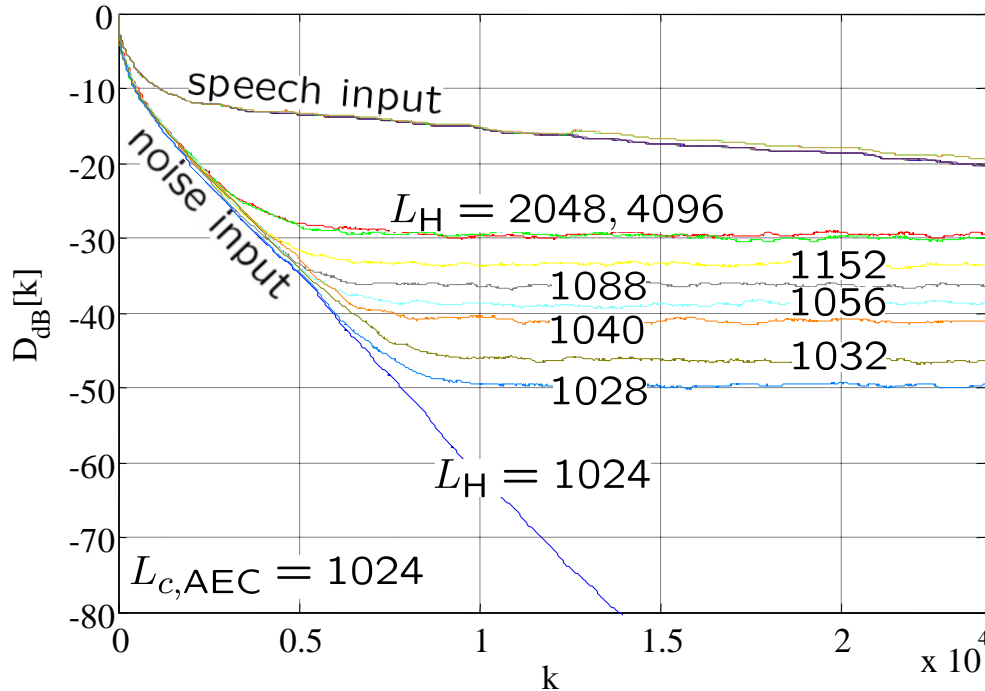
$$\mathbf{c}_{AEC}[k] = [c_{AEC,0}[k], c_{AEC,1}[k], \dots, c_{AEC,L_{c,AEC}-1}[k]]^T$$

$$\mathbf{c}_{AEC}[k] = \mathbf{h}_c[k] + \mathbf{E} \{ \mathbf{x}_c[k] \mathbf{x}_c^T[k] \}^{-1} \mathbf{E} \{ \mathbf{x}_c[k] \mathbf{x}_t^T[k] \} \mathbf{h}_t[k]$$

→ A bias is introduced for a non white input signal!

Tail Effect of Acoustic Echo Cancellation (III)

- Echo cancellation is done by system identification



- Evaluation by means of system misalignment:

$$D_{dB}[k] = \frac{\|\mathbf{h}[k] - \mathbf{c}_{AEC}[k]\|^2}{\|\mathbf{h}[k]\|^2}$$

L_H : Considered length of RIR

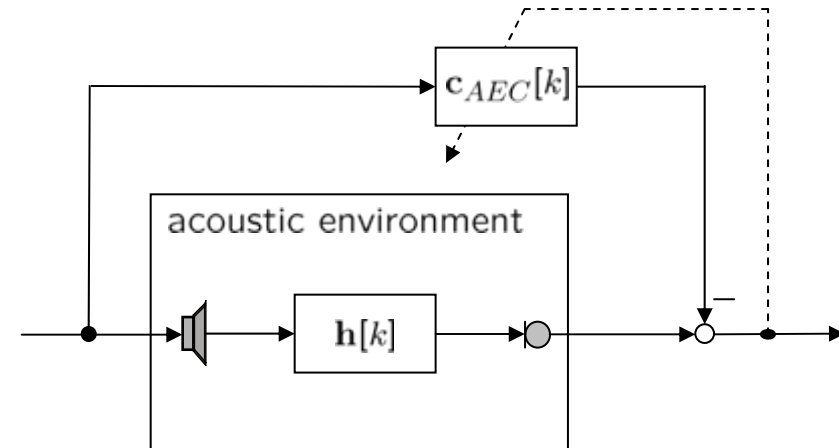
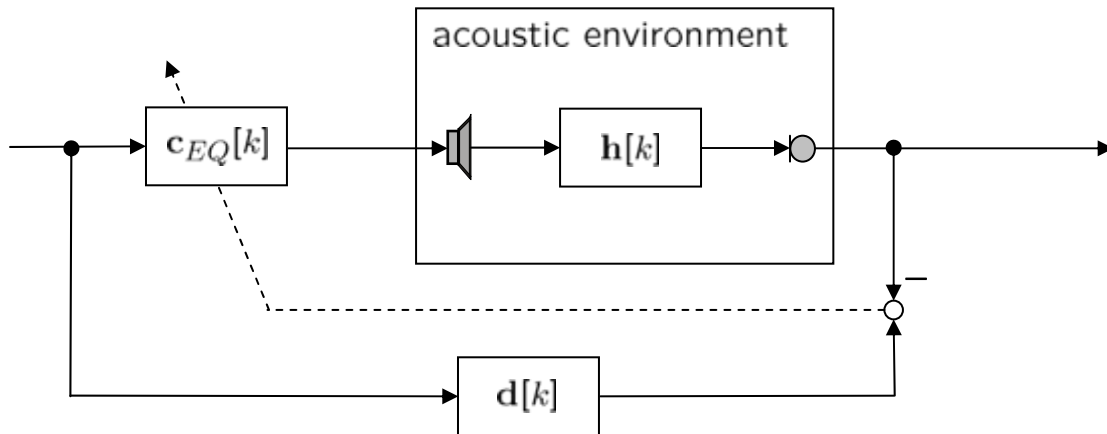
$L_{c,AEC} = 1024$: Order of AEC

$$\mathbf{c}_{AEC}[k] = \mathbf{h}_c[k] + \mathbf{E} \{ \mathbf{x}_c[k] \mathbf{x}_c^T[k] \}^{-1} \mathbf{E} \{ \mathbf{x}_c[k] \mathbf{x}_t^T[k] \} \mathbf{h}_t[k]$$

→ A bias is introduced for a non white input signal!

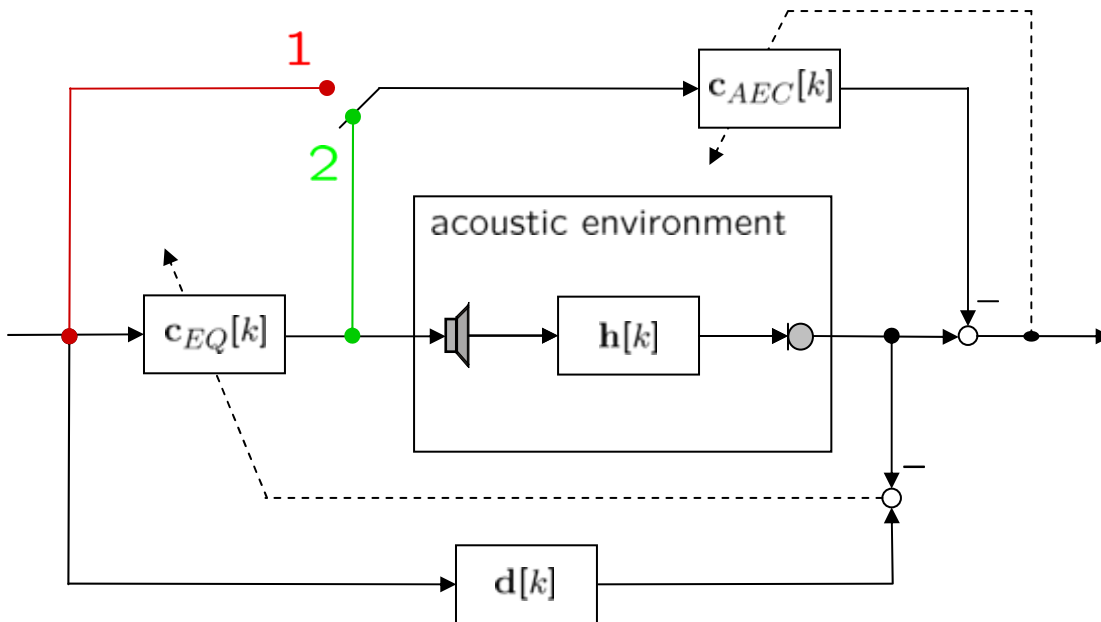
Possible Combinations of AEC and EQ

- Listening Room Compensation / Room Impulse Response Shortening
- Acoustic Echo Cancellation



Possible Combinations of AEC and EQ

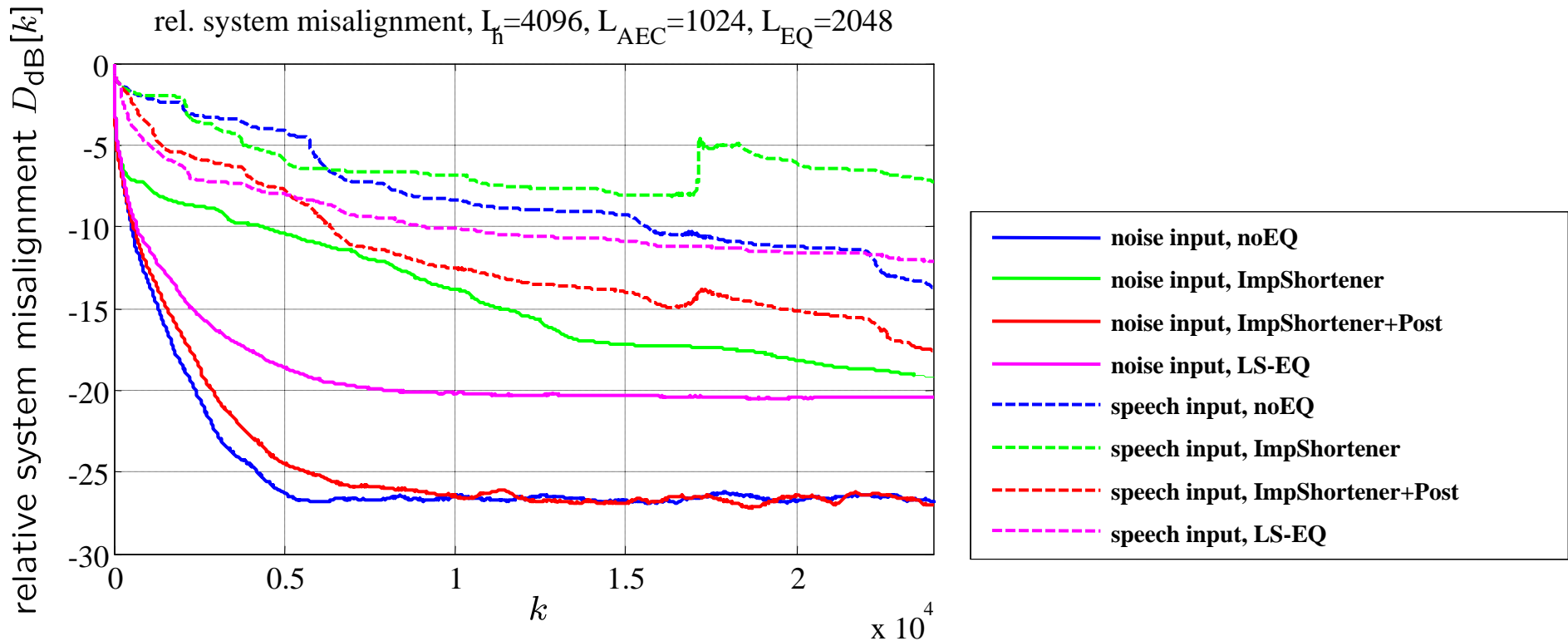
- Two Combinations are possible:



- Outer AEC**
 - AEC has to identify shorter overall system $c_{EQ}[k] * h[k]$.
 - Inner AEC**
 - AEC identifies RIR $h[k]$ which can be used for equalizer.
-
- What are the mutual influences?

Influence of the EQ on the AEC

- Evaluation by means of the system misalignment: $D_{dB}[k] = 10 \cdot \log_{10} \frac{\|\mathbf{h}[k] - \mathbf{c}_{AEC}[k]\|^2}{\|\mathbf{h}[k]\|^2}$

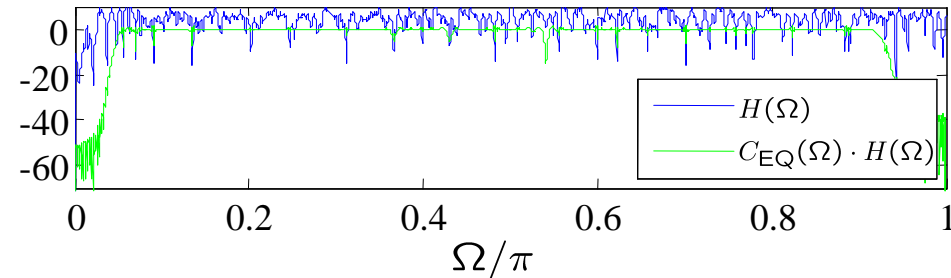
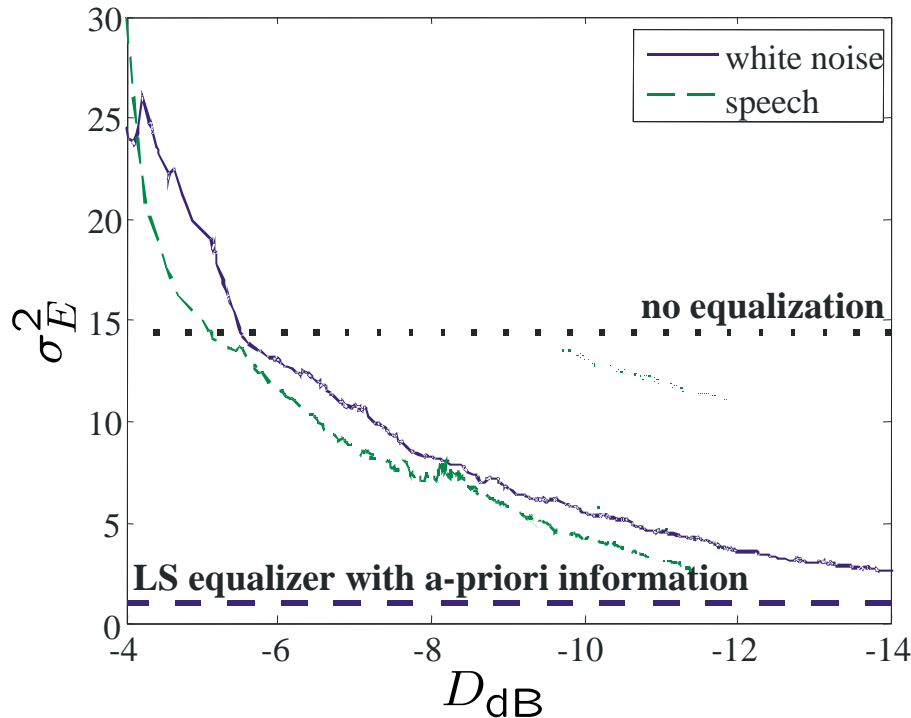


Influence of the system identification on the EQ

- Evaluation by means of the variance of the overall system:

$$\sigma_E^2 = \frac{1}{m_{max} - m_{min}} \sum_{m=m_{min}}^{m_{max}} (20 \cdot \log_{10}|E[m]| - \bar{E}_{dB})^2$$

$$\bar{E}_{dB} = \frac{1}{m_{max} - m_{min}} \sum_{m=m_{min}}^{m_{max}} 20 \cdot \log_{10}|E[m]|$$



$$\text{var} \{H(200..3700\text{Hz})\} = 14.365$$

$$\text{var} \{C_{EQ}(200..3700\text{Hz}) \cdot H(200..3700\text{Hz})\} = 1.0379$$

Conclusion

- Coloration introduced by the Equalizer in front of an AEC leads to less accurate system identification.
- The Impulse Response Shortener with post-processing leads to the best results.
- Tail effect of the AEC leads to less accurate system identification.
- Equalizer needs reliable estimate of the RIR for LRC.

Thank you for your attention!

References

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- [4] S. Haykin: Adaptive Filter Theory, Prentice Hall, 4th. Edition, 2001
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