Signalverarbeitung für offene Hörgeräte

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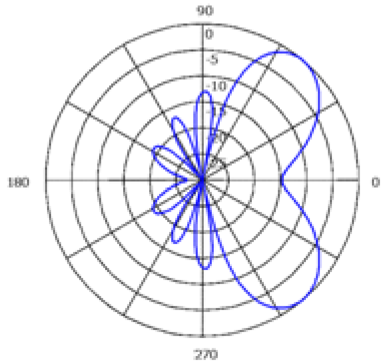
Outline

- Hearing aids: open vs. closed fittings
  - Leakage through open fitting
  - Active ear mould with internal microphone
- Noise reduction algorithms
  - Multi-channel Wiener filter (MWF)
  - Integration with active noise control: feedforward $\rightarrow$ combined feedforward-feedback
- Experimental results
  - SNR improvement and robustness
- Conclusions and future work

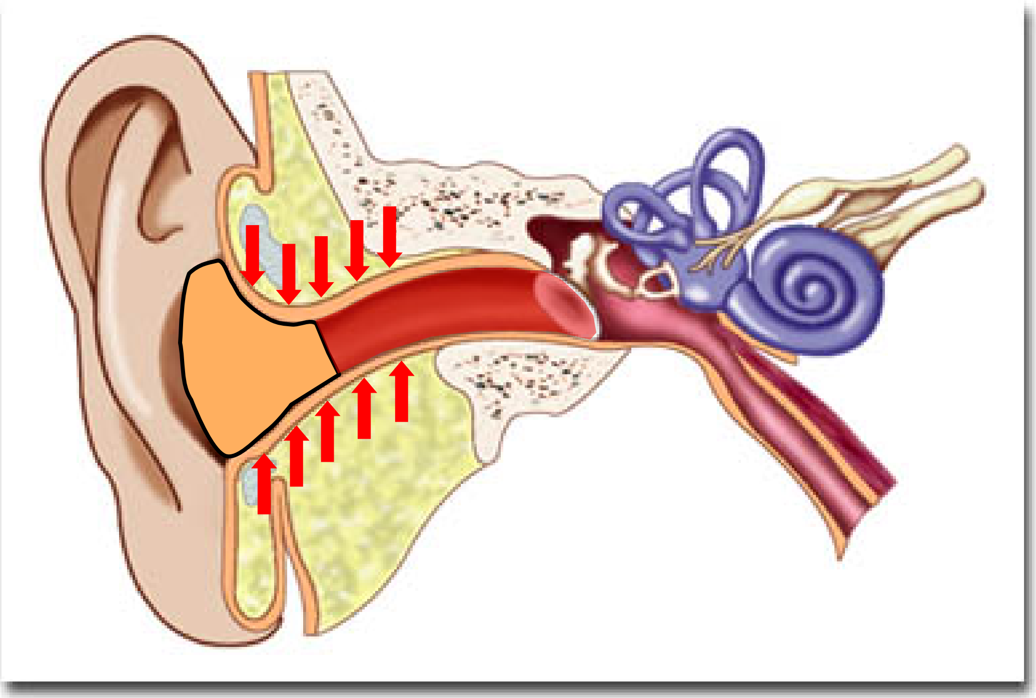
Signal processing in hearing aids

- Digital hearing aids allow for advanced acoustical signal processing:
  - multiple microphones: spectral + spatial processing
  - many hearing impaired fitted with hearing aid at both ears
- Cochlear loss:
  - Frequency-specific amplification
  - Dynamic range compression
- Binaural and central loss:
  - Noise reduction
  - Binaural Algorithms (cue preservation)
- "Technical" requirements
  - Feedback control (40-60 dB acoustic gain!)
  - Occlusion effect / 'own voice' detection
  - Classification of acoustic environment
  - (fully digital, 1V supply from very small battery, 5-6d battery time, wireless binaural link)



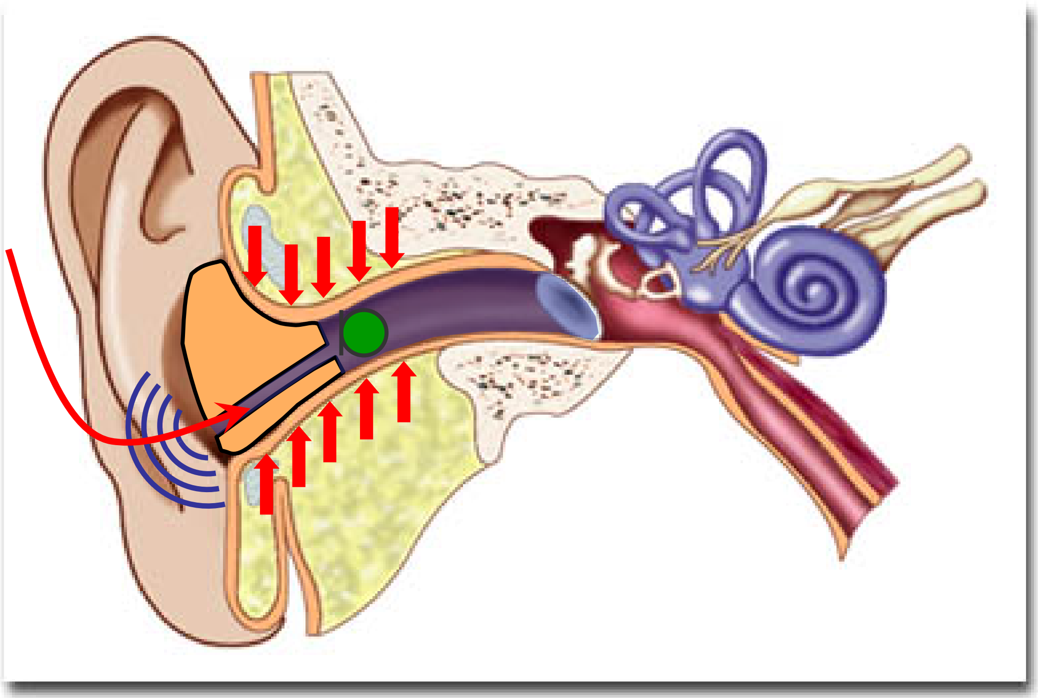
Open vs. closed fittings

- **Closed-fitting:**
  - Increase in low-frequency sound pressure when ear canal is blocked from the acoustical environment
  - Own voice is being perceived as hollow (occlusion effect)



Open vs. closed fittings

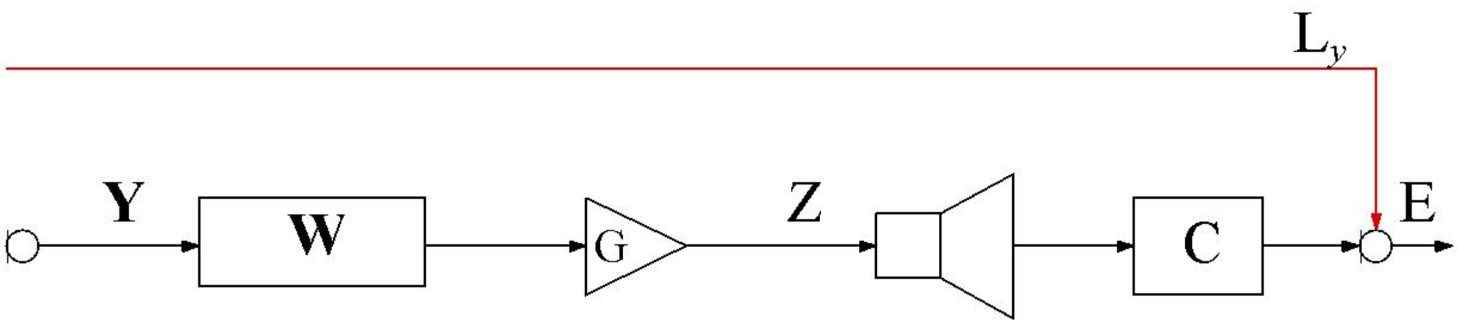
- **Open-fitting (venting):**
  - Reduces occlusion effect
  - However, undesired perceptual effects (direct + delayed sound)
  - Increased risk of feedback
  - Ambient noise leakage



→ combination of (multi-microphone) speech enhancement and active noise control using internal microphone

Noise reduction algorithms

Hearing aid configuration



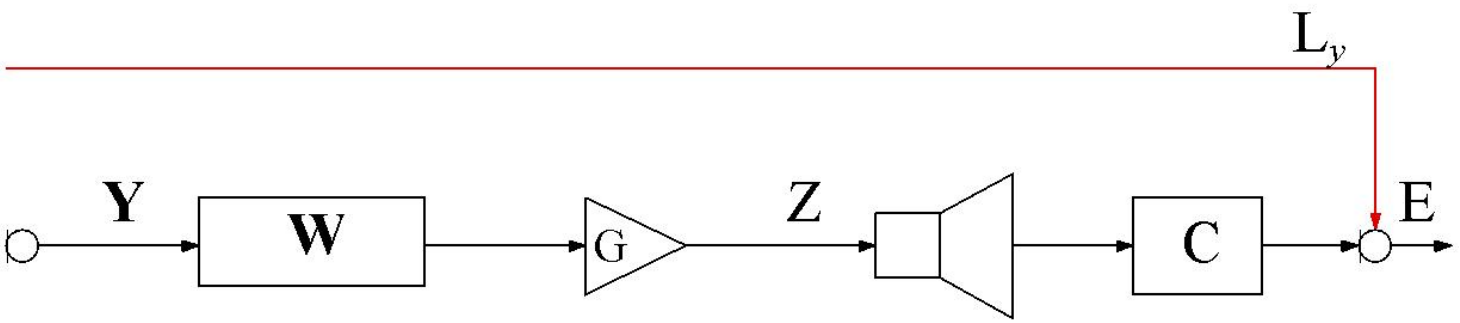
- Configuration: microphone array with *M* external microphones

$$Y_m(\omega) = X_m(\omega) + V_m(\omega), \quad m = 0 \ldots M-1$$

speech component
noise component

- Receiver (loudspeaker) signal: $\boxed{Z = G\mathbf{W}^H\mathbf{Y}}$ (G: amplification of HA)
- Error microphone signal: $\boxed{E = CZ + L_y}$ (C: secondary path)

Multi-channel Wiener filter (MWF)



- **MWF:** estimate speech component in microphone signal (usually front mic) + possible trade-off between noise reduction and speech distortion

$$D = G X_1 e^{-j\omega\Delta}$$

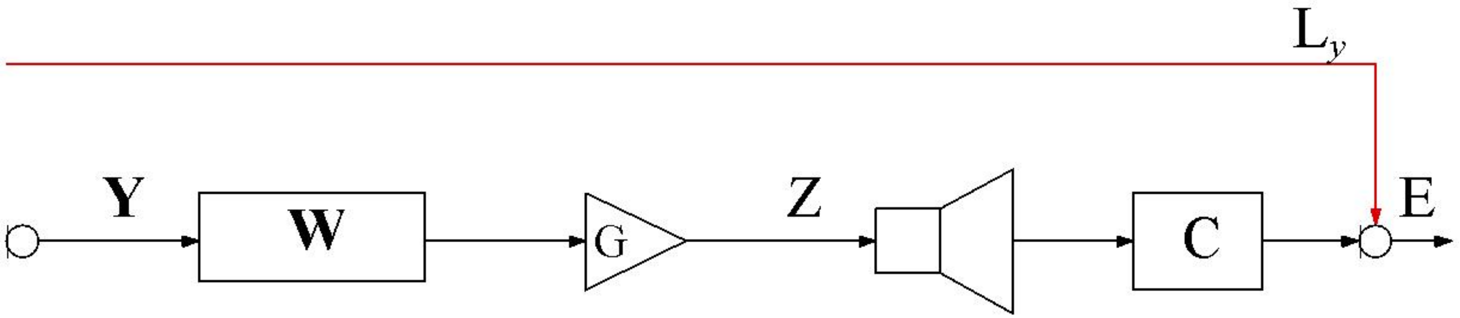
$$J_{\text{MWF}}(\mathbf{W}) = \mathcal{E}\{|Z - D|^2\}$$

$$\boxed{\mathbf{W}_{\text{MWF}} = \mathbf{R}_y^{-1}\mathbf{R}_x\mathbf{e}_{1,\Delta}}$$

- $\mathbf{e}_{1,\Delta} = [e^{+j\omega\Delta} \quad \cdots \quad 0 \quad \cdots \quad 0]^T$
- $\mathbf{R}_v = \mathcal{E}\{\mathbf{V}\mathbf{V}^H\}$: noise correlation matrix
- $\mathbf{R}_y = \mathcal{E}\{\mathbf{Y}\mathbf{Y}^H\}$: speech + noise correlation matrix
- $\mathbf{R}_x = \mathbf{R}_y - \mathbf{R}_v$: speech correlation matrix

[S. Doclo, A. Spriet, J. Wouters, M. Moonen, Speech Communication, 2007.]

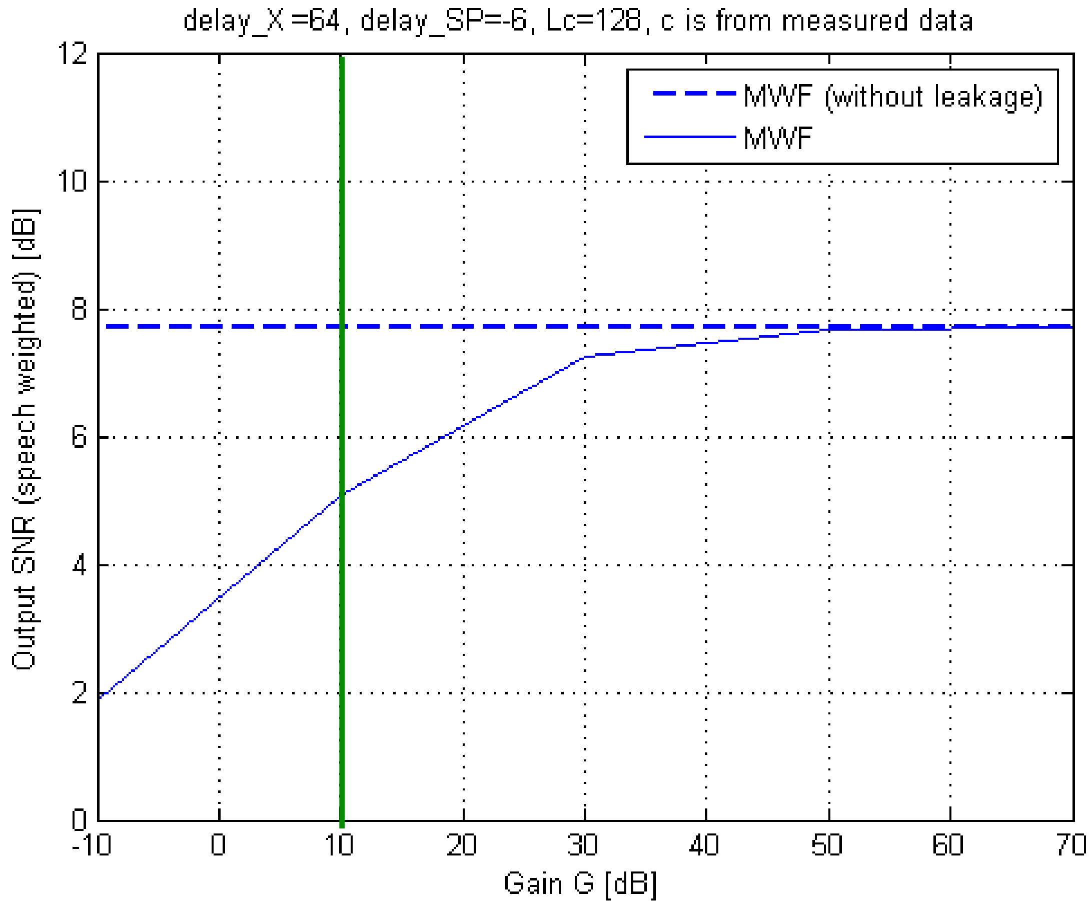
Multi-channel Wiener filter (MWF)



- **MWF:** estimate of speech component in microphone signal (usually front mic) + possible trade-off between noise reduction and speech distortion
  - Estimate $\mathbf{R}_y$ during speech-dominated time-frequency segments, estimate $\mathbf{R}_v$ during noise-dominated segments, requiring robust voice activity detection (VAD) mechanism
  - No assumptions about positions of microphones and sources
  - Different implementations:
    - Batch (off-line) vs. adaptive (update correlation matrices)
    - Using spatial prediction (SP) between speech components [Chen 2008]

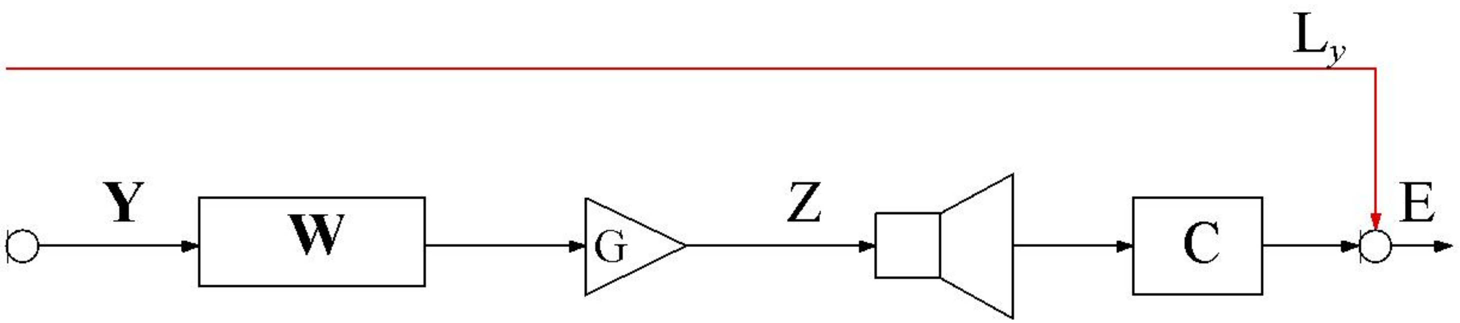
[S. Doclo, A. Spriet, J. Wouters, M. Moonen, Speech Communication, 2007.]

MWF: effect of noise leakage



Leakage degrades noise reduction performance, especially for small G

MWF + Active Noise Control (ANC)

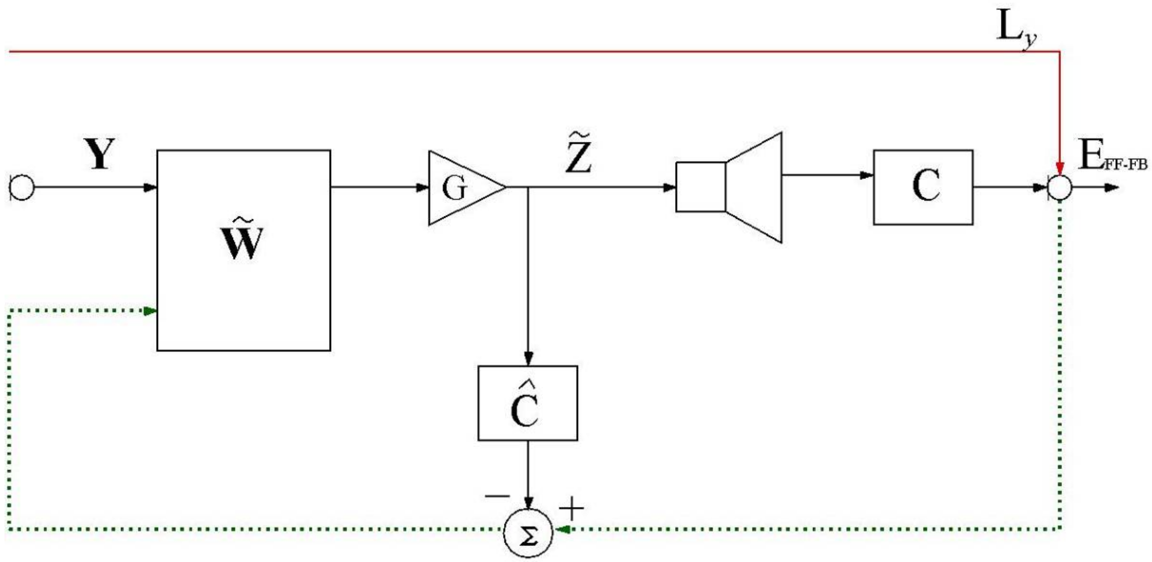


- Use external microphones + internal **error microphone**
- Difference with “standard” ANC: estimate of speech component + anti-noise
- **Feedforward (FF) configuration** [Serizel 2010]
  - Take into account leakage component

$$J_{FF}(\mathbf{W}) = \mathcal{E}\{|E - D|^2\} = \mathcal{E}\{|CZ + L_y - D|^2\}$$

$$\boxed{\mathbf{W}_{FF} = (GC^*\mathbf{R}_y)^{-1}(G\mathbf{R}_x\mathbf{e}_{1,\Delta} - \mathbf{r}_{yl_y})}$$

MWF + Active Noise Control (ANC)



$$\tilde{Z} = G\widetilde{\mathbf{W}}^H\tilde{\mathbf{Y}}$$

$$\tilde{\mathbf{Y}} = \begin{bmatrix} \mathbf{Y} \\ L_y \end{bmatrix}$$

$$E_{\text{FF-FB}} = C\tilde{Z} + L_y$$

- **Combined Feedforward-Feedback (FF-FB) configuration**
  - Leakage component in error microphone is used as **additional** input
  - Can be estimated if (estimate of) secondary path C is available

$$J_{\text{FF-FB}}(\widetilde{\mathbf{W}}) = \mathcal{E}\{|E_{\text{FF-FB}} - D|^2\} = \mathcal{E}\{|GC\mathbf{W}^H \begin{bmatrix} \mathbf{Y} \\ L_y \end{bmatrix} + L_y - D|^2\}$$

$$\boxed{\widetilde{\mathbf{W}}_{\text{FF-FB}} = (GC^*\tilde{\mathbf{R}}_y)^{-1}(G\tilde{\mathbf{R}}_x\mathbf{e}_{1,\Delta} - \tilde{\mathbf{r}}_{yl_y})}$$

Comparison of the algorithms

- MWF:

$$J_{\text{MWF}}(\mathbf{W}) = \mathcal{E}\{|G\mathbf{W}^H\mathbf{Y} - D|^2\}$$

$$\boxed{\mathbf{W}_{\text{MWF}} = \mathbf{R}_y^{-1}\mathbf{R}_x\mathbf{e}_{1,\Delta}}$$

  - leakage signal is not taken into account

- FF ANC:

$$J_{\text{FF}}(\mathbf{W}) = \mathcal{E}\{|CG\mathbf{W}^H\mathbf{Y} + L_y - D|^2\}$$

$$\boxed{\mathbf{W}_{\text{FF}} = (GC^*\mathbf{R}_y)^{-1}(G\mathbf{R}_x\mathbf{e}_{1,\Delta} - \mathbf{r}_{yl_y})}$$

  - leakage signal is taken into account
  - leakage signal is not filtered

- FF-FB ANC:

$$J_{\text{FF-FB}}(\widetilde{\mathbf{W}}) = \mathcal{E}\{|CG\mathbf{W}^H \begin{bmatrix} \mathbf{Y} \\ L_y \end{bmatrix} + L_y - D|^2\}$$

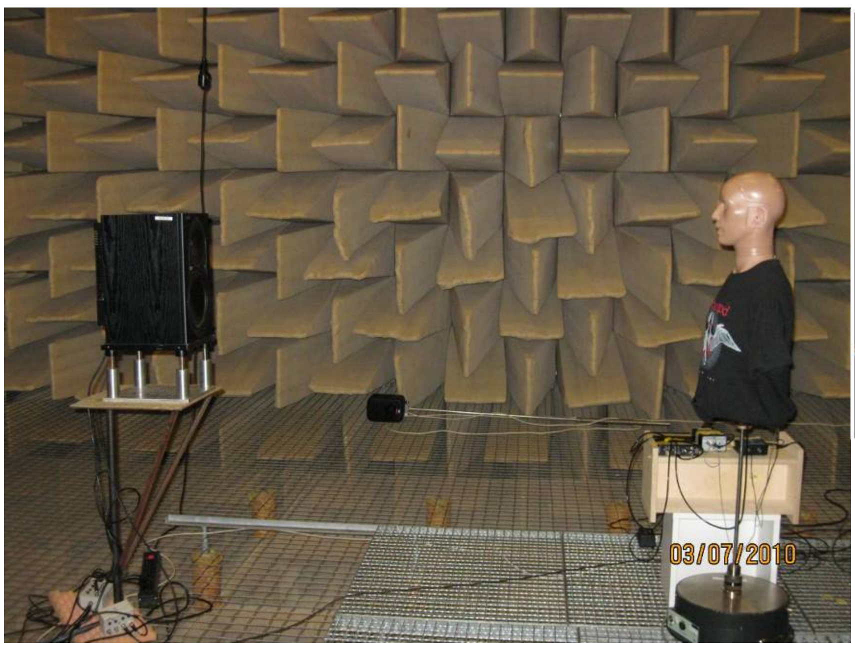
$$\boxed{\widetilde{\mathbf{W}}_{\text{FF-FB}} = (GC^*\widetilde{\mathbf{R}}_y)^{-1}(G\widetilde{\mathbf{R}}_x\mathbf{e}_{1,\Delta} - \widetilde{\mathbf{r}}_{yl_y})}$$

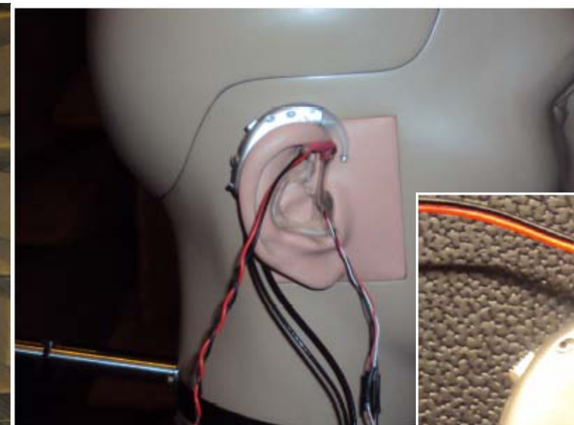
  - leakage component of error microphone is used as an additional input
  - leakage component of error microphone is filtered

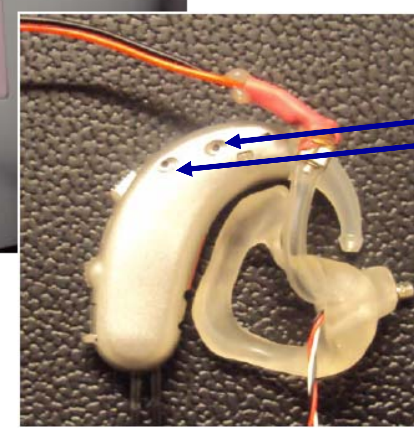
Experimental results

Recordings

- Anechoic room recordings with KEMAR HATS
  - Sound sources @ 3m from HATS, every 5° angle
- BTE hearing aid + active ear mould (vent size = 2mm):
  - 2 external microphones
  - external receiver (Knowles, TWFK-30017-000)
  - internal microphone (Knowles, FG-23329-PO7) + KEMAR microphone







Recordings

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- Used signals:
  - Speech source: HINT, angle = 0°
  - Noise source: babble noise, angles = 90°, 180°, 270°
  - f_s = 16 kHz
- Simulation parameters:
  - Secondary path *C* estimated and known (L_c= 128)
  - MWF: L = 128, Δ = 64

Performance Analysis

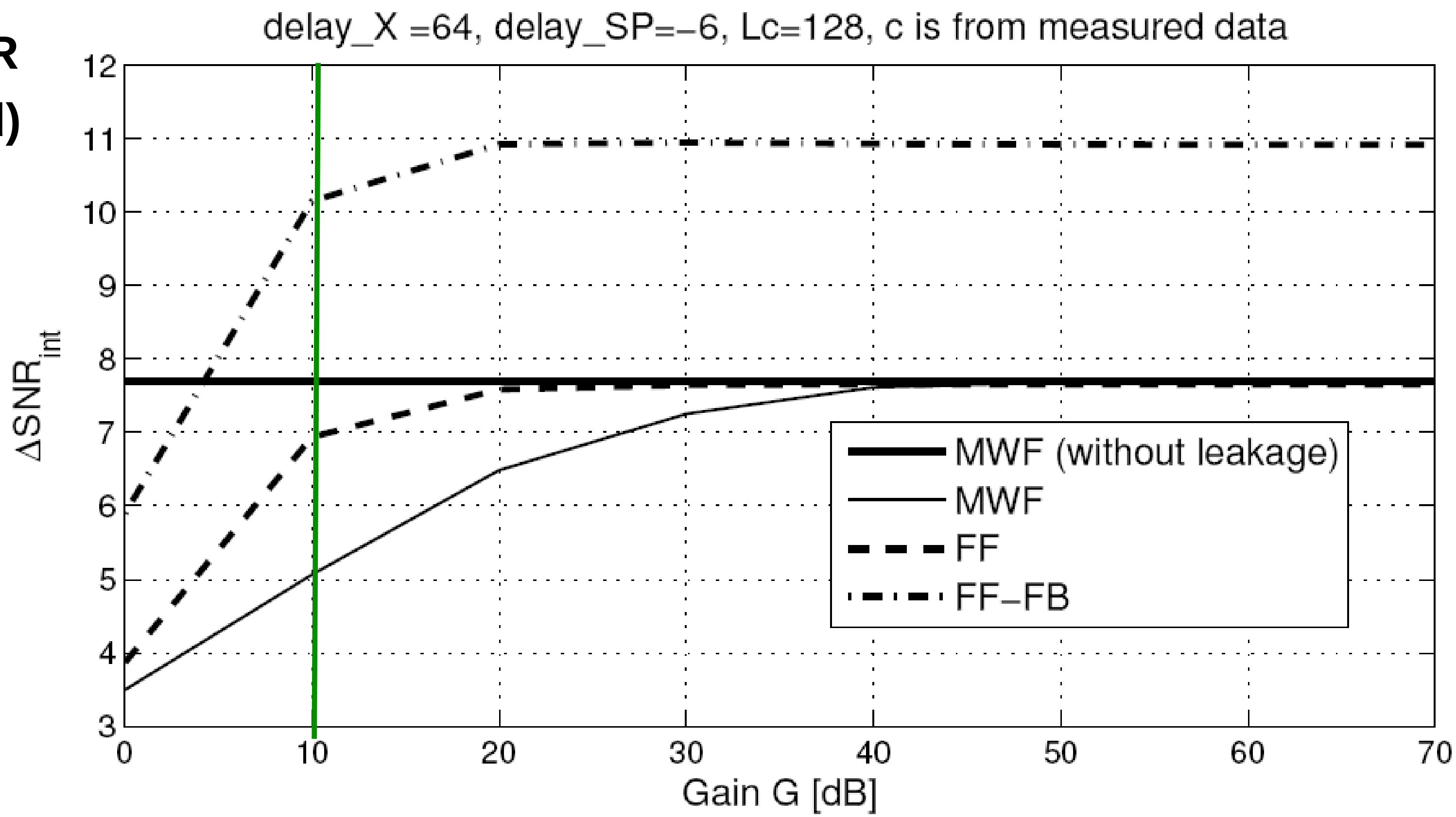
- Performance measures:
  - Frequency-dependent SNR improvement: $\Delta\mathrm{SNR}_j = 10\log_{10}\frac{P_{j,e_x}}{P_{j,e_v}} - 10\log_{10}\frac{P_{j,x_1}}{P_{j,v_1}}$
  - Speech-intelligibility-weighted broadband SNR improvement

$$\Delta\mathrm{SNR}_{int} = \sum_{j=1}^{J} I_j \Delta\mathrm{SNR}_j$$

- SNR improvement for:
  - Different amplifications *G* (0-70dB) $\rightarrow$ *different noise leakage power*
  - Different algorithms (MWF, FF, FF-FB)
- Three cases:
  - *case ER-ER:* both filters and performance are computed at error microphone
  - *case KE-KE:* both filters and performance are computed at KEMAR microphone
  - *case ER-KE:* filters computed at error microphone, performance at KEMAR microphone $\rightarrow$ investigate **robustness**

Experimental results (1)

Case ER-ER (broadband)

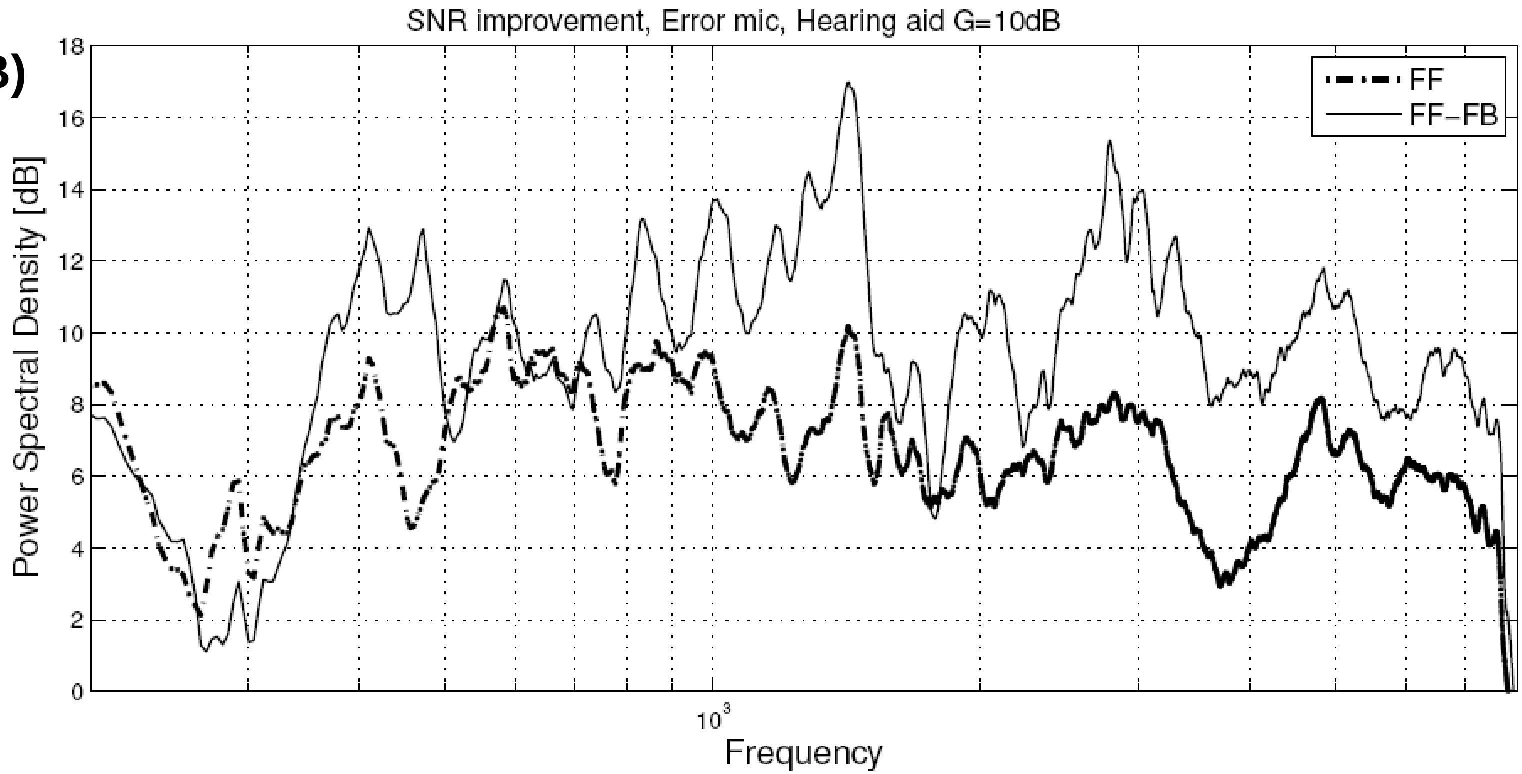


Combined FF-FB ANC algorithm outperforms FF ANC and standard MWF algorithm

Experimental results (2)

Case ER-ER

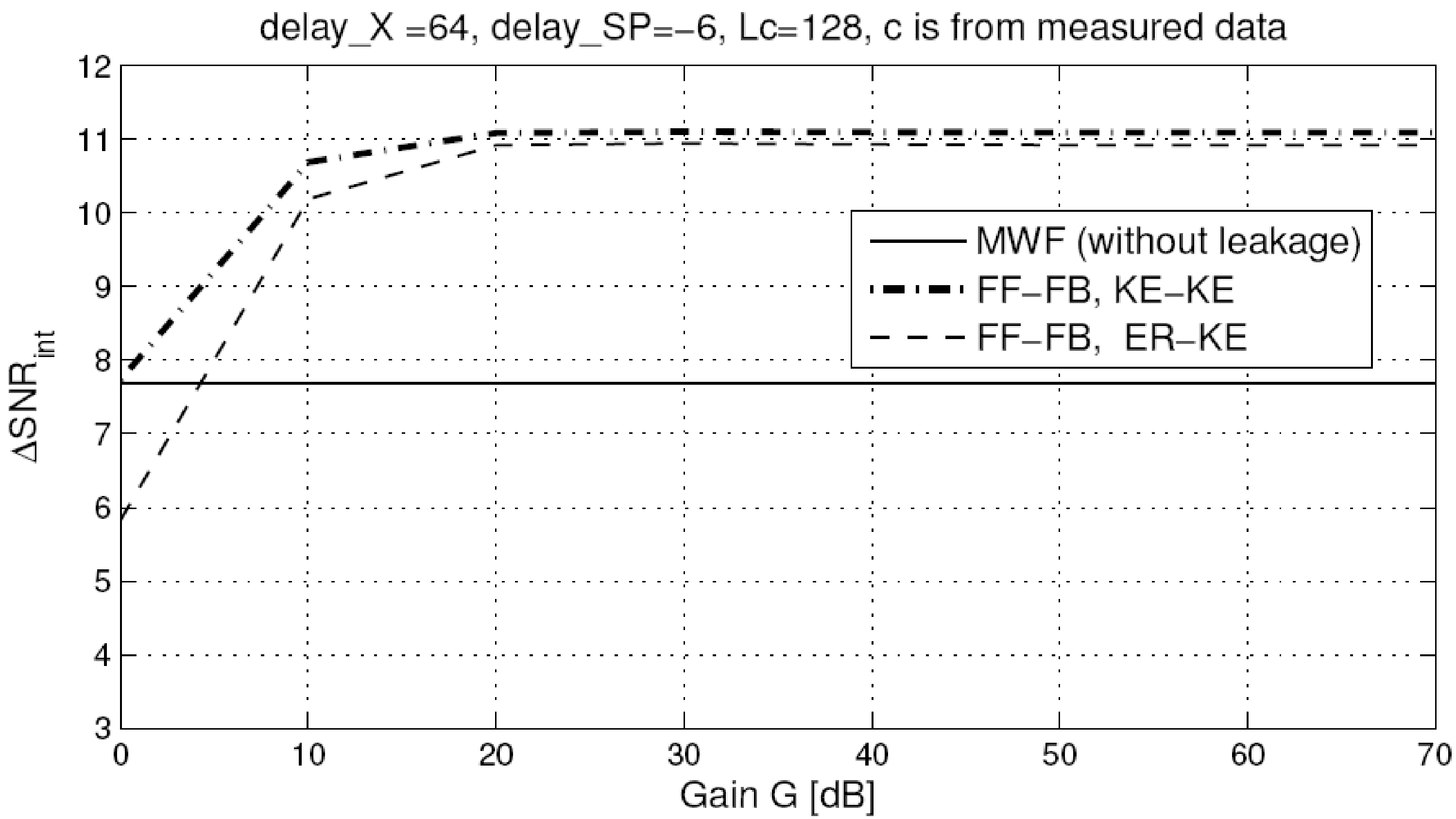
(freq, G=10dB)



Combined FF-FB ANC algorithm outperforms FF ANC and standard MWF algorithm

Experimental results (3)

Robustness (broadband)

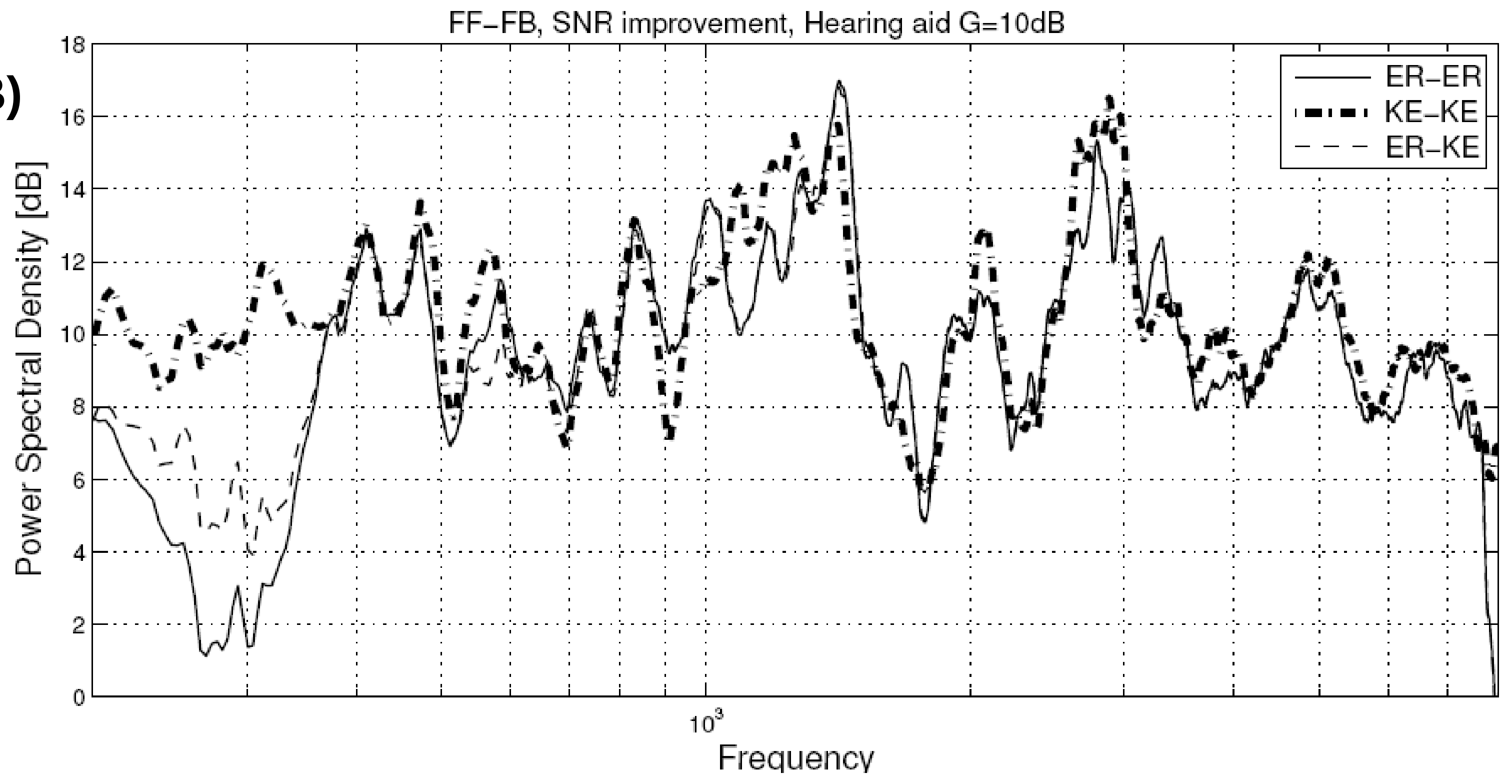


Performance at KEMAR microphone is hardly degraded when Using filters computed at error microphone→ robustness

Experimental results (4)

Robustness

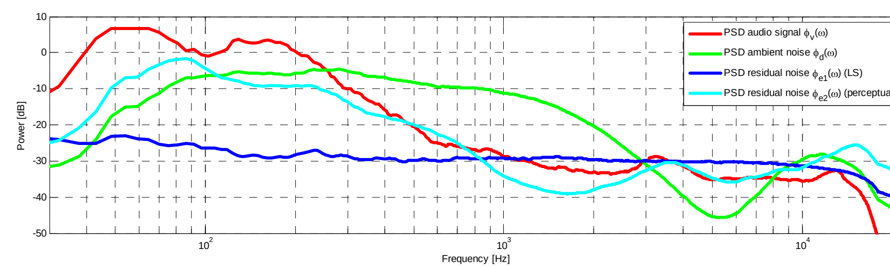
(freq, G=10dB)

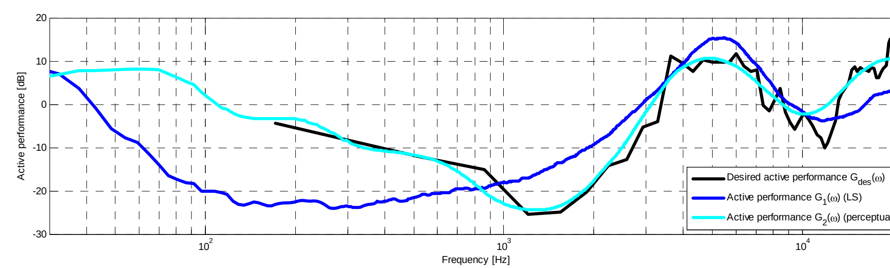


Main difference in lower frequencies (<400 Hz), to be further investigated

Future work

- Adaptive algorithms (e.g. estimate of secondary path)
- Combination with feedback suppression
- Integration of ear canal models and psycho-acoustic hearing properties in ANC filter optimisation
  - → Use estimate of the sound pressure at the ear drum
- Real-time implementation (low-latency, speedgoat) and subjective validation





Conclusions

- Open fittings: no occlusion effect, but leakage degrades noise reduction performance, especially for small gains
- Use of active ear mould with internal microphone:
  - **FF ANC**: leakage is taken into account
  - **FF-FB ANC**: leakage is used as additional input
- Combined FF-FB ANC algorithm outperforms FF ANC and standard MWF algorithm for noise reduction
- Performance computed at KEMAR microphone is hardly degraded, showing the robustness of the proposed approach.

Questions ?

House of Hearing, Oldenburg