

# Acoustically Transparent Earpiece: Equalization, Feedback cancellation, Active noise control and Own voice pickup

**Prof. Dr. Simon Doclo**

*Dept. of Medical Physics and Acoustics and Cluster of Excellence  
Hearing4all, University of Oldenburg, Germany*

Webinar Danish Sound Cluster, May 9 2023

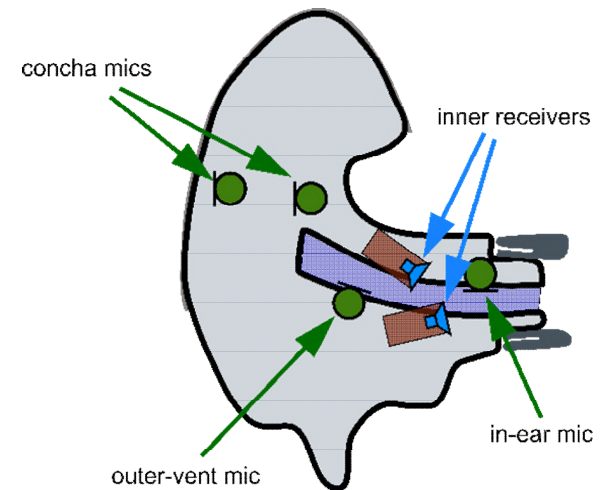
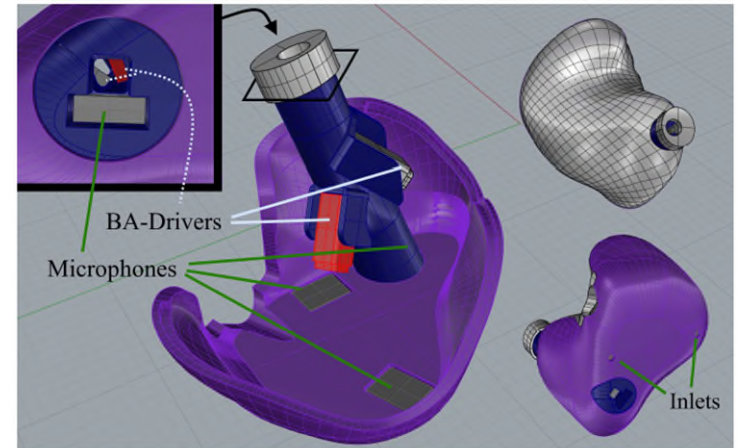
# Acoustically Transparent Earpiece

- **Long-term objective:**
  - Acoustically transparent speech communication using earpiece with multiple microphones and receivers/loudspeakers
- Develop, implement, and evaluate **individualized algorithms** for
  1. sound pressure **equalization (transparency)**
  2. acoustic **feedback cancellation**
  3. **active noise/occlusion control**
  4. **own voice extraction**



# Acoustically Transparent Earpiece (Hearpiece)

- One-size-fits-all design: fits about 90% of human ears
- Vent: **2 microphones, 2 receivers**
- Concha: **2 microphones**
- Two versions: vented + closed
- **Available at Hörzentrum / InEar GmbH:**  
<https://www.hz-ol.de/en/hearpiece.html>



[Denk et al., AES Conference  
Headphone Technology, 2019]

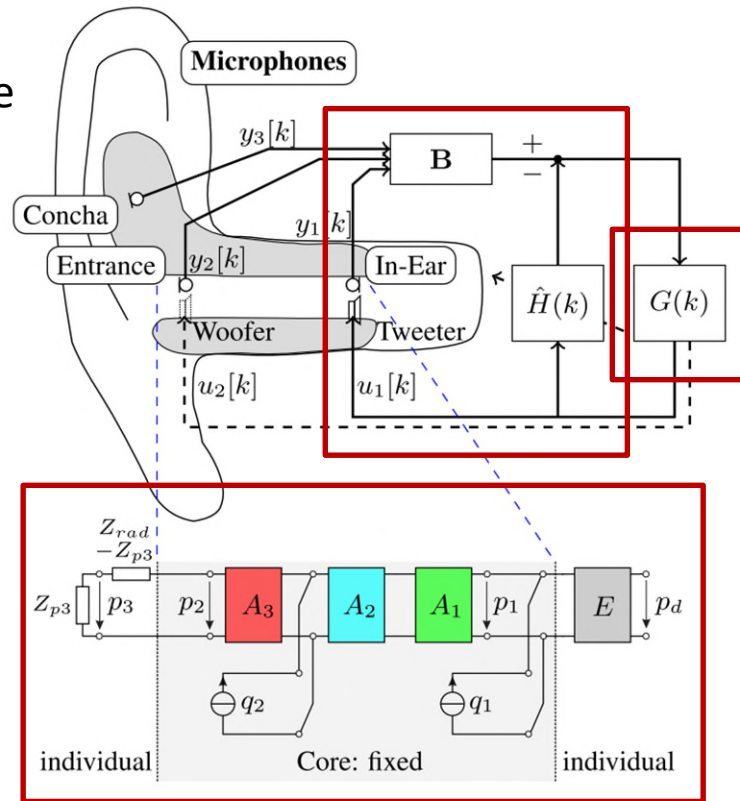
# Acoustically Transparent Earpiece

## 1. Acoustically transparent sound presentation:

- Enable hearing comparable to open ear (equalization using single/multiple receivers)

## 2. Individualized Electro-Acoustic Model:

- Better understand acoustics
- Predict sound pressure and transfer functions (eardrum)



## 3. Acoustic Feedback cancellation

- Exploit multiple microphones to steer null towards position of receiver
- Exploit multiple receivers

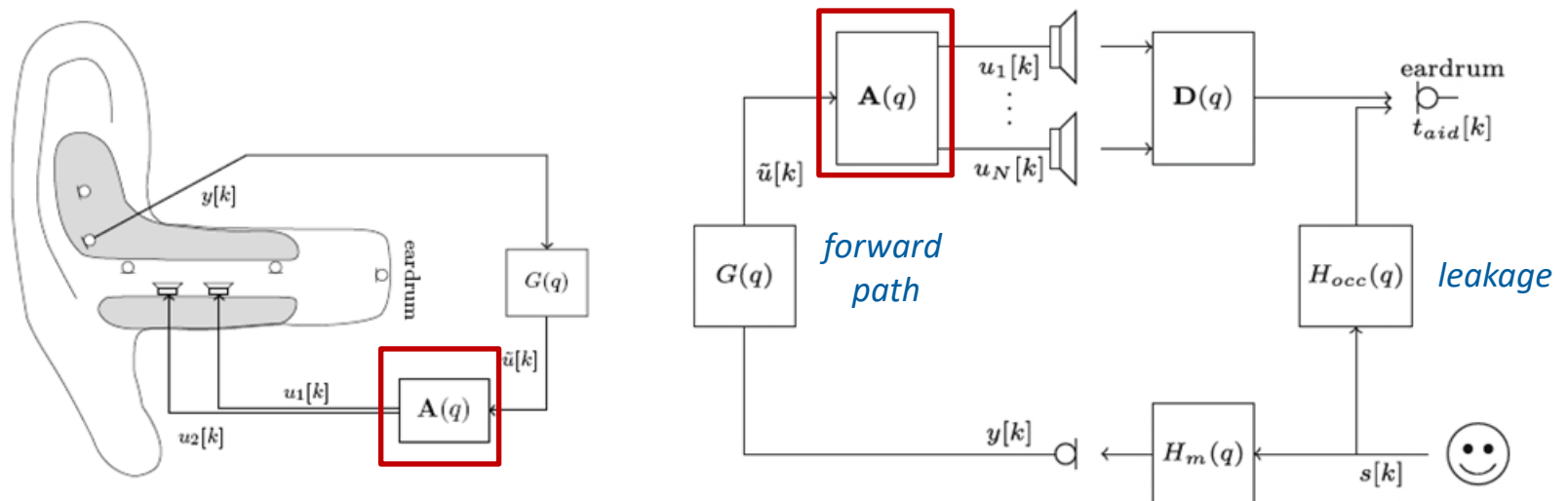
## 4. Hearing support:

- Amplification and dynamic range compression
- Active noise and occlusion control

# 1. Acoustically transparent sound presentation

## • Single/Multi-Loudspeaker Equalization

- **Goal:** Achieve sound pressure at aided ear that is (physically or perceptually) equivalent to sound pressure at open ear
- **Design and apply equalization filter(s)  $A(q)$**  to concha microphone signal, taking into account leakage and hearing device processing (forward path)



$$G(q)H_m(q)D^T(q)A(q) = G(q)H_{open}(q) - H_{occ}(q)$$

# 1. Acoustically transparent sound presentation

- **Single/Multi-Loudspeaker Equalization**

- **Goal:** Achieve sound pressure at aided ear that is (physically or perceptually) equivalent to sound pressure at open ear
- **Design and apply equalization filter(s)  $A(q)$**  to concha microphone signal, taking into account leakage and hearing device processing (forward path)
- **Robust least-squares-based design procedure** (with group delay compensation and frequency-dependent regularization)
- **Requires** (one or multiple) measurements of all transfer functions

*multiple measurements*                      *regularization*

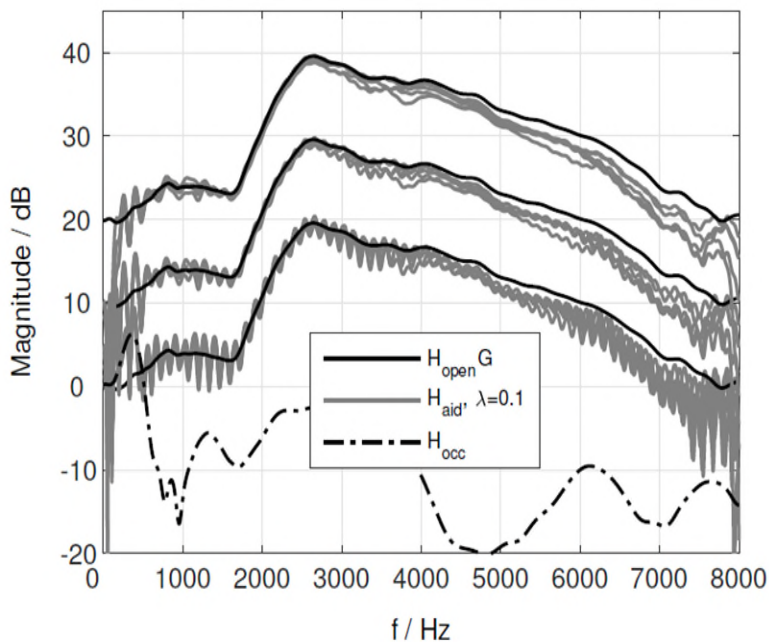
$$J_{mfr\Delta LS}(\mathbf{a}) = \sum_{i=1}^I \|\mathbf{D}_{\Delta,i}\mathbf{a} - \tilde{\mathbf{v}}_{\Delta,i}\|_2^2 + \lambda \|\mathbf{W}\mathbf{F}\mathbf{a}\|_2^2$$

$$\mathbf{a}_{mfr\Delta LS} = \left( \bar{\mathbf{D}}_{\Delta}^T \bar{\mathbf{D}}_{\Delta} + \lambda \mathbf{F}^H \mathbf{W}^H \mathbf{W} \mathbf{F} \right)^{-1} \bar{\mathbf{D}}_{\Delta}^T \bar{\mathbf{v}}_{\Delta}$$

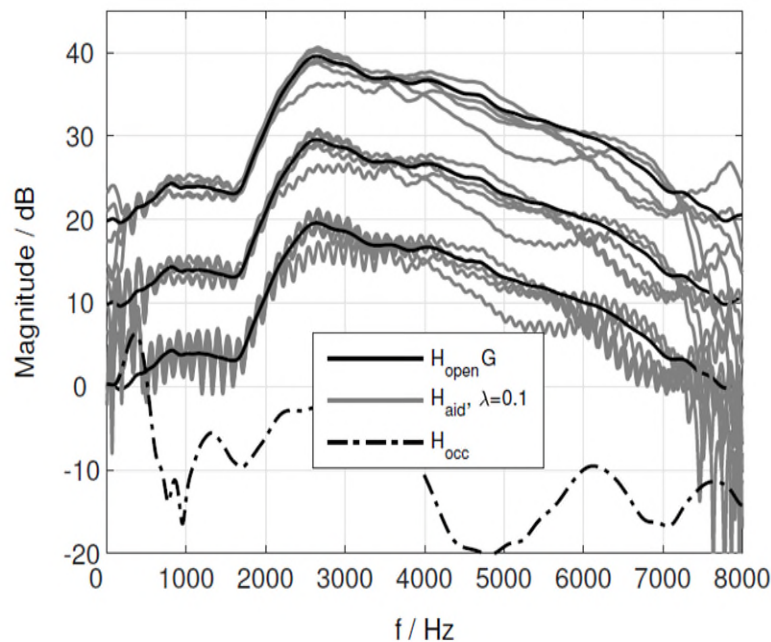
$$\tilde{\mathbf{v}}_{\Delta} = \left( \mathbf{H}_{m,\Delta}^T \mathbf{G}_{\Delta}^T \mathbf{G}_{\Delta} \mathbf{H}_{m,\Delta} \right)^{-1} \mathbf{H}_{m,\Delta}^T \mathbf{G}_{\Delta}^T \left( \mathbf{G}_{\Delta} \tilde{\mathbf{h}}_{open,\Delta} - \tilde{\mathbf{h}}_{occ,\Delta} \right)$$

# 1. Acoustically transparent sound presentation

- **Single/Multi-Loudspeaker Equalization**



(a)  $N = 1$ .



(b)  $N = 2$ .

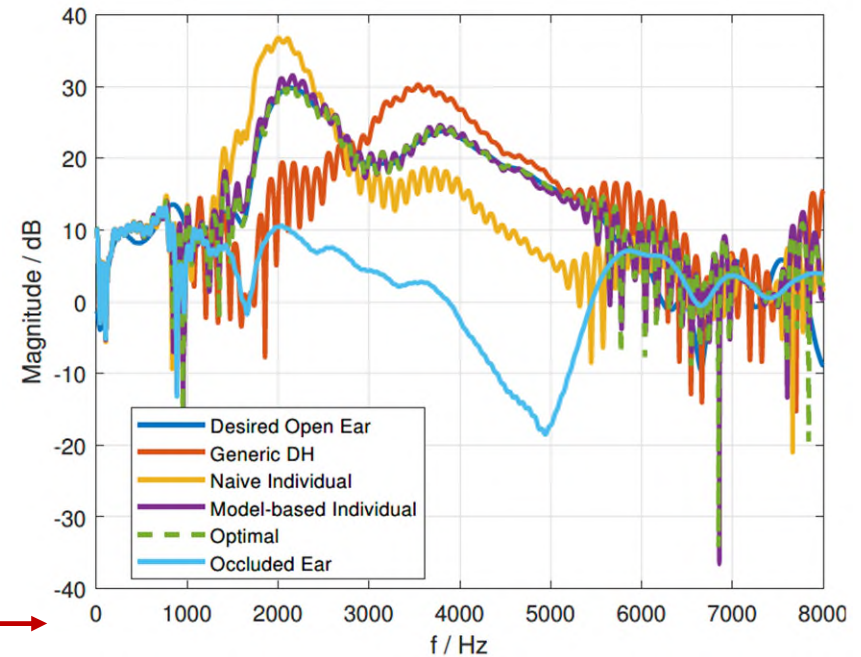
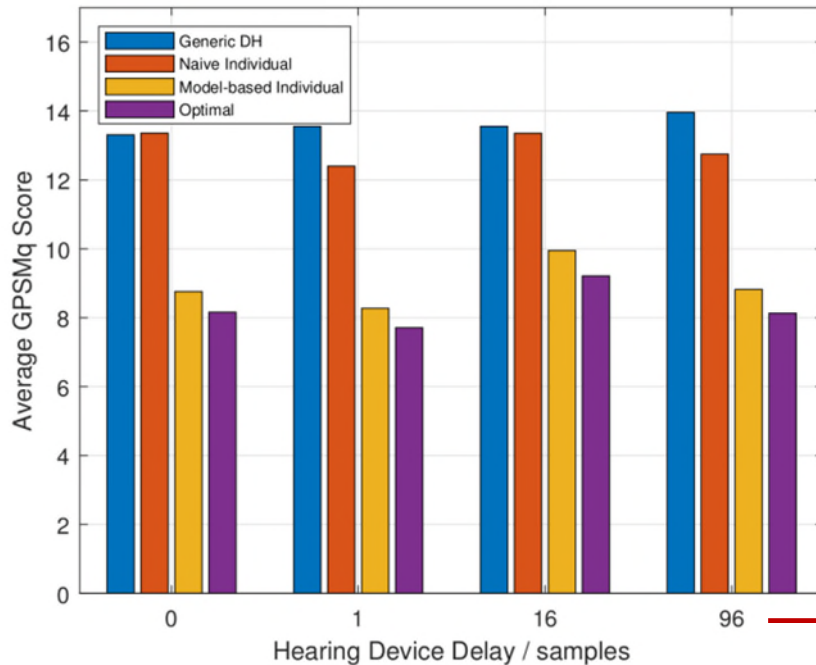
Robust equalization possible both using 1 and 2 loudspeakers

$f_s = 16 \text{ kHz}$ ,  $\tau = 6 \text{ ms}$ ,  
 $L_A = 99$ ,  $d_H = 32$ ,  $\lambda = 0.1$ ,  
 $l = 4$ ,  $G_0 = [0, 10, 20] \text{ dB}$

# 1. Acoustically transparent sound presentation

- **Single/Multi-Loudspeaker Equalization**

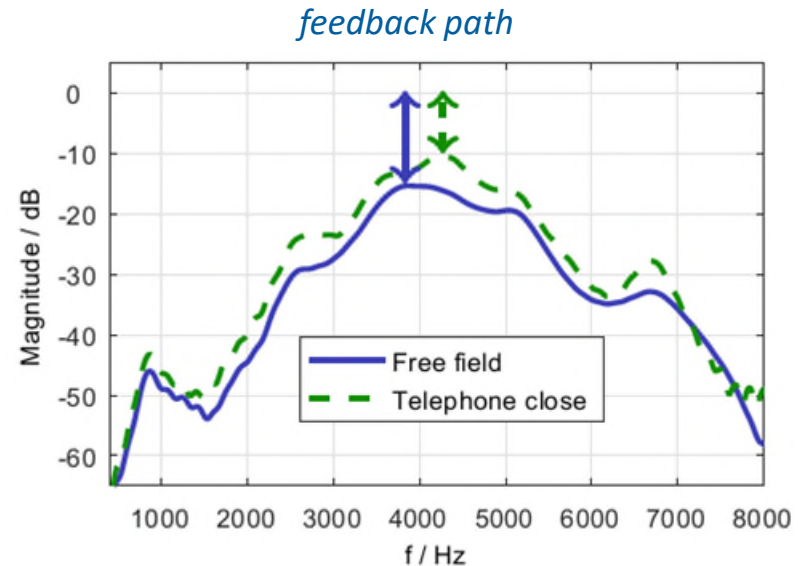
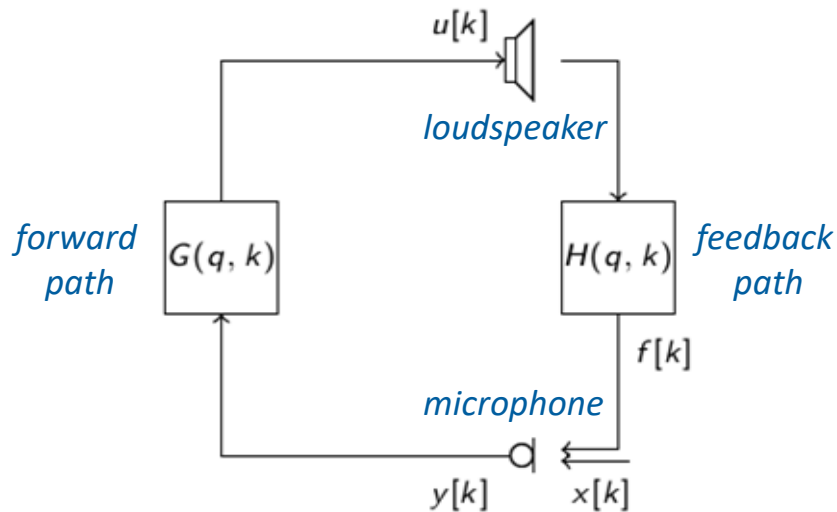
- **Individualized equalization filters** (based on individualized electro-acoustic model) outperform equalization filters based on dummy-head measurements or based on in-ear microphone signal





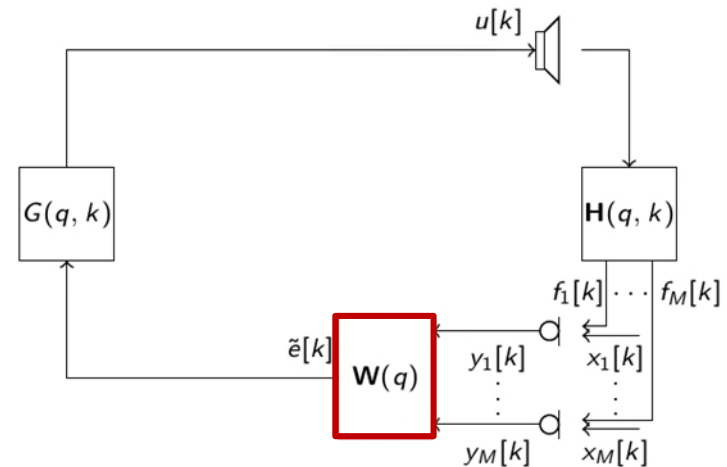
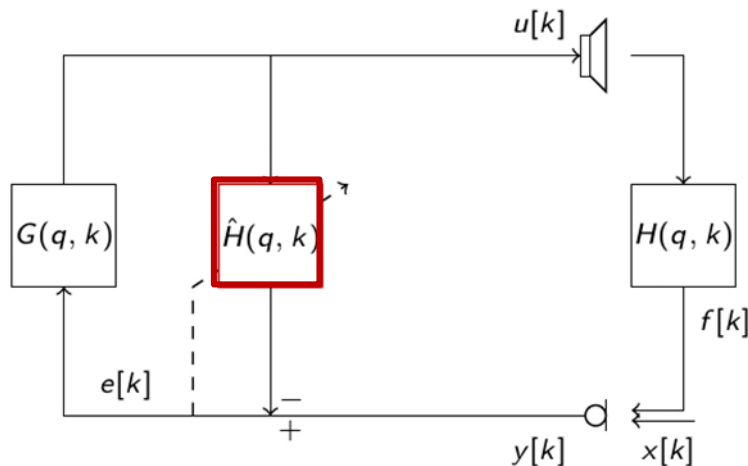
## 2. Acoustic feedback cancellation

- Feedback arises due to **acoustic coupling** between loudspeaker(s) and microphone(s)



## 2. Acoustic feedback cancellation

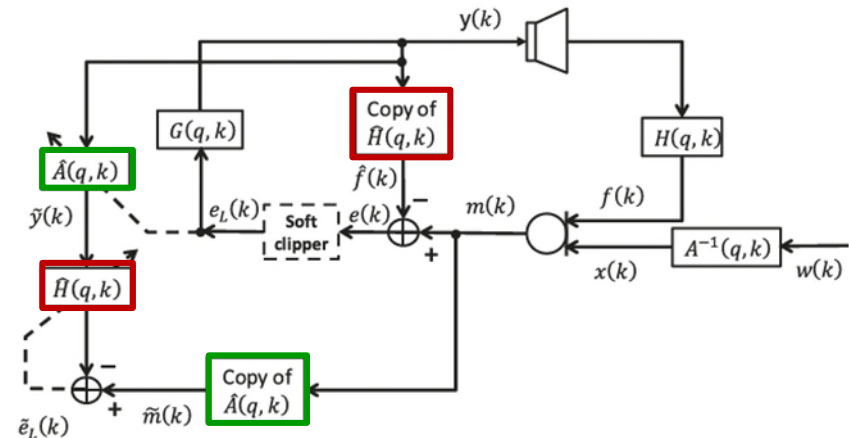
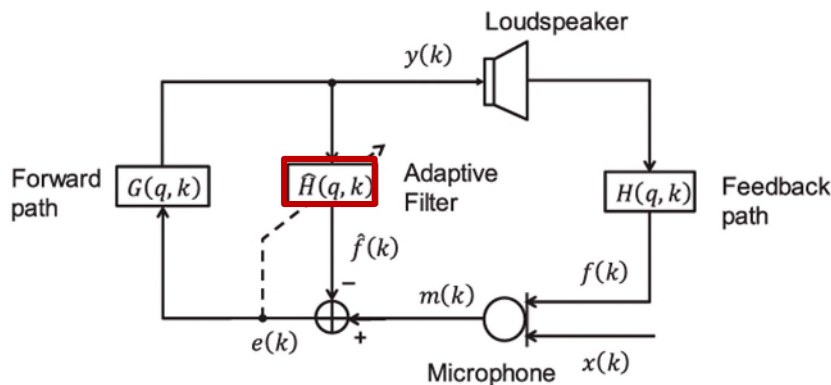
- Feedback arises due to **acoustic coupling** between loudspeaker(s) and microphone(s)
- Feedback suppression approaches:**
  - Feedforward suppression → **distortion**
  - Adaptive feedback cancellation → decorrelation between loudspeaker and incoming signal
  - Spatial filtering → requires multiple microphones



## 2. Acoustic feedback cancellation

- Adaptive feedback cancellation

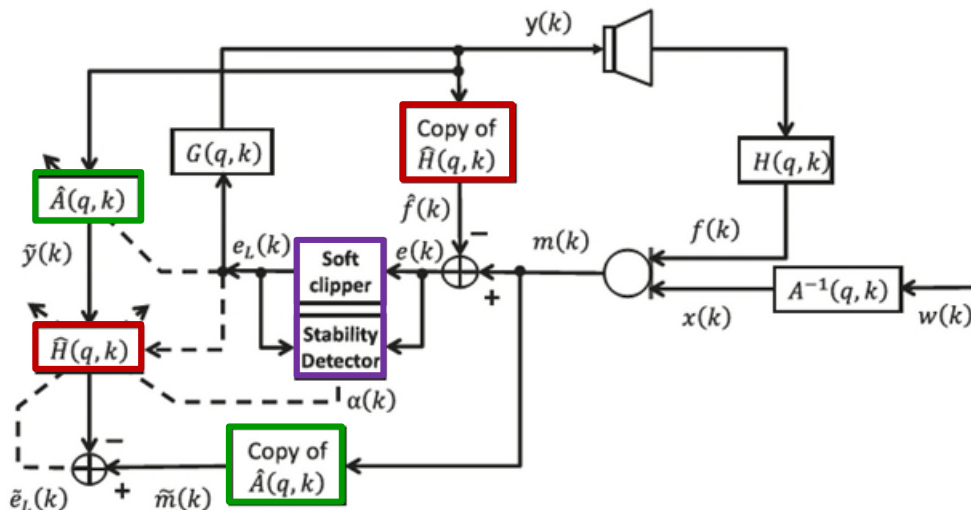
1. Normalized least mean squares (NLMS): bias due to correlation, fast re-convergence from howling
2. Prediction error method (PEM-NLMS): pre-whitening based on auto-regressive model  $\rightarrow$  reduced bias, but slow re-convergence from howling



## 2. Acoustic feedback cancellation

### Adaptive feedback cancellation

1. Normalized least mean squares (NLMS): bias due to correlation, fast re-convergence from howling
2. Prediction error method (PEM-NLMS): pre-whitening based on auto-regressive model → reduced bias, but slow re-convergence from howling
3. **Hybrid algorithm (H-NLMS)**: switched combination of NLMS and PEM-NLMS update, controlled by soft-clipping-based stability detector



$$\hat{\mathbf{h}}(k+1) = \hat{\mathbf{h}}(k) + [1 - \alpha(k)] \underbrace{\mu_1(k) \tilde{\mathbf{y}}(k) \tilde{e}_L(k)}_{\text{PEM-NLMS}} + \alpha(k) \underbrace{\mu_2(k) \mathbf{y}(k) e_L(k)}_{\text{NLMS}}$$

Binary control signal:

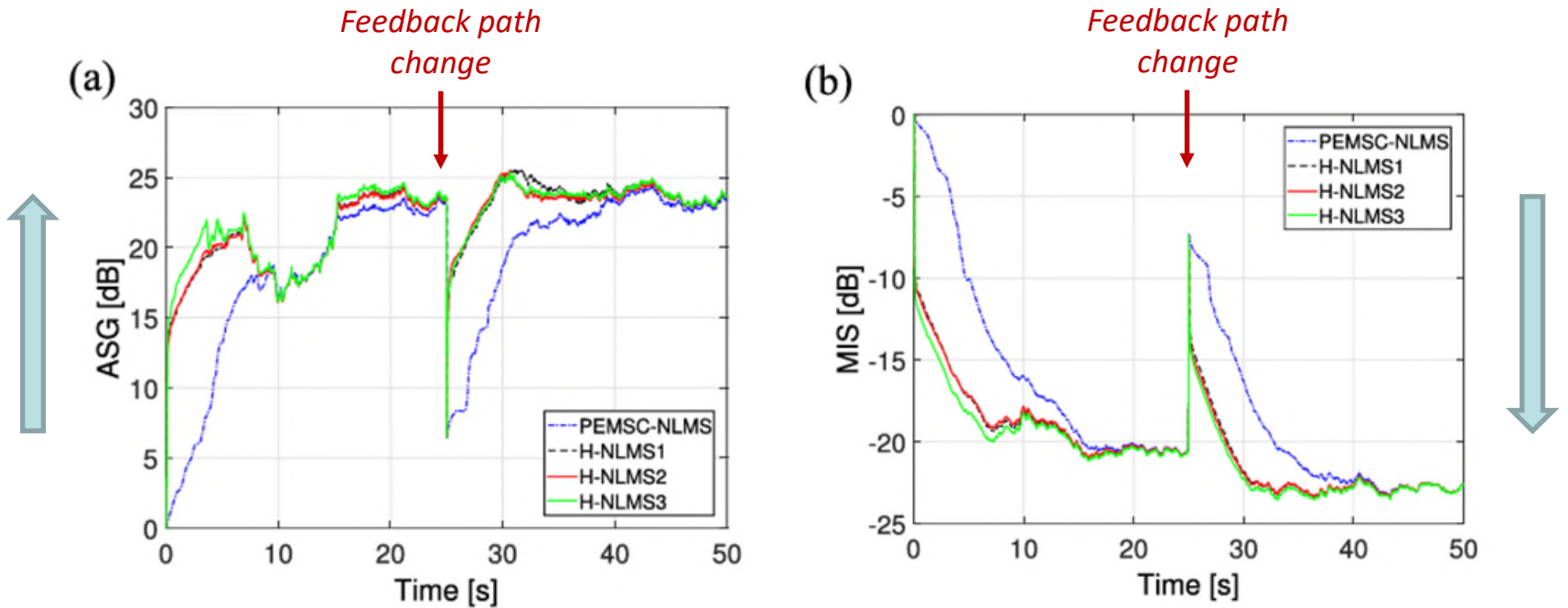
$$\alpha(k) = I\{|e(k) - e_L(k)| < \gamma\}$$

$$e_L(k) = \beta \tanh\left(\frac{e(k)}{\beta}\right)$$

[Nordholm et al., JASA, 2018]

## 2. Acoustic feedback cancellation

- Simulation results: added stable gain (ASG), misalignment (MIS)



**H-NLMS algorithm converges much faster than PEM-NLMS while maintaining similar misalignment**

$f_s = 16 \text{ kHz}$ ,  $L_h = 64$ ,  $L_a = 20$   
 (Levinson-Durbin), forward path:  $\tau = 6 \text{ ms}$ ,  $G_0 = 45 \text{ dB}$ ,  
 $\beta = 2$ ,  $\gamma = 0.15$ ,  $\mu_1 = 0.001$ ,  
 $\mu_2 = [0.1, 0.2, 0.5]$

## 2. Acoustic feedback cancellation

- **Spatial filtering:** reduce acoustic feedback in the vent microphone by steering a (robust) spatial null towards the hearing aid receiver
- Perfect feedback cancellation:

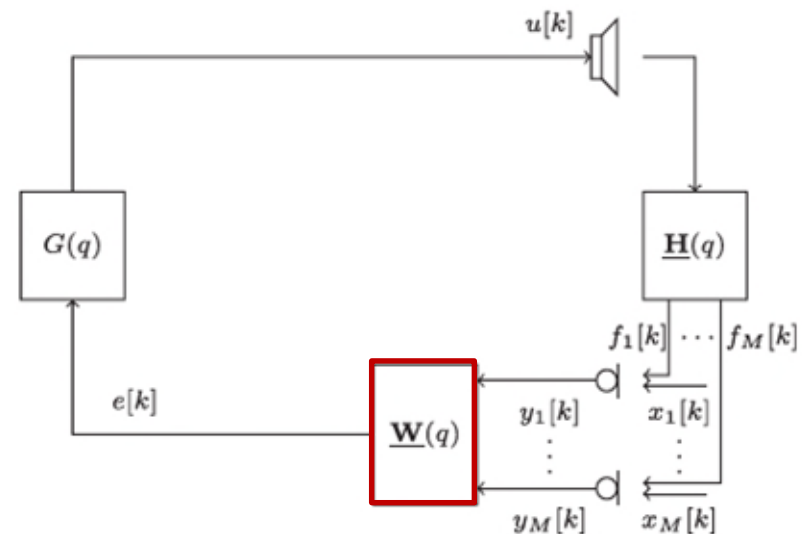
$$\underline{\mathbf{W}}^T(q)\underline{\mathbf{H}}(q) = 0$$

↑  
*feedback paths*

- Incoming signal preservation:

$$\underline{\mathbf{W}}^T(q)\underline{\tilde{\mathbf{D}}}(q) = q^{-L_d}$$

↑  
*relative transfer functions  
for incoming source*

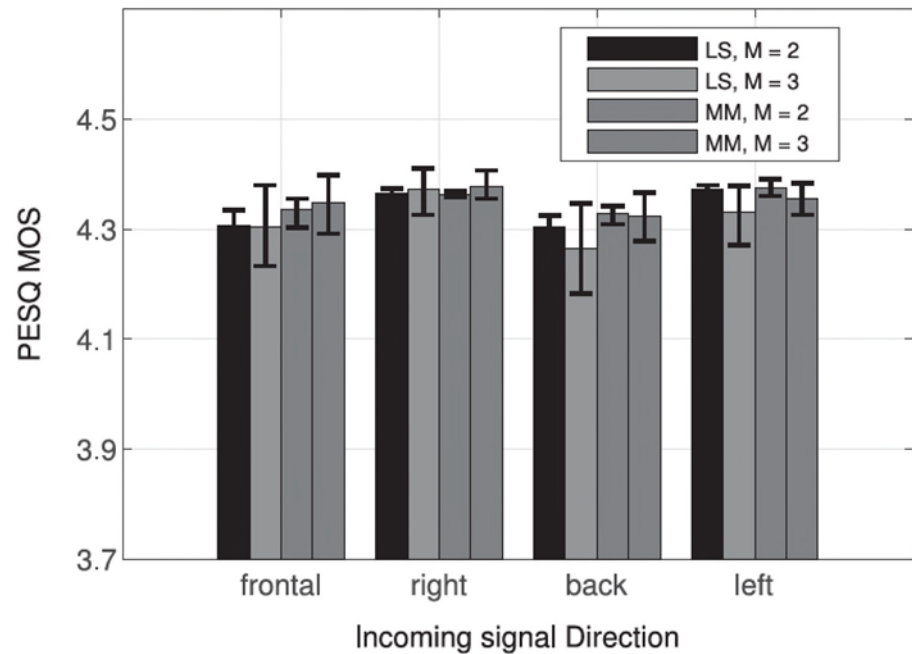
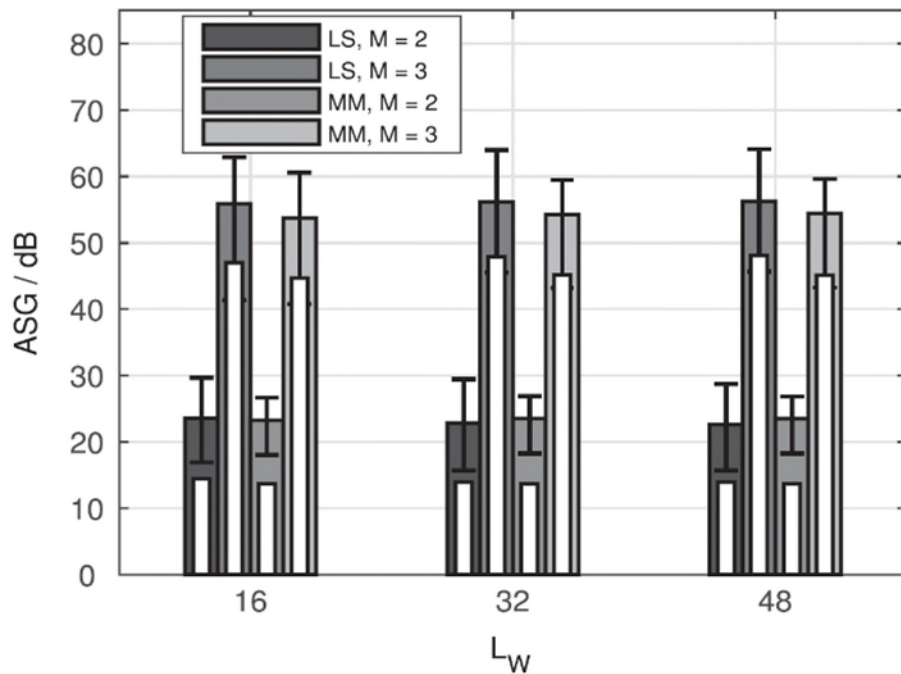


## 2. Acoustic feedback cancellation

- Robust design procedures for **fixed beamformer** based on (multiple) measurements of acoustic feedback paths

	Minimize residual feedback power	Maximize maximum stable gain
Hard constraint	$\min_{\mathbf{w}} \sum_{i=1}^I \ (\mathbf{H}^{(i)})\mathbf{w}\ _2^2$ <p>subject to <math>\tilde{\mathbf{D}}^{(j)}\mathbf{w} = \tilde{\mathbf{e}}_{L_d} \quad \forall j = 1, \dots, J</math></p> <p><i>Constrained least-squares problem</i></p>	$\min_{\mathbf{w}} \max_{\omega_n, i}  (\underline{\mathbf{H}}^{(i)})^H(\omega_n)\underline{\mathbf{W}}(\omega_n) ^2$ <p>subject to <math>\tilde{\mathbf{D}}^{(j)}\mathbf{w} = \tilde{\mathbf{e}}_{L_d} \quad j = 1, \dots, J</math></p> <p><i>Linear programming problem</i></p>
Soft constraint	$\min_{\mathbf{w}} \ \tilde{\mathbf{H}}\mathbf{w}\ _2^2 + \lambda \ \tilde{\mathbf{D}}\mathbf{w} - \tilde{\mathbf{e}}_{L_d}\ _2^2$ <p><i>Unconstrained least-squares problem</i></p>	$\min_{\mathbf{w}} \max_{\omega_n, i}  (\underline{\mathbf{H}}^{(i)})^H(\omega_n)\underline{\mathbf{W}}(\omega_n) ^2 + \lambda \ \tilde{\mathbf{D}}\mathbf{w} - \tilde{\mathbf{e}}_{L_d}\ _2^2$ <p><i>Quadratically constrained quadratic programming (QCQP) problem</i></p>

## 2. Acoustic feedback cancellation



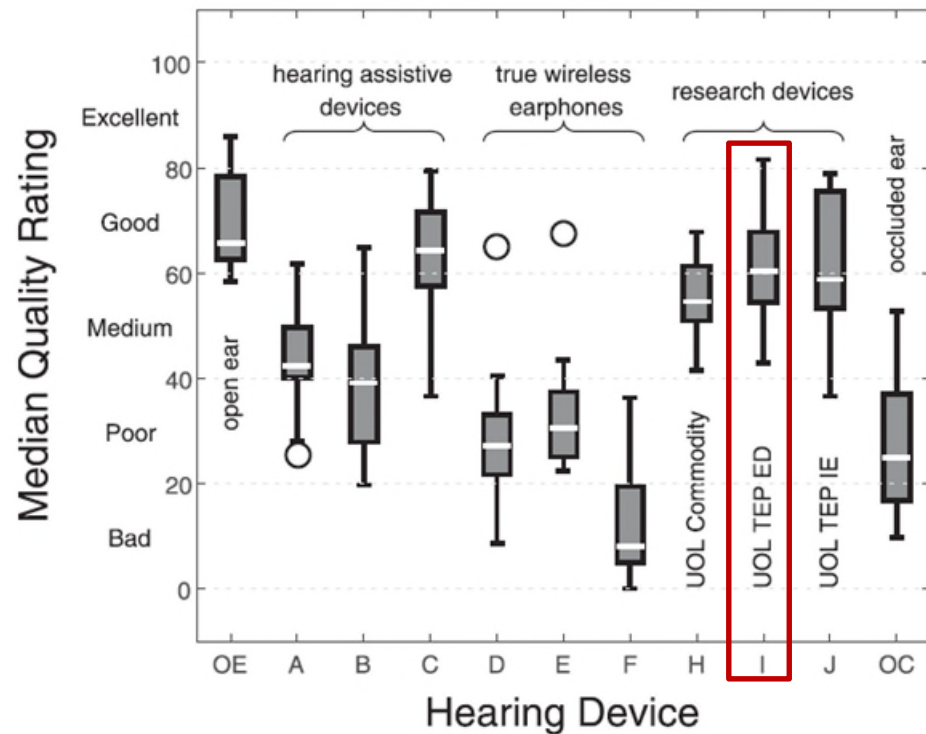
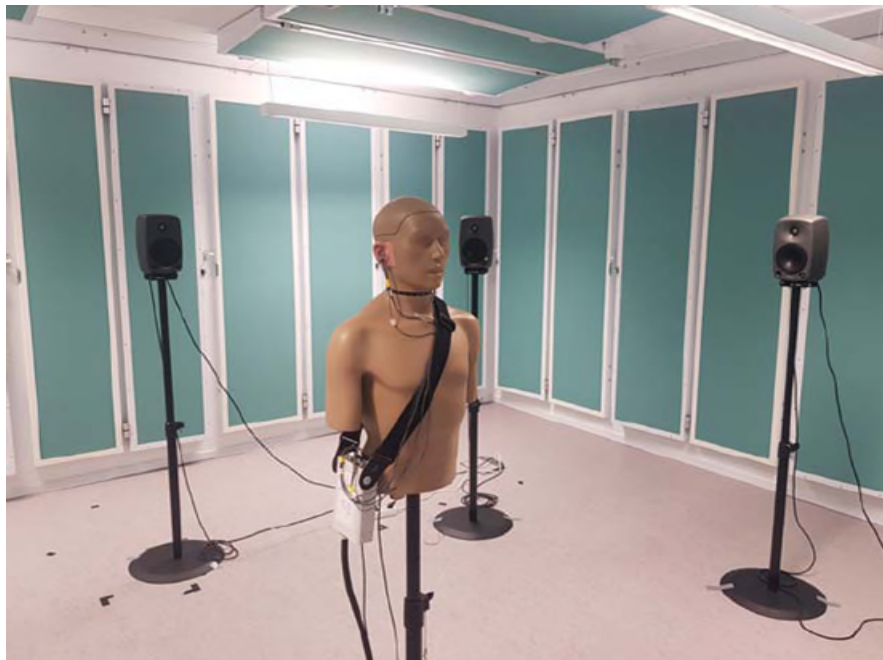
**Robust reduction of acoustic feedback of up to 50dB while hardly distorting incoming signal**

$f_s = 16 \text{ kHz}$ ,  $T_{60} = 300 \text{ ms}$ ,  
 $l = 20$ , 4 source directions,  
 forward path:  $\tau = 6 \text{ ms}$ ,  
 $G_0 = 45 \text{ dB}$ ,  $\varepsilon_{MSG} = 10 \text{ dB}$



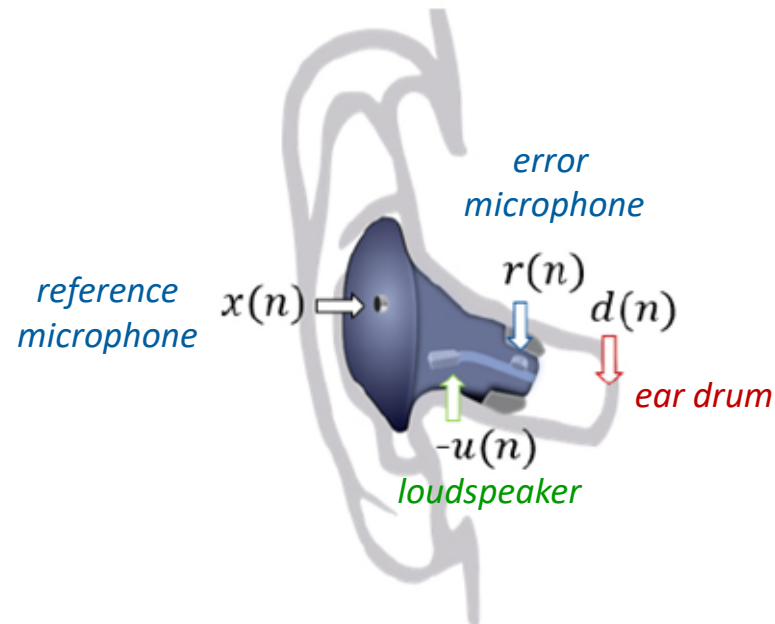
# Acoustically transparent sound presentation

- Acoustic transparency feature compared to six commercial hearables



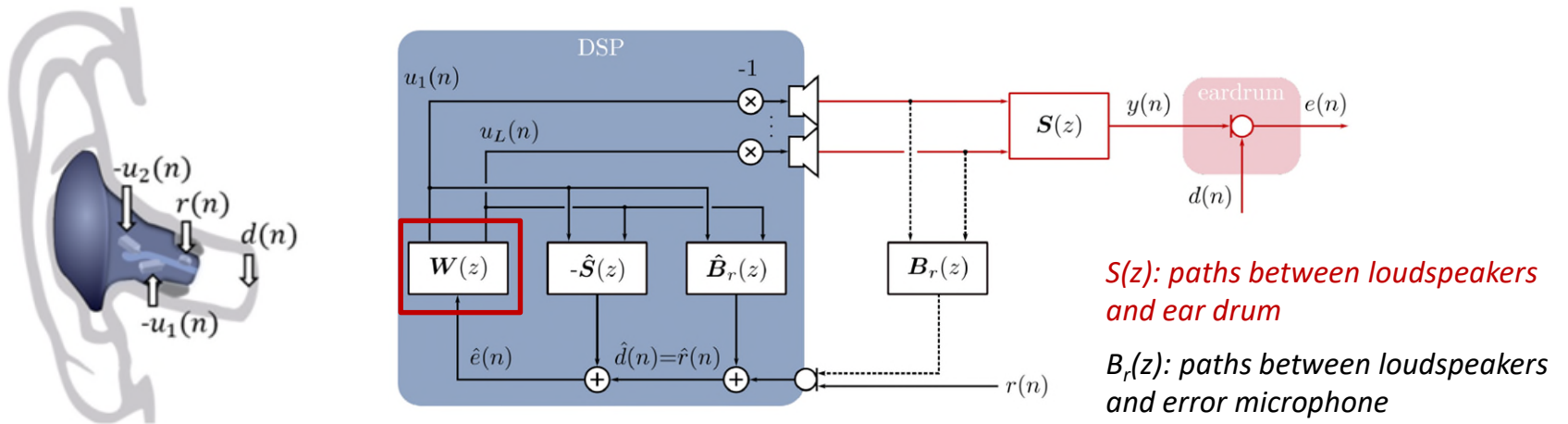
## 3. Active noise control

- **Aim:** play back anti-noise by one or more loudspeakers aiming at generating “zone of quiet” at certain position (e.g. ear drum)
- **Approaches:**
  - **Feedforward ANC:** filter reference microphone signal
  - **Feedback ANC:** filter error microphone signal



### 3. Active noise control

- Virtual sensing fixed feedback ANC exploiting **multiple loudspeakers**, aiming at minimizing sound pressure at **ear drum**



- Minimize power spectral density of sound pressure at ear drum

$$\Phi_{ee}(f) = \left(1 - \frac{|\Phi_{dr}(f)|^2}{\Phi_{dd}(f)\Phi_{rr}(f)}\right)\Phi_{dd}(f) + \left|\frac{\Phi_{dr}(f)}{\Phi_{rr}(f)} - \frac{W^T(f)S(f)}{1 + W^T(f)(\hat{S}(f) + B_r(f) - \hat{B}_r(f))}\right|^2 \Phi_{rr}(f)$$

Virtual microphone arrangement  $\xrightarrow{d(n) = r(n)}$   $\Phi_{ee}^{vma}(f) = \left|\frac{1}{1 + W^T(f)\hat{S}(f)}\right|^2 \Phi_{rr}(f)$

### 3. Active noise control

- Turn non-convex optimization problem into **convex optimization problem**

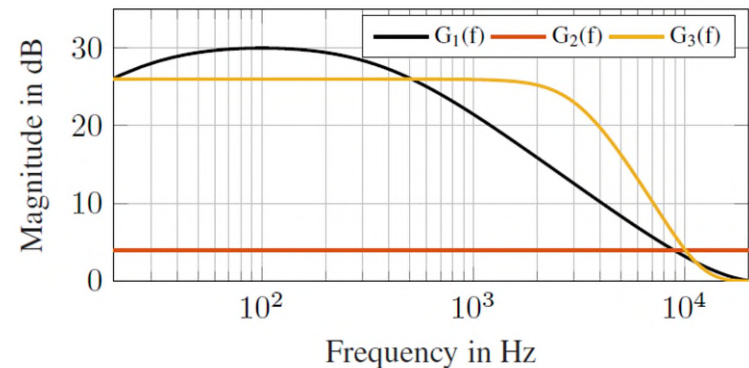
$$\min_{\mathbf{w}} \left| \frac{1}{1 + \mathbf{W}^T(f)\hat{\mathbf{S}}(f)} \right|^2 \Phi_{rr}(f) \quad \longrightarrow \quad \max_{\mathbf{w}} \sum_{k=0}^{L_{\text{DFT}}/2-1} \left| 1 + \mathbf{W}^T(\Omega_k)\hat{\mathbf{S}}(\Omega_k) \right|^2 G_1^2(\Omega_k)$$

**subject to constraints** (stability, amplification, gain)

$$|\varrho - \mathbf{W}^T(\Omega_k)\hat{\mathbf{S}}(\Omega_k)| \leq |\varrho + \mathbf{W}^T(\Omega_k)\hat{\mathbf{S}}(\Omega_k)| + 2\rho, \forall \Omega_k$$

$$1 \leq G_2(\Omega_k)|1 + \mathbf{W}^T(\Omega_k)\hat{\mathbf{S}}(\Omega_k)|, \forall \Omega_k$$

$$W_l(\Omega_k) \leq G_3(\Omega_k), \forall \Omega_k$$



[Rivera Benois et al., ICASSP, 2022]

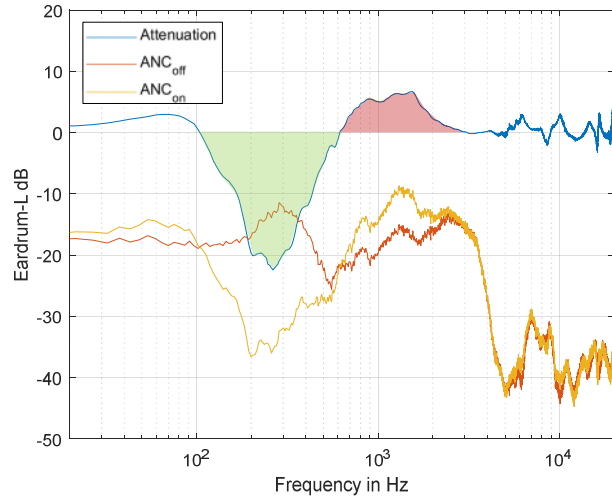
# 3. Active noise control

Loudspeakers

Classic  
Single

$$\hat{B}_r(z)$$

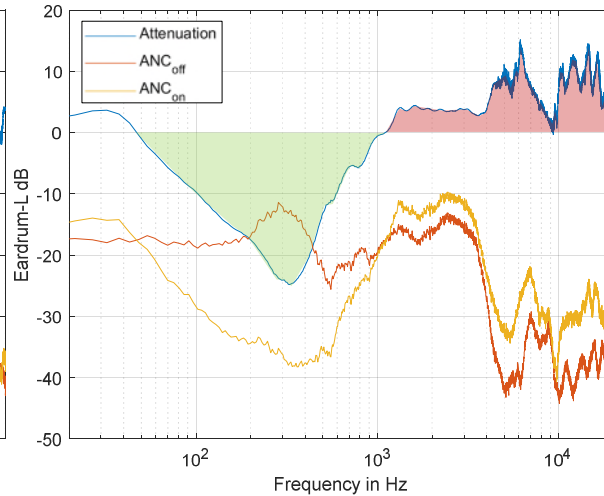
Available acoustic paths



Virtual Microphone  
Single

$$\hat{B}_r(z)$$

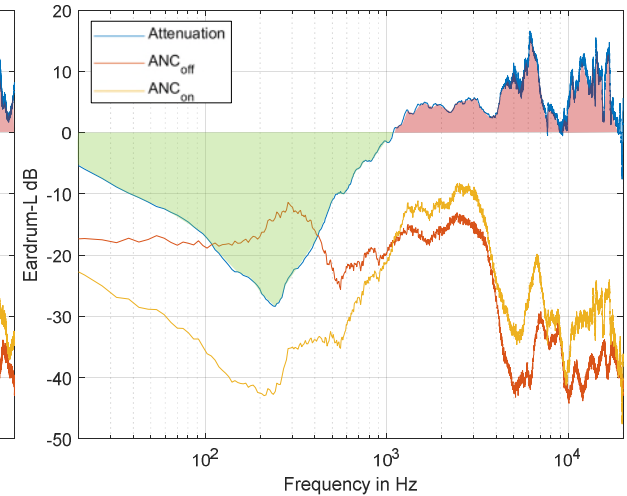
$$\hat{S}(z)$$



Virtual Microphone  
Multi

$$\hat{B}_r(z)$$

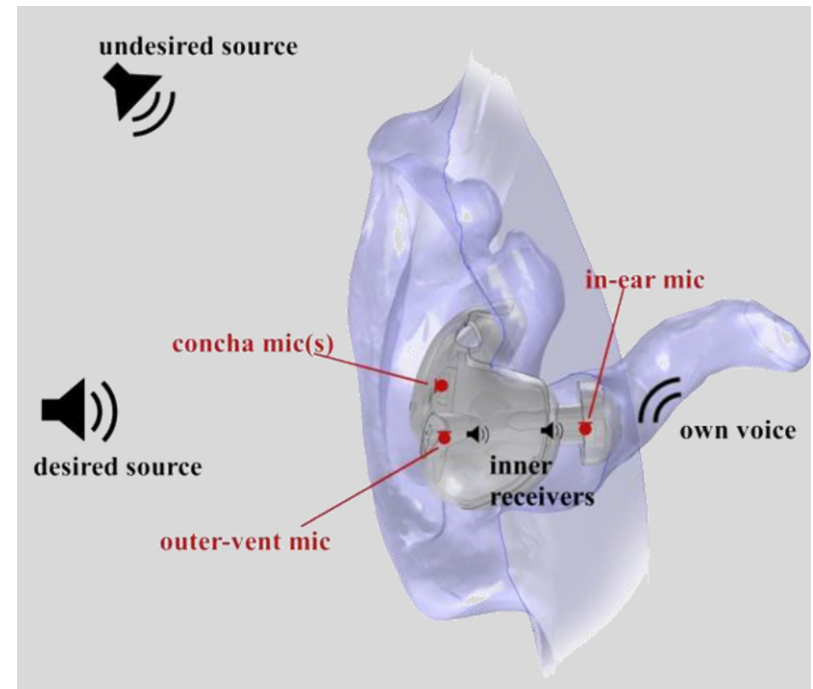
$$\hat{S}(z)$$



**Multi-loudspeaker virtual sensing ANC approach improves attenuation magnitude and bandwidth**

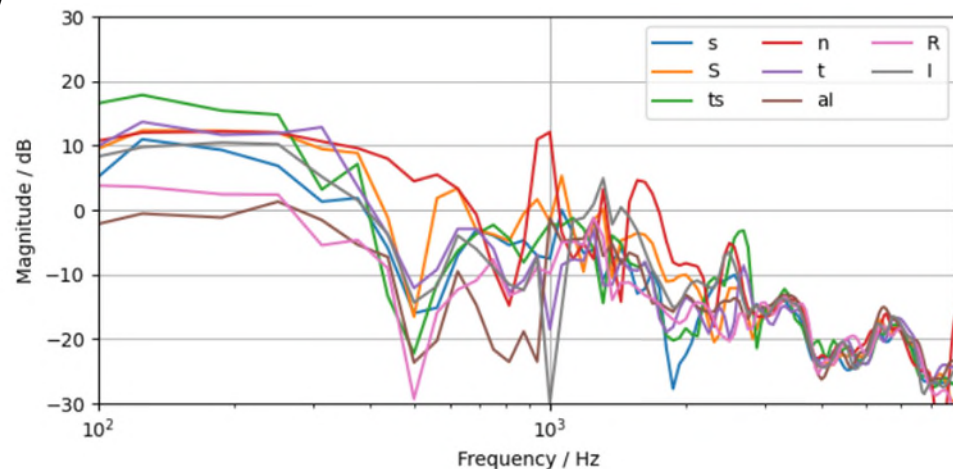
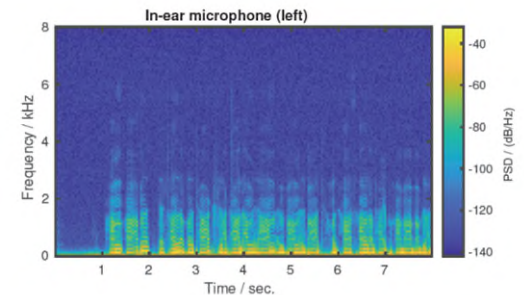
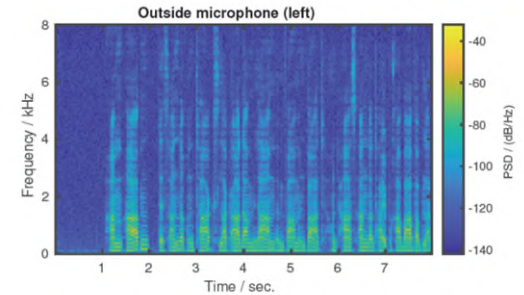
## 4. Own voice extraction

- **Aim:** enhance own voice of user wearing earpiece in noisy acoustic environment (e.g. industrial workplace)
- **Approach:** exploit in-ear microphone, possibly in combination with outer microphone(s)



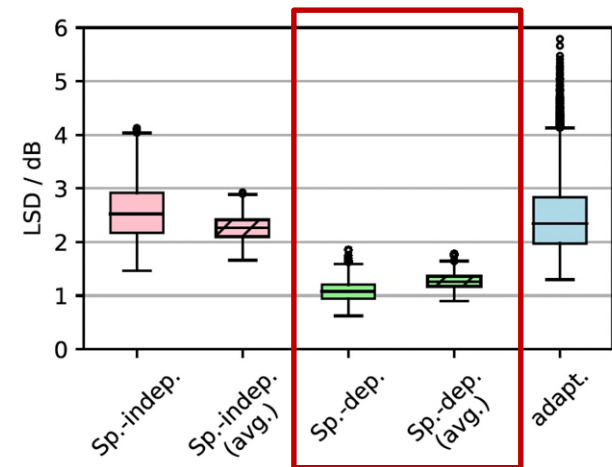
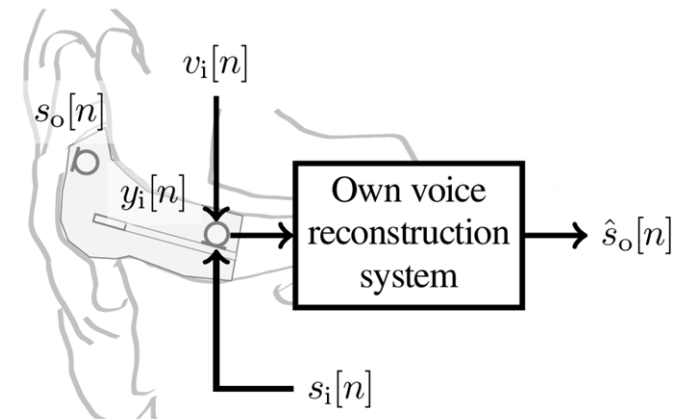
## 4. Own voice extraction

- **Different characteristics** for own voice and external noise at in-ear and outer microphones
  - **Outer microphone:** full bandwidth own voice, possibly low SNR (external noise)
  - **In-ear microphone:** bandlimited own voice (up to about 2 kHz), high SNR (external noise), body noise
- **Relative transfer characteristics for own voice**
  - Time-varying (speech-dependent)
  - User- and device-dependent



## 4. Own voice extraction

- **Objectives of algorithm:** estimate clean speech signal at outer microphone from
  - **in-ear microphone:** combined bandwidth extension, equalization and noise reduction (body + external noise)
  - **in-ear and outer microphone**
- **Limited training data available:**
  - use **models** to generate simulated data (data augmentation):
    - Fixed relative transfer function (sp.-indep.)
    - Phoneme-dependent relative transfer function (sp.dep.)
  - **domain transfer** (train with simulated data, fine-tune with real recordings)

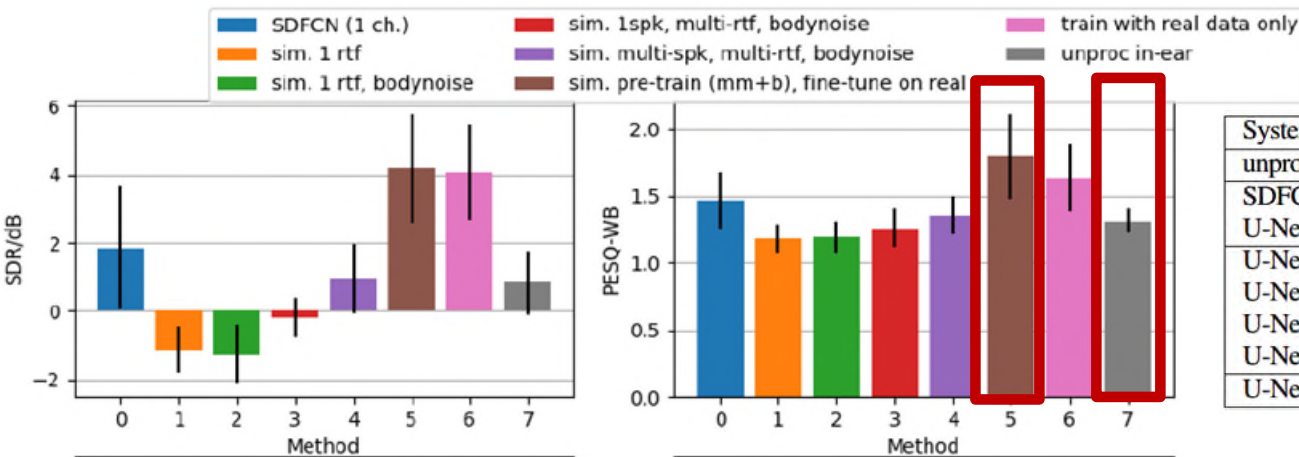
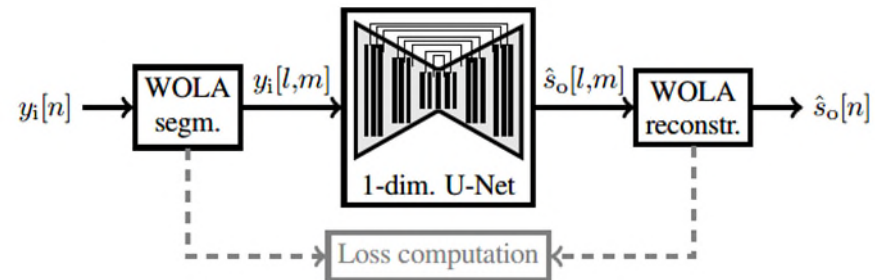




## 4. Own voice extraction

- Results:

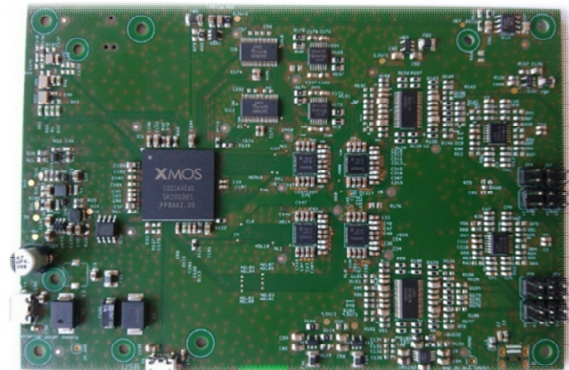
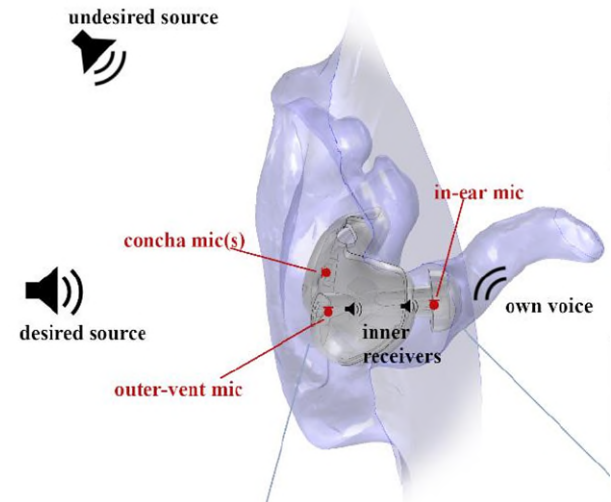
- Based on bandwidth extension system using U-Net architecture [Wang 2021]
- Only exploiting in-ear microphone signal, only body noise (no external noise)
- Different training procedures** (real data, simulated data with one/multiple RTFs and one/multiple talkers, **simulated data + fine-tuning with real data**)



System	Data	RTFs used	LSD / dB	PESQ
unproc.	-	-	2.51	1.31
SDFCN	[R]	-	1.53	1.47
U-Net	[R]	-	1.48	1.64
U-Net	[S]	1T, s-RTF	1.35	1.18
U-Net	[S+]	1T, s-RTF	1.54	1.19
U-Net	[S+]	1T, m-RTF	1.51	1.26
U-Net	[S+]	14T, m-RTF	1.24	1.36
U-Net	[S+R]	14T, m-RTF	<b>1.05</b>	<b>1.80</b>

## Current / Future work

- **Direction-selective** acoustic transparency and active noise control, steered by CASA
- **Deep learning-based active noise control**
- **Speech communication exploiting in-ear microphone** (phoneme-dependent own voice models, DNN-based algorithms)
- Individualized and phoneme-dependent occlusion models and **active occlusion reduction**
- Implementation on **low-latency processor** (cooperation with Fraunhofer HSA)
- **Integration** with speech enhancement algorithms and self-adjusted hearing support



## Acknowledgments / references



- S. Vogl and M. Blau, "Individualized prediction of the sound pressure at the eardrum for an earpiece with integrated receivers and microphones," *The Journal of the Acoustical Society of America*, vol. 145, no. 2, pp. 917-930, Feb. 2019.
- H. Schepker et al., "Null-steering beamformer-based feedback cancellation for multi-microphone hearing aids with incoming signal preservation," *IEEE/ACM Transactions on Audio, Speech, and Language Processing*, vol. 27, no. 4, pp. 679-691, Apr. 2019.
- F. Denk et al., "A one-size-fits-all earpiece with multiple microphones and drivers for hearing device research," in *Proc. AES Conference on Headphone Technology*, San Francisco, USA, Aug. 2019.
- H. Schepker, S. Nordholm, S. Doclo, "Acoustic feedback suppression for multi-microphone hearing devices using a soft-constrained null-steering beamformer," *IEEE/ACM Trans. Audio, Speech and Language Processing*, vol. 28, pp. 929-940, 2020.
- F. Denk, H. Schepker, S. Doclo, B. Kollmeier, "Acoustic Transparency in Hearables - Technical Evaluation," *Journal of the Audio Engineering Society*, vol. 68, no. 7/8, pp. 508-521, Jul./Aug. 2020.
- H. Schepker, F. Denk, B. Kollmeier, S. Doclo, "Acoustic Transparency in Hearables - Perceptual Sound Quality Evaluations," *Journal of the Audio Engineering Society*, vol. 68, no. 7/8, pp. 495-507, Jul./Aug. 2020.
- F. Denk and B. Kollmeier (2021). The Hearpiece database of individual transfer functions of an in-the-ear earpiece for hearing device research.
- H. Schepker, F. Denk, B. Kollmeier, S. Doclo, "Robust single- and multi-loudspeaker least-squares-based equalization for hearing devices," *EURASIP Journal of Audio, Speech and Music Processing*, 2022.
- P. Rivera Benois, R. Roden, M. Blau, S. Doclo, "Optimization of a Fixed Virtual Sensing Feedback ANC Controller for In-Ear Headphones with Multiple Loudspeakers," in *Proc. IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP)*, Singapore, May 2022.
- M. Ohlenbusch, C. Rollwage, S. Doclo, "Training Strategies for Own Voice Reconstruction in Hearing Protection Devices using an In-ear Microphone," in *Proc. International Workshop on Acoustic Signal Enhancement (IWAENC)*, Bamberg, Germany, Sep. 2022.
- M. Ohlenbusch, C. Rollwage, S. Doclo, "Speech-dependent Modeling of Own Voice Transfer Characteristics for In-ear Microphones in Hearables," in *Proc. Forum Acusticum*, Torino, Italy, Sep. 2023.



# Questions ?

<http://www.sigproc.uni-oldenburg.de>

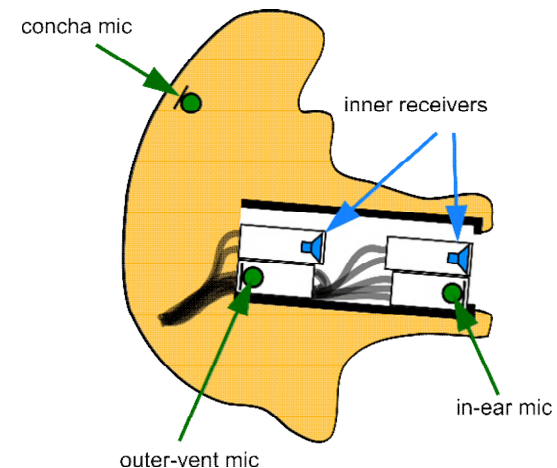
**You Tube** Signal Processing Uni Oldenburg





## Acoustically Transparent Earpiece (v1)

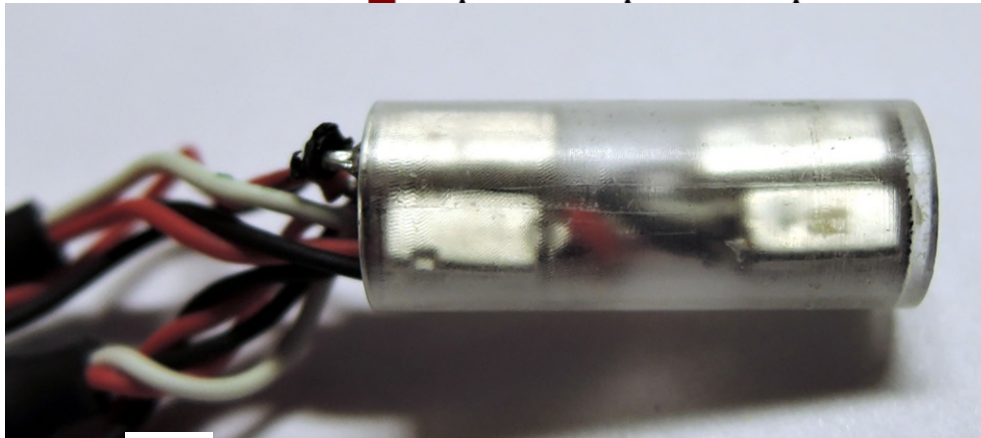
- **Custom in-the-ear earpiece** with multiple integrated microphones and receivers and relatively open acoustics
  - Vent/core: **2 microphones and 2 receivers** (woofer/tweeter)
  - Concha: **1 microphone**
- Insertion into individual silicone ear mould or generic earplugs



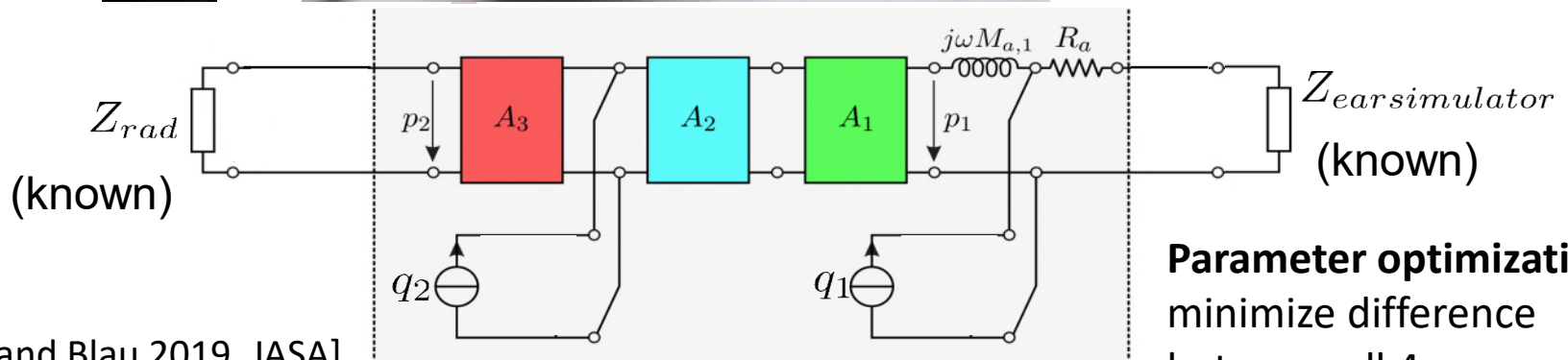
[Denk et al., *International Journal of Audiology*, 2018]

# Electro-acoustic model

## - Earpiece Model (Fixed)



C711 coupler

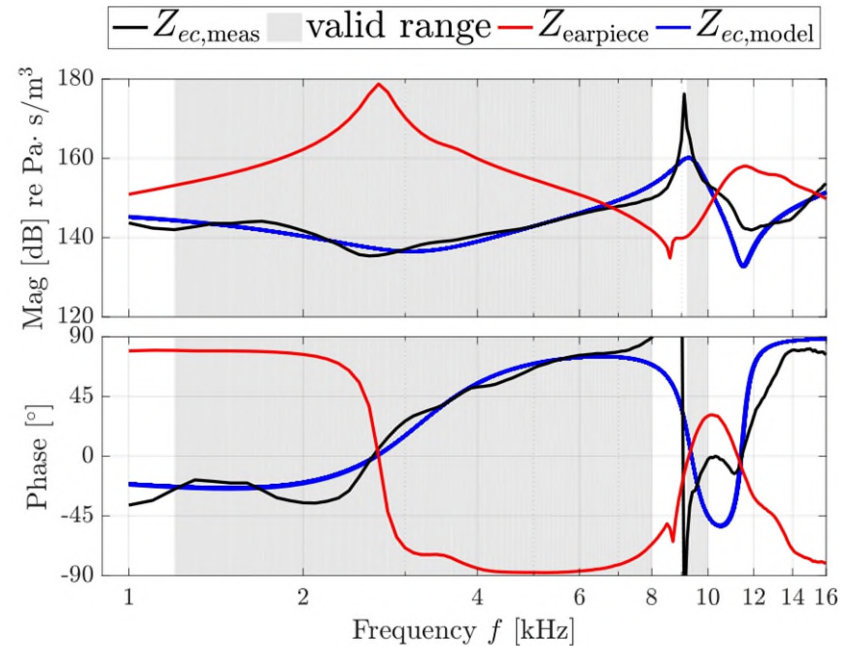
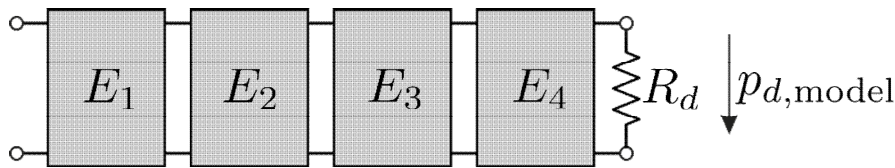


**Parameter optimization:**  
 minimize difference  
 between all 4 measured  
 and modeled transfer  
 functions

[Vogl and Blau 2019, JASA]

# Electro-acoustic model

## - Ear Canal Model (Individualized)



**Parameter optimization** (4 radii, 1 length, 1 resistive load) by minimizing the difference between measured and modeled ear canal (Nelder-Mead simplex optimization procedure):

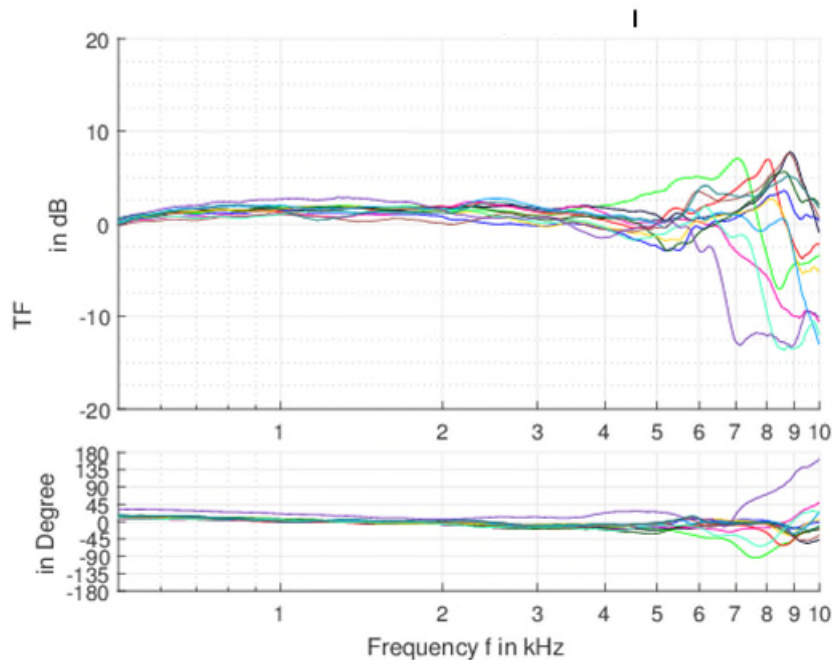
$$J(p) = \sum_{f_{\text{valid}}} (db(Z_{ec,\text{meas}}) - db(Z_{ec,\text{model}}(p)))^2 + 10 \cdot (arg(Z_{ec,\text{meas}}) - arg(Z_{ec,\text{model}}(p)))^2$$



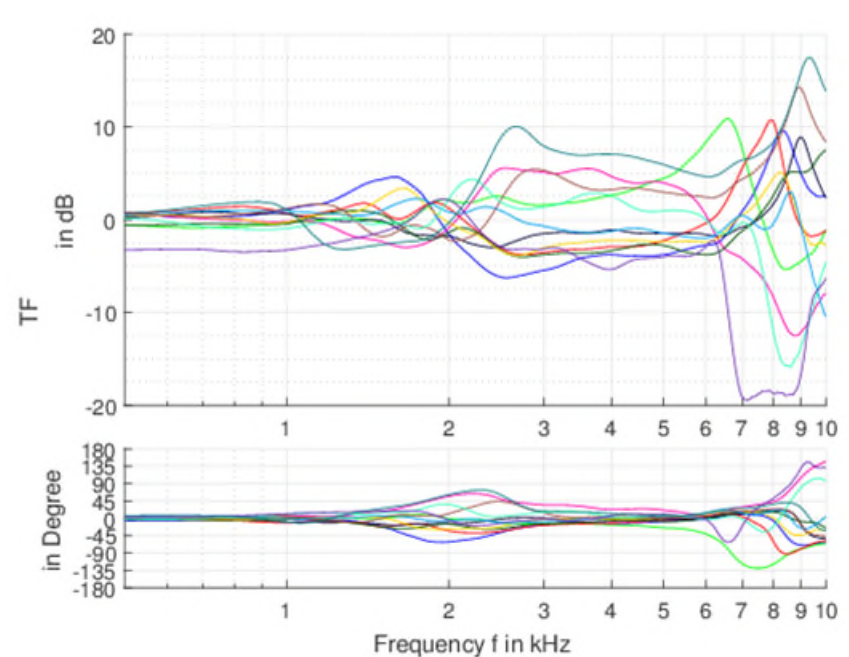
# Electro-acoustic model

## - Evaluation (sound pressure at ear drum) for 12 subjects

$\frac{p_{d,Model} \text{ (i.e. individualized)}}{p_{d,Probe Tube Meas}}$

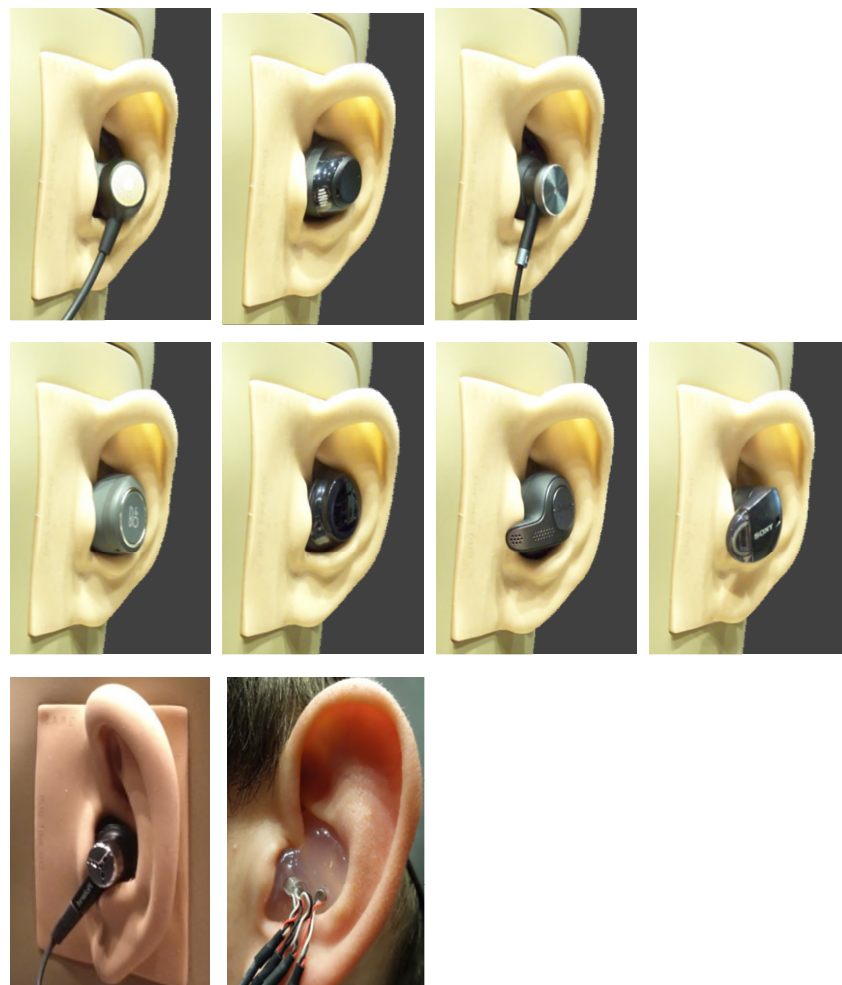


$\frac{p_{d,from average correction of p1} \text{ (i.e. non-individualized)}}{p_{d,Probe Tube Meas}}$

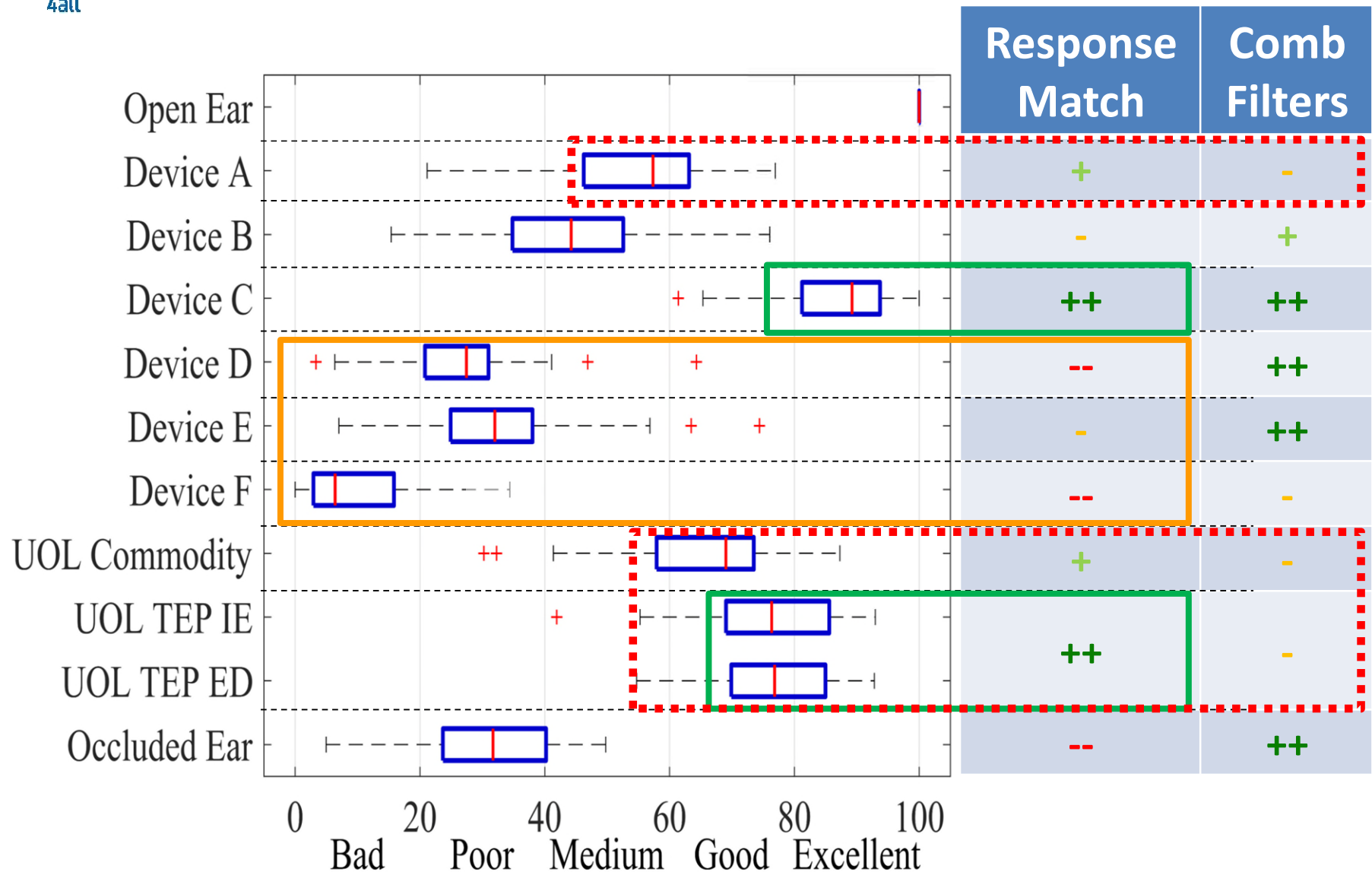


**accurate prediction of sound pressure at ear drum possible using individualized electro-acoustic model up to about 6 kHz**

- **7 commercial hearables**
  - 3 hearing support: **Devices A-C**
  - 4 wireless earbuds: **Devices D-G**
- **2 research prototypes**
  - UOL Commodity: consumer hardware based hearing aid prototype  
[Schädler 2017, Buhl, Denk et al. 2019]
  - UOL Acoustically Transparent Earpiece: Adaptation to individual ear acoustics  
[Denk et al. 2018, Schepker, Denk et al. 2019]



# Results: Open-ear reference given



# Results: No reference given

