# Highlights from Hearing4all for patients with hearing aids and the subclinical population

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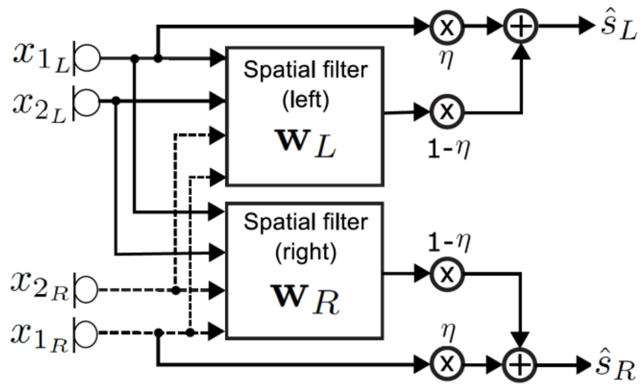
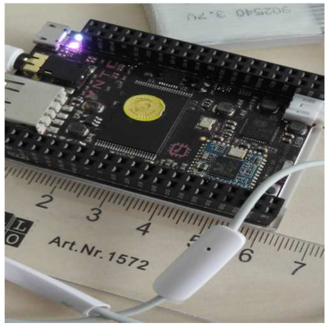
# Highlights

- **Algorithms for hearing devices**
  - Speech enhancement: binaural noise reduction and dereverberation
  - Speech control based on deep learning
  - Binaural bandwidth-adaptive dynamic compression
- **Hardware / Technology**
  - Acoustically transparent earpiece
  - Ultra low-power processor architecture
  - Demonstrator platforms



# Algorithms for hearing devices

# 1. Speech enhancement

- Hearing impaired suffer from a loss of speech understanding in adverse acoustic environments with competing speakers, background noise and reverberation

Apply **acoustic signal pre-processing techniques** in order to improve speech quality and intelligibility

# 1. Speech enhancement

- **Binaural noise reduction and cue preservation**
  - Preserving binaural cues (ITD, ILD, IC) in noise reduction algorithms is important both for ***spatial awareness*** and for ***speech intelligibility***
  - Several extensions of binaural speech enhancement approaches (MVDR beamformer, MWF), that preserve the binaural cues both for diffuse noise as well as for interfering sources

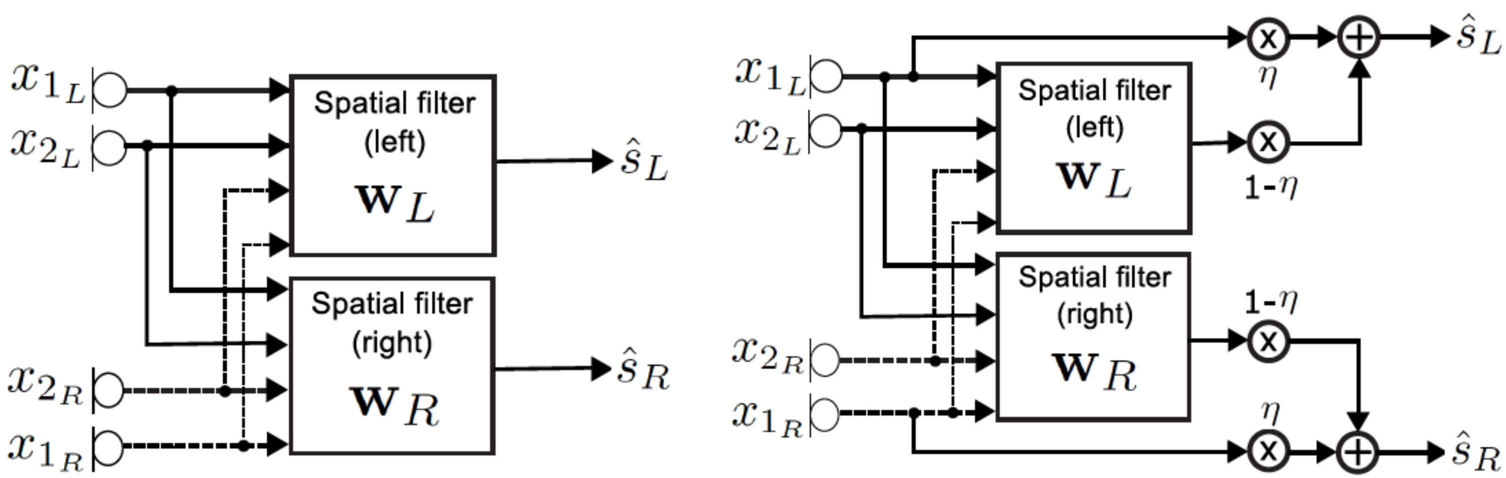
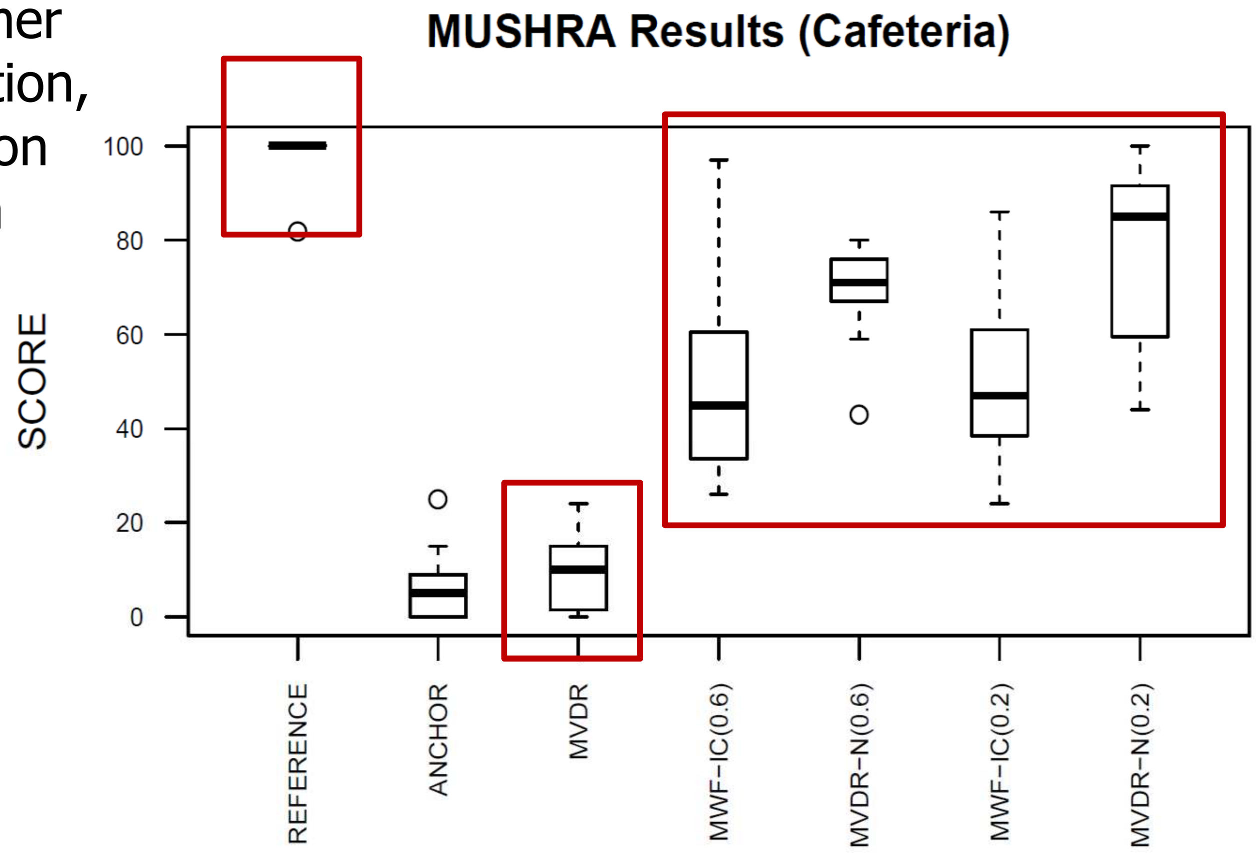



**Figure 1.2** Block diagram for binaural spatial filtering: a) incorporating constraints into spatial filter design, b) mixing with scaled reference signals.

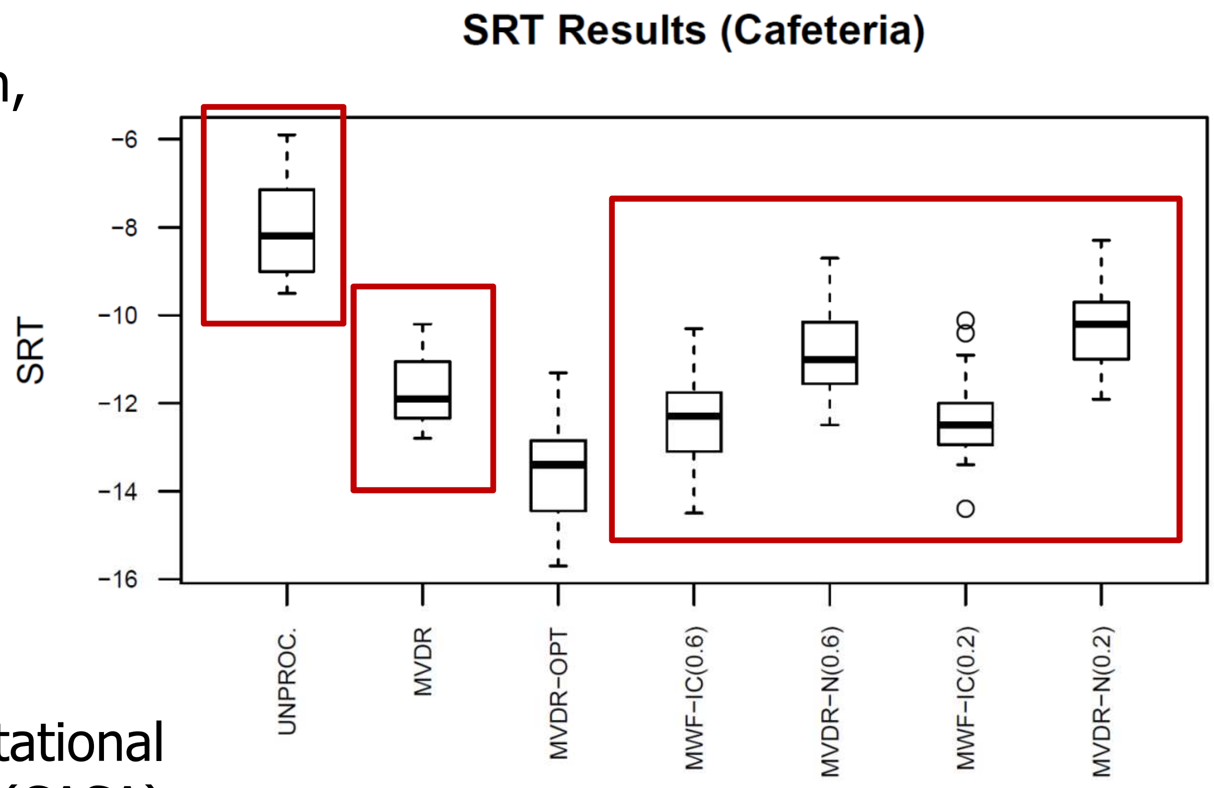
# 1. Speech enhancement

- **Binaural noise reduction and cue preservation**
  - Perceptual evaluation of binaural MVDR beamformer with partial noise estimation, exploiting IC discrimination ability of auditory system
  - **Cue preservation improves spatial quality …**



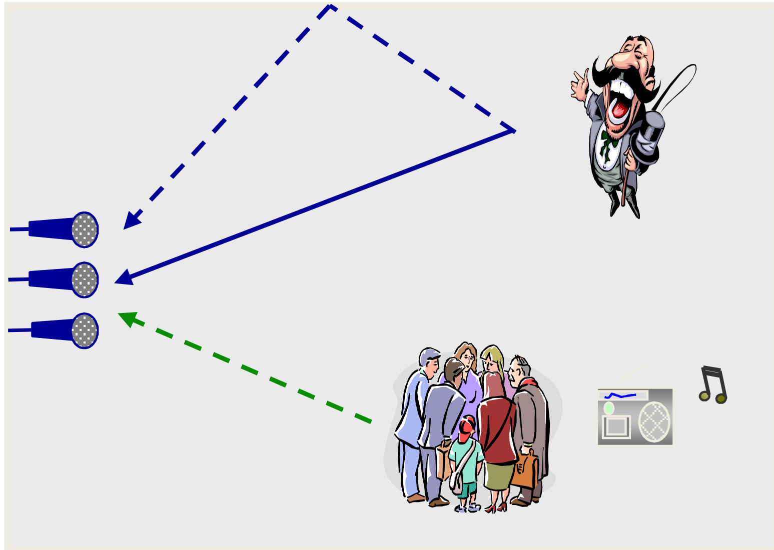
# 1. Speech enhancement

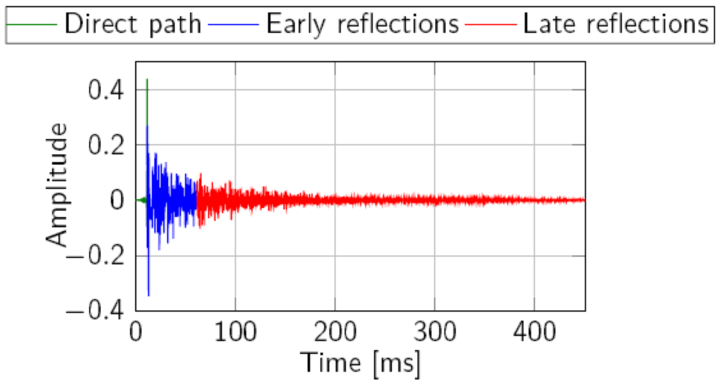
- **Binaural noise reduction and cue preservation**:
  - Perceptual evaluation of binaural MVDR beamformer with partial noise estimation, exploiting IC discrimination ability of auditory system
  - **Cue preservation improves spatial quality …
    at no degradation
    of speech intelligibility**
  - Current work:
    - Integration with computational acoustic scene analysis (CASA)
    - Extension with remote microphone



# 1. Speech enhancement

- **Dereverberation**:
  - Reverberation causes degradation of speech quality
  - Objective: blindly estimate clean speech signal from one or more reverberant microphone signals
  - Exploit **knowledge / statistical models of room acoustics and speech signals**



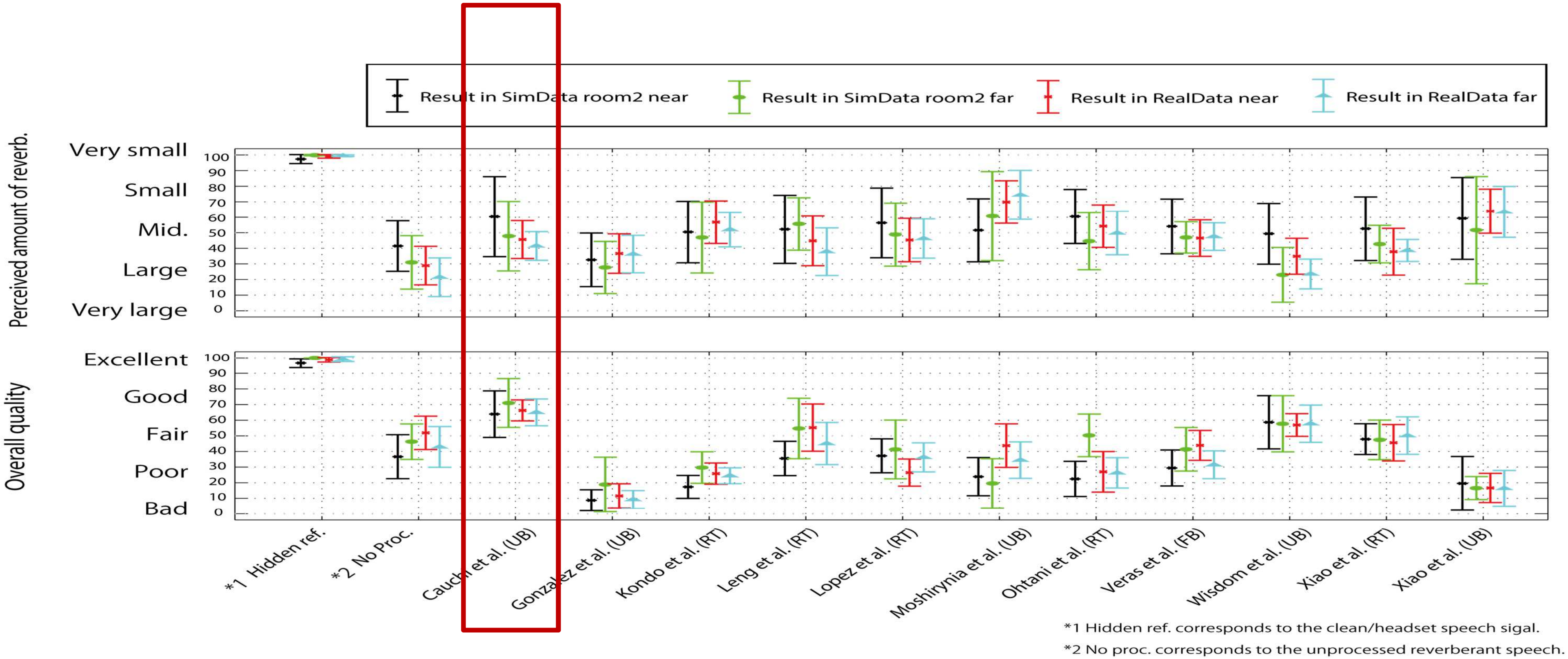



## Approaches

1. Single- and multi-microphone **spectral enhancement**
2. **Multi-channel linear prediction**: probabilistic estimation using statistical model of desired signal

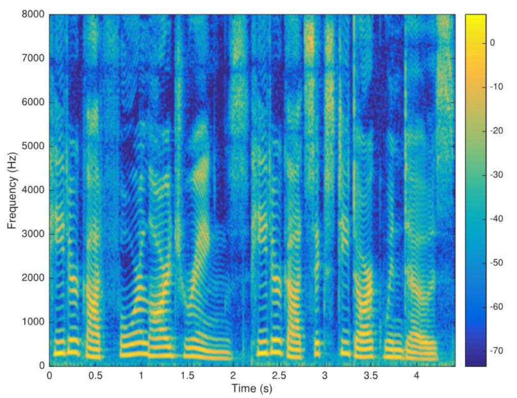
# 1. Speech enhancement

- **REVERB Challenge:** international competition for speech enhancement and ASR in reverberant environments

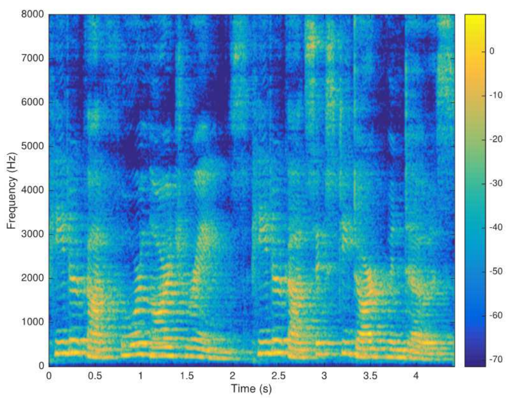


# 1. Speech enhancement

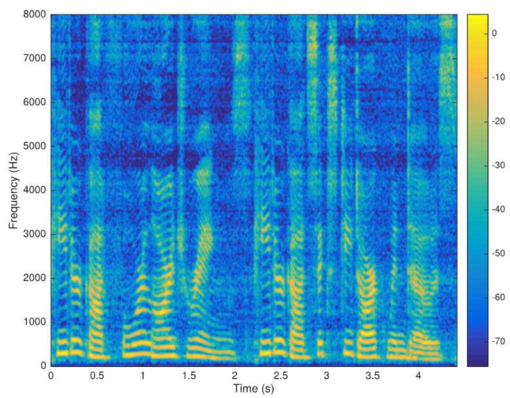
- **Extension** of single-microphone to multi-microphone techniques
  - Robust dereverberation using multi-channel equalization exploiting sparsity properties of clean speech
  - **Blind dereverberation based on multi-channel linear prediction** with sparse priors




Clean




Microphone

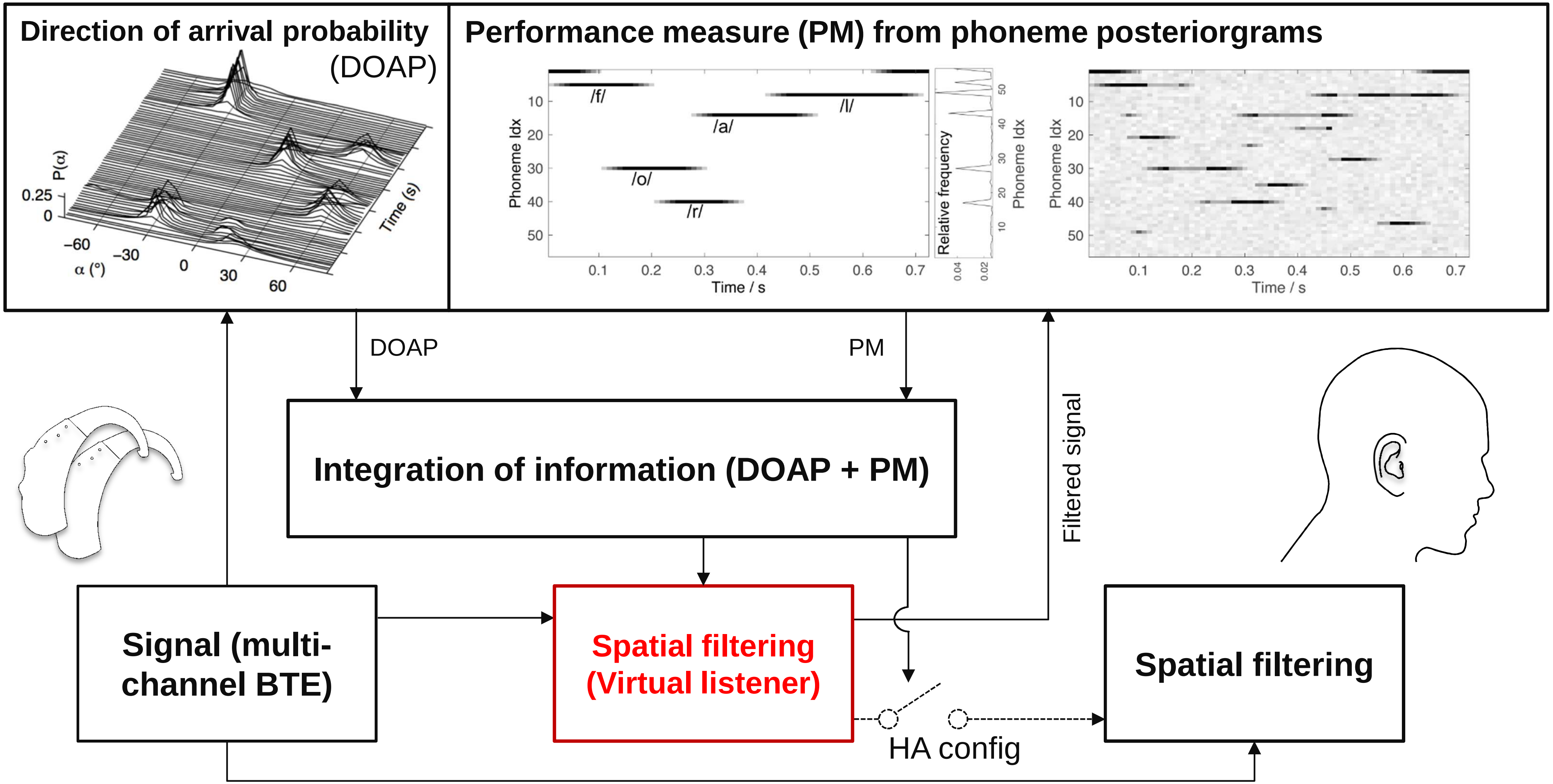



MCLP

| | PESQ | FWSSNR |
|---|---|---|
| Mic | 1.21 | 3.61 |
| MCLP | 2.40 | 7.92 |

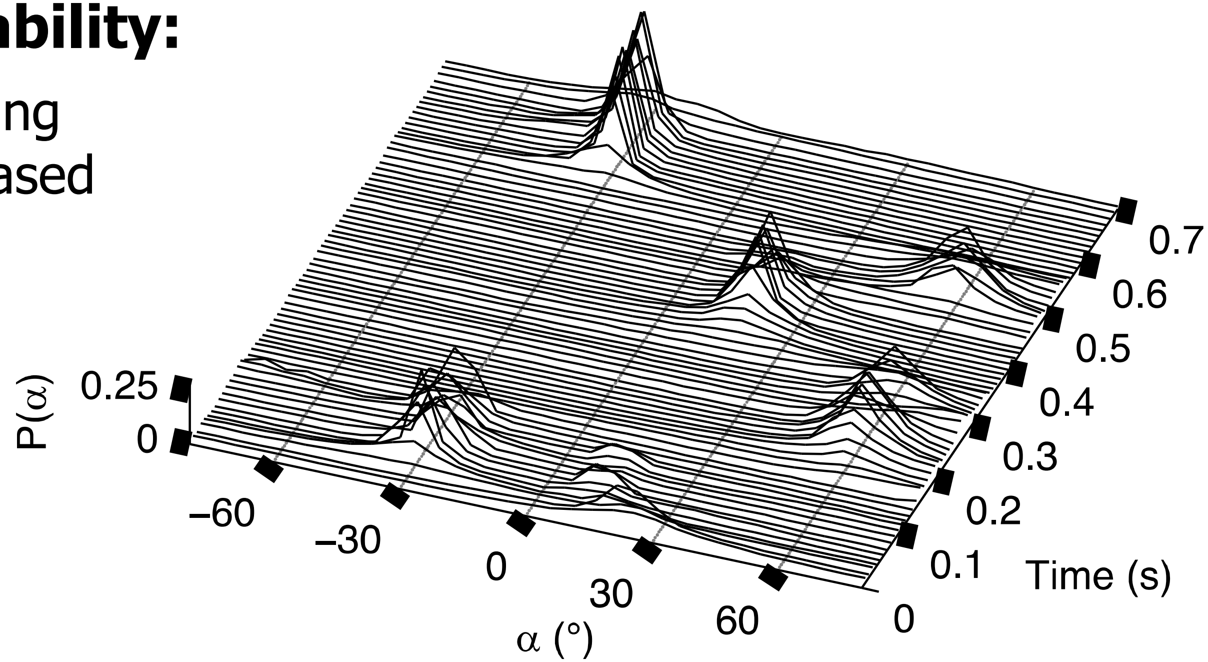
$T_{60} \approx$ 700ms, M=2 (BRIR), distance 4m, fs=16 kHz; STFT: 64ms (overlap 16ms); MCLP: $L_g$=30, $\tau$=2, p=0

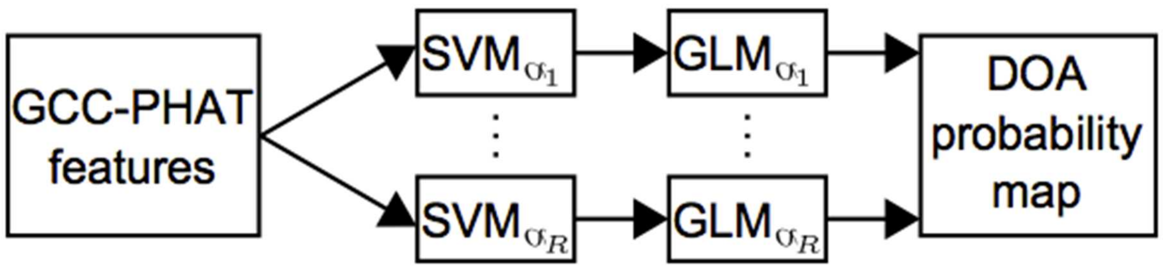
# 2. Speech control based on deep learning



# 2. Speech control based on deep learning

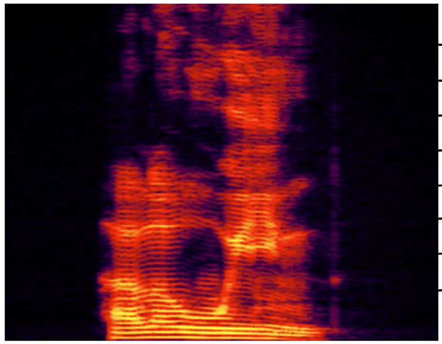
- **Direction of arrival probability:**
  - Discriminative approach using Support Vector Machines based on GCC-PHAT features
  - Trained on (anechoic) head-related data
  - Generalizes well to reverberant scenarios without knowledge of room acoustics



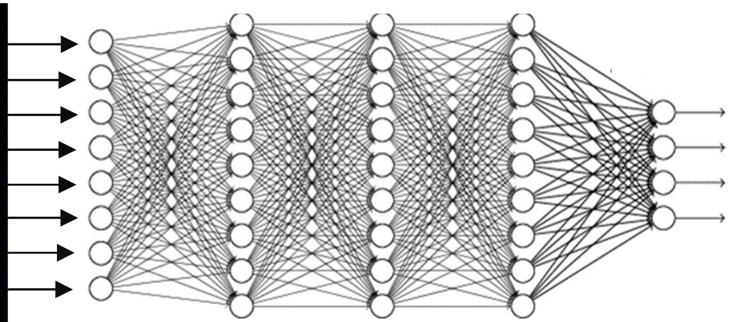



# 2. Speech control based on deep learning

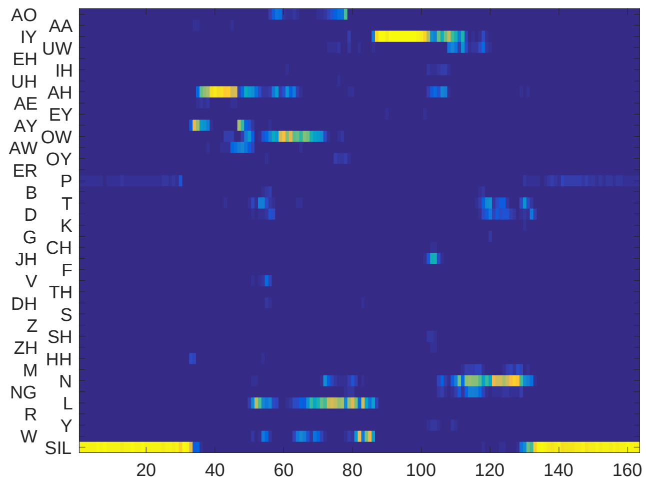
- **Speech probability performance measure:** posterior probabilities (phoneme posteriorgram) obtained from a Deep Neural Net (DNN)



"Halloween"



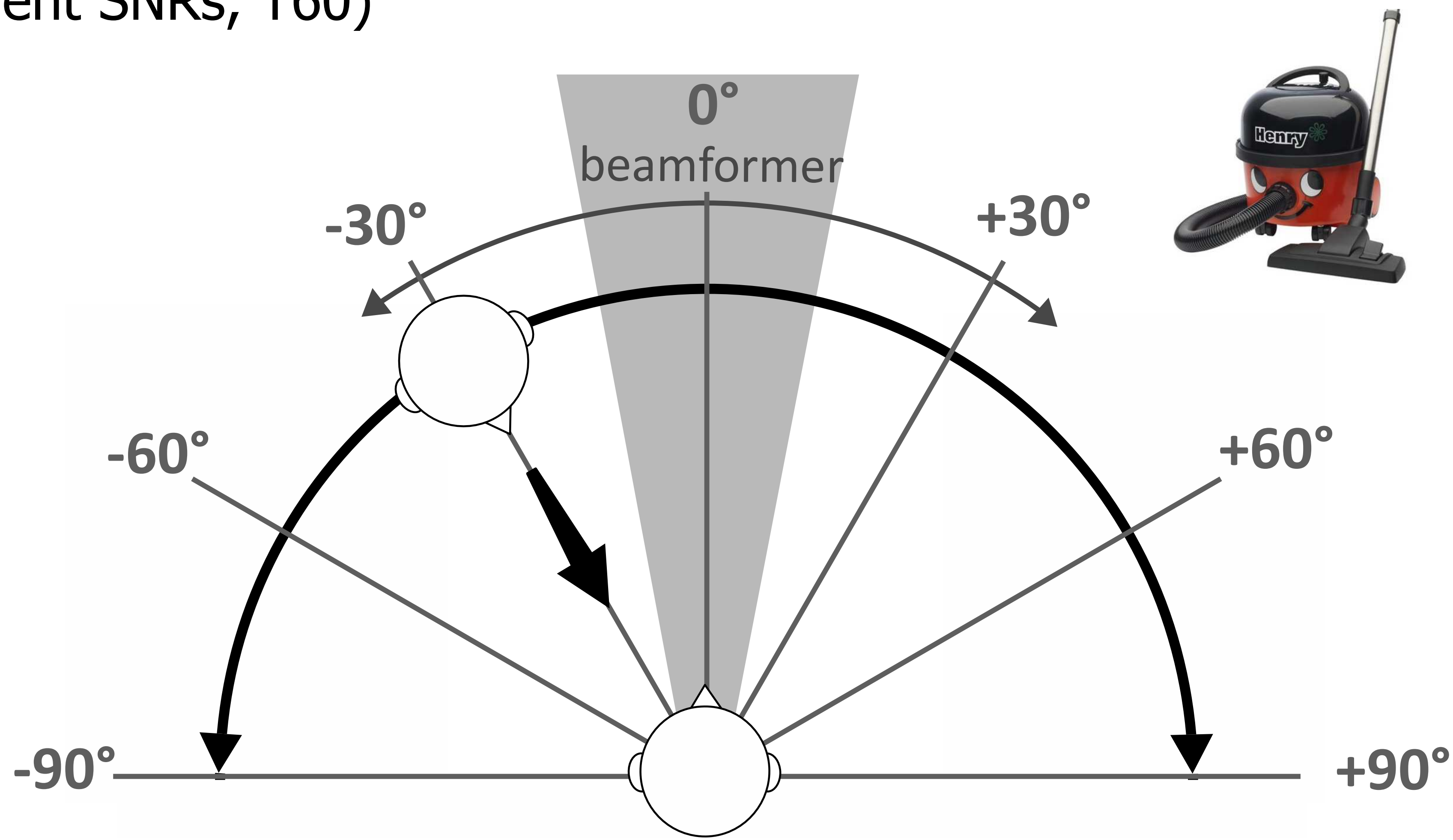
**Deep Neural Net**




Phoneme posteriorgram

# 2. Speech control based on deep learning

**Acoustic scenario:** speaker and localized noise source (different SNRs, T60)



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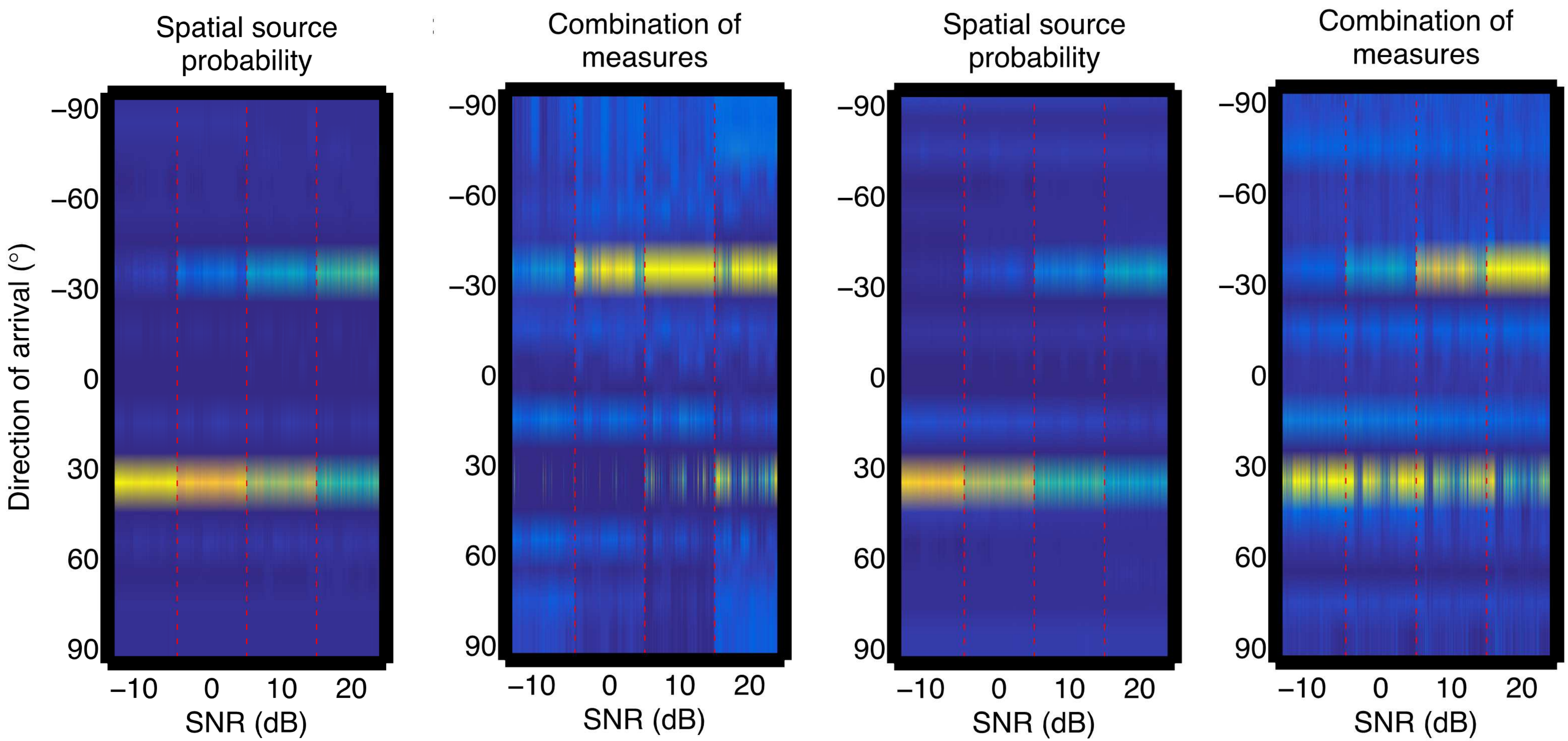
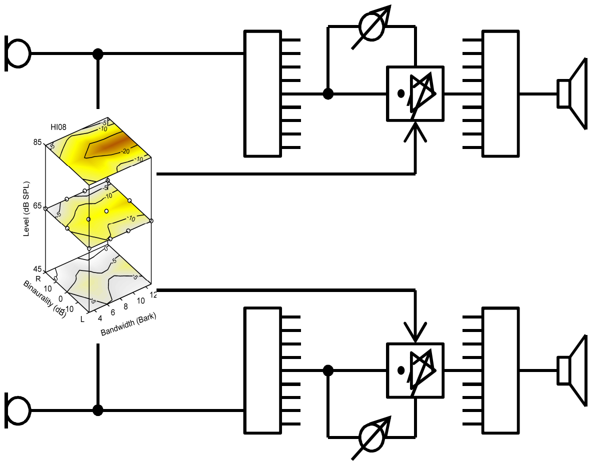


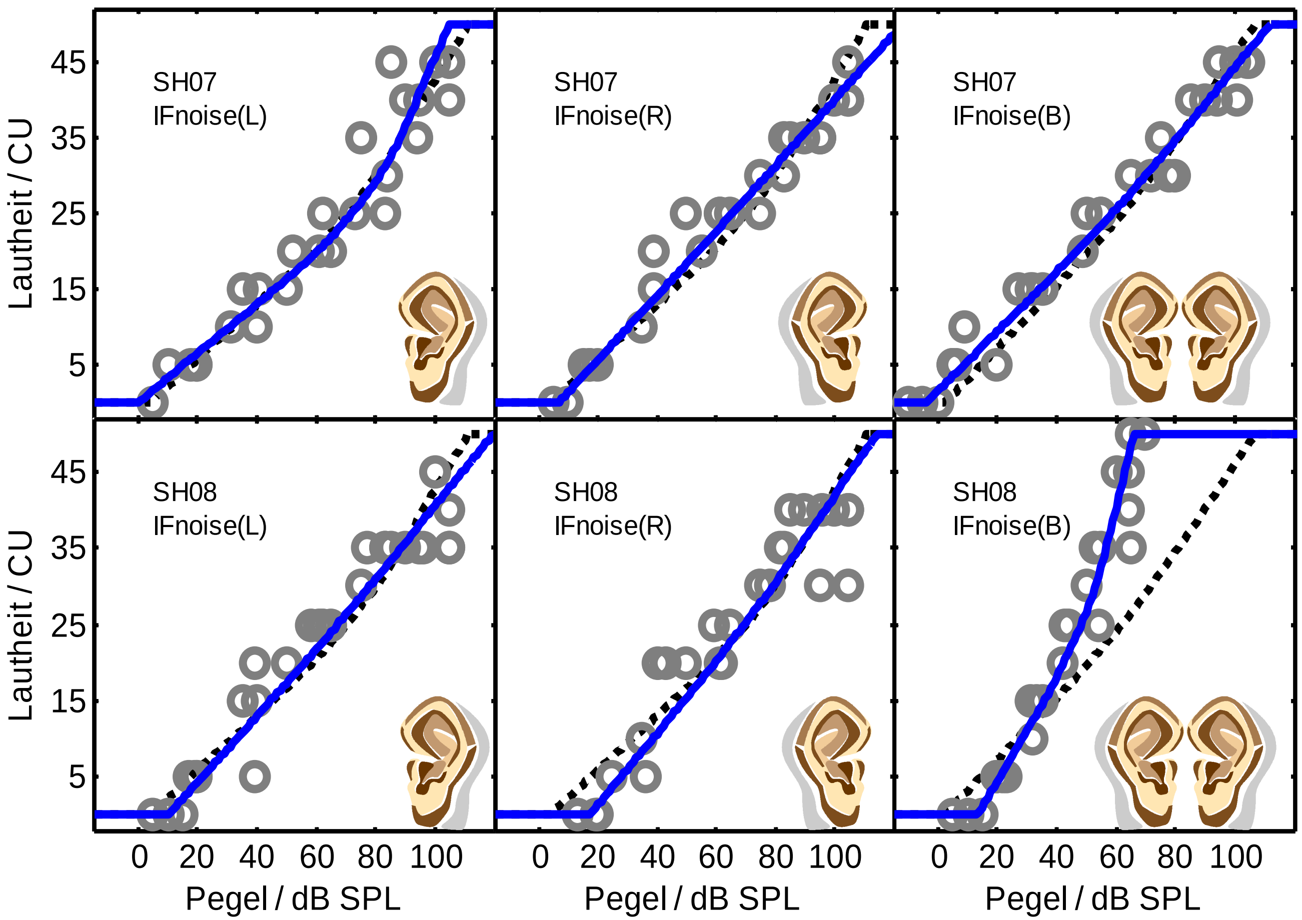

**Anechoic** **Office**

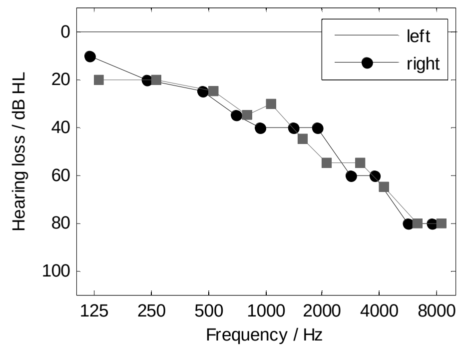
# 3. Binaural bandwidth-adaptive dynamic compression

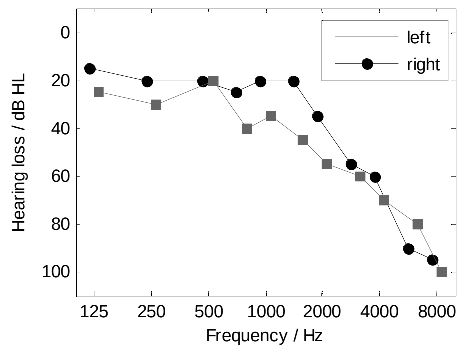
- Hearing aid fitting: discomfort at high loudness levels is a major complaint
- **Aim:** improve loudness restoration and fitting of hearing devices using dynamic range compression (DRC)
- **Recent finding:** increased binaural loudness summation for broadband binaural signals in HI listeners ("excess loudness sensitivity")



# 3. Binaural bandwidth-adaptive dynamic compression



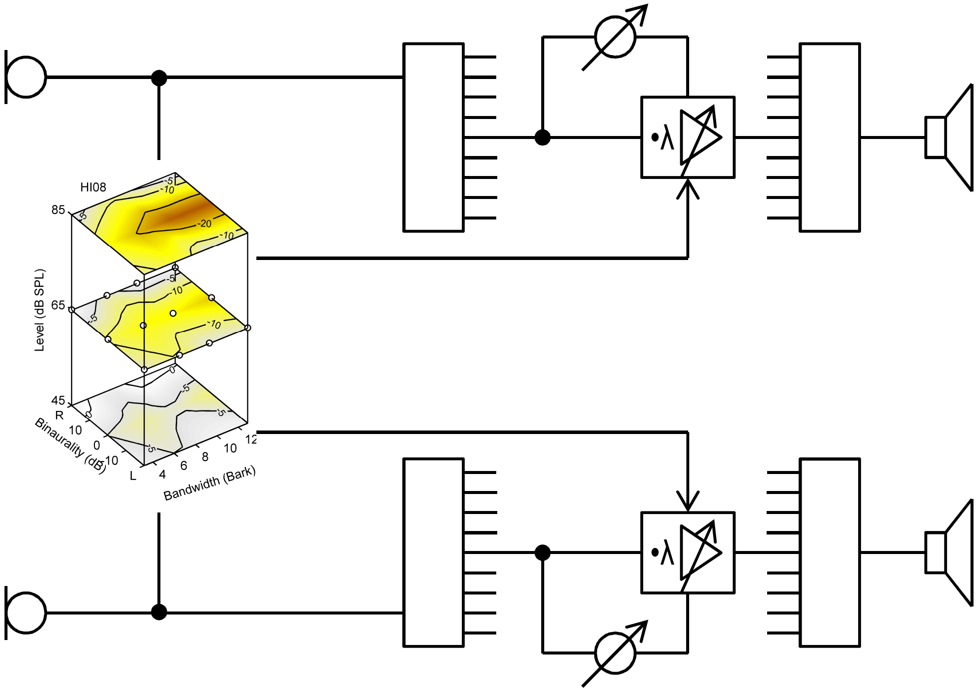



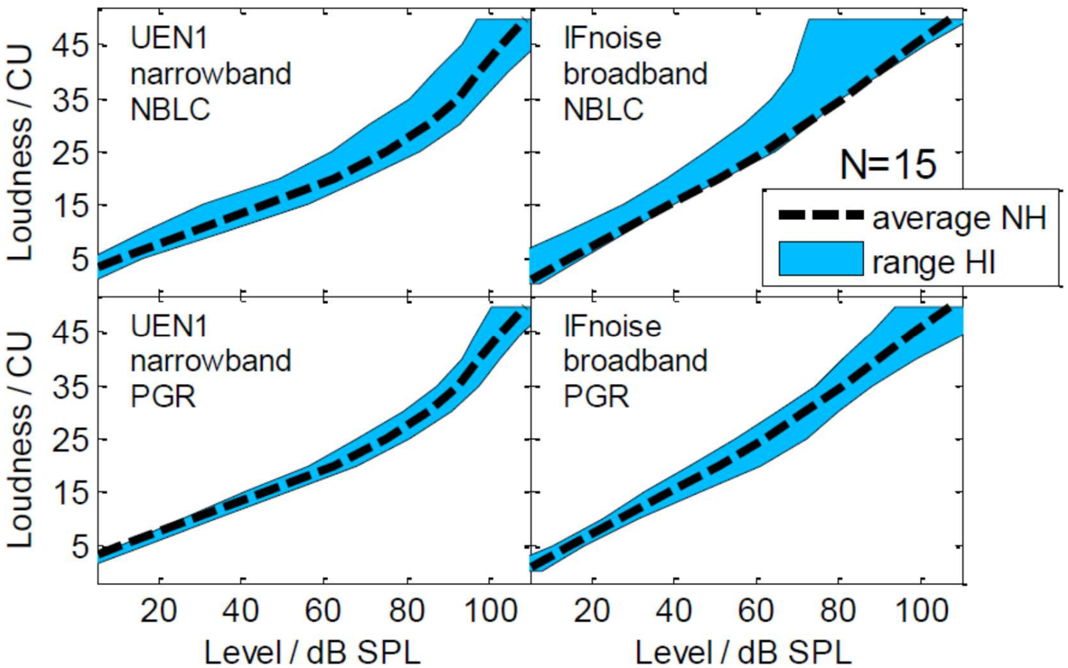




- Increased binaural broadband loudness perception can not be seen in the audiogram

# 3. Binaural bandwidth-adaptive dynamic compression

- **New algorithm:** binaural bandwidth-adaptive dynamic compression (BBDC) to normalize the loudness perception in HI listeners for narrow- and broadband signals for monaural and binaural presentation
- **New fitting strategy**, accounting for binaural loudness perception using loudness scaling measurements

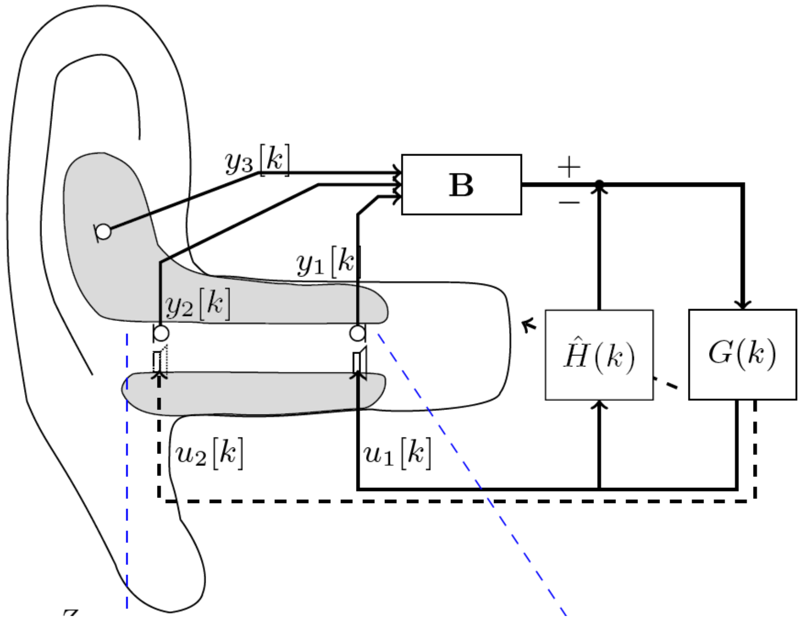


# Hardware / Technology

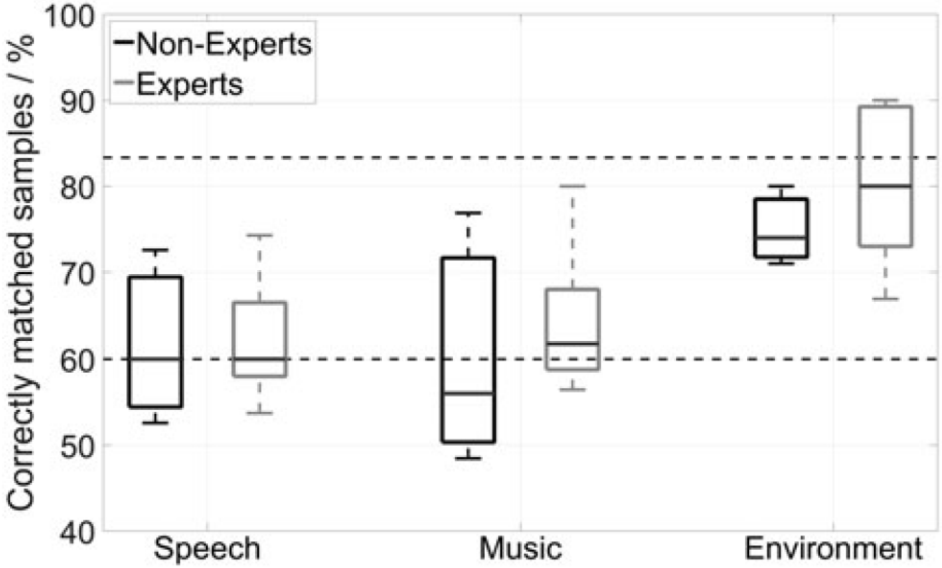
# 1. Acoustically transparent earpiece

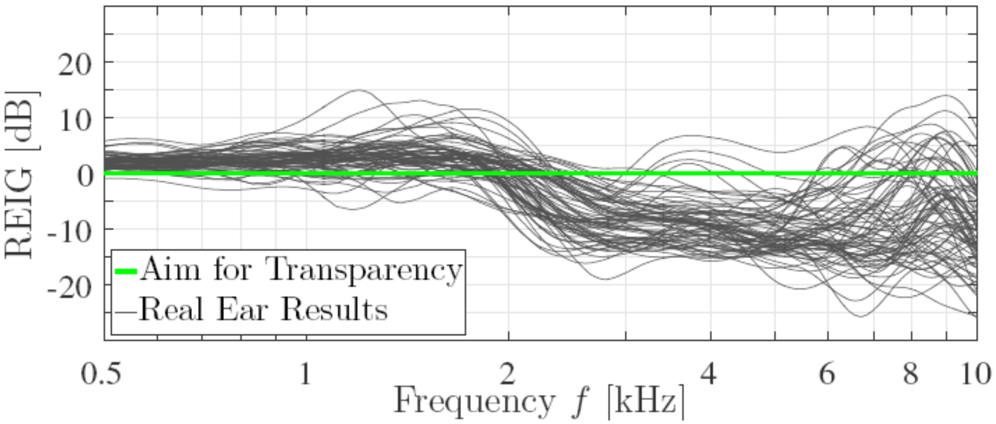
- **Objective:** develop a hearing device that is *acoustically transparent*, i.e. allows hearing comparable to the open ear while being capable of providing a desired sound enhancement at the eardrum
- Applications: **hearing aids** as well as in **assistive listening devices**
- Individualized silicon earmould with **core containing 2 receivers and 2 microphones** (+ concha microphone)
- Integration with EEG electrodes



# 1. Acoustically transparent earpiece

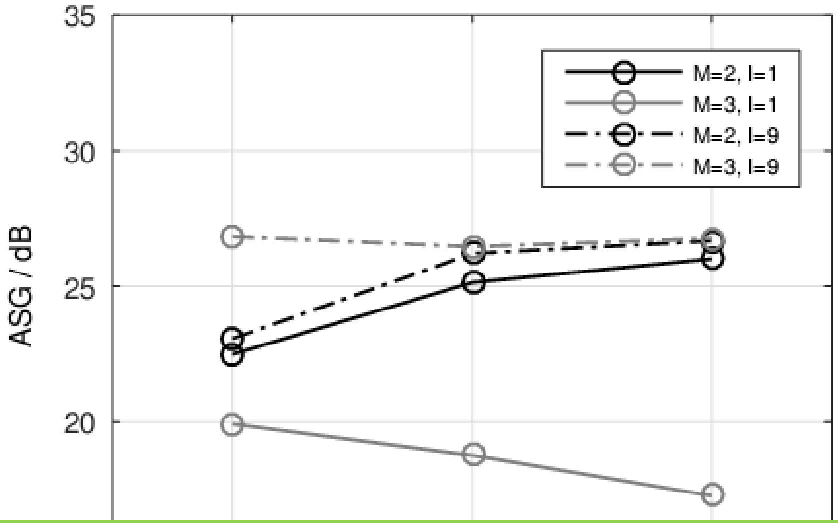
- **Sound equalization:** optimize equalization filter such that superposition of direct sound and reproduced sound at eardrum is (physically or perceptually) equal to open ear condition
- **In-situ individualized calibration**
- **Results:** psycho-acoustic experiments + physical evaluation (REIG)



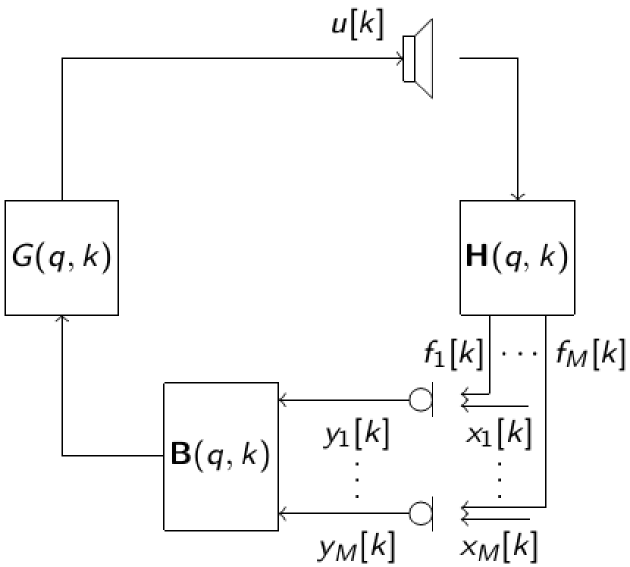
# 1. Acoustically transparent earpiece

- **Multi-Loudspeaker Multi-Microphone Feedback Cancellation**
  - Reduce acoustic feedback in the vent microphones by steering a (robust) spatial null towards the receiver



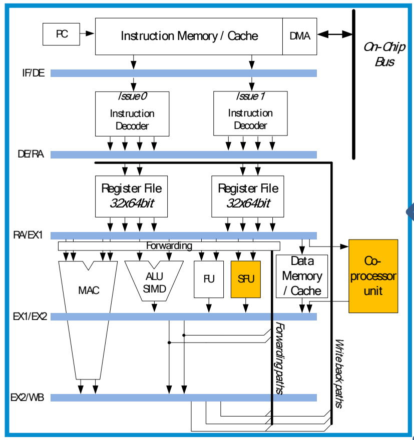

Robustly achieve an average ASG of >25dB

$L_B$

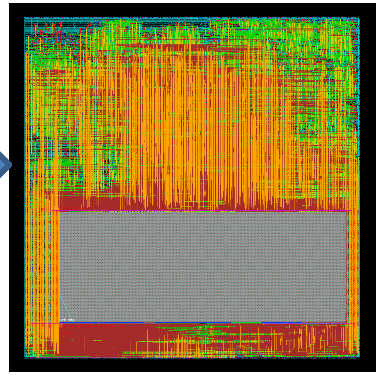


# 2. Ultra low-power processor architecture

- **System-on-Chip** for hearing aids
- Ultra low-power processor architecture (KAVUAKA) optimized for **real-time processing of complex audio algorithms** (CASA algorithm and beamforming algorithms as case study)




KAVUAKA

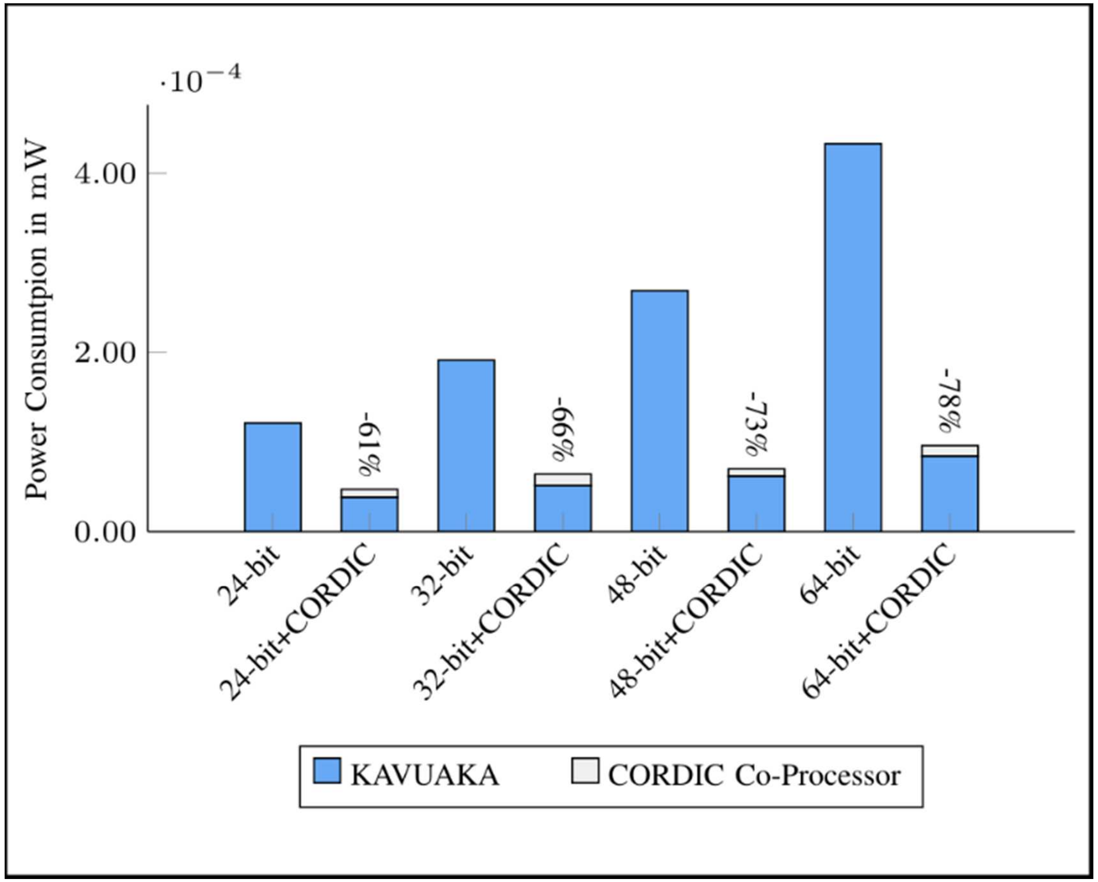


KAVUAKA in TSMC 40nm

Algorithms

# 2. Ultra low-power processor architecture

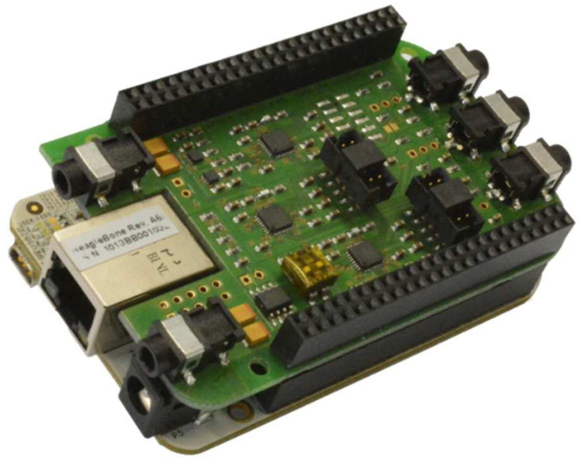
## Power Evaluation of Hearing Aid ASIP Optimizations

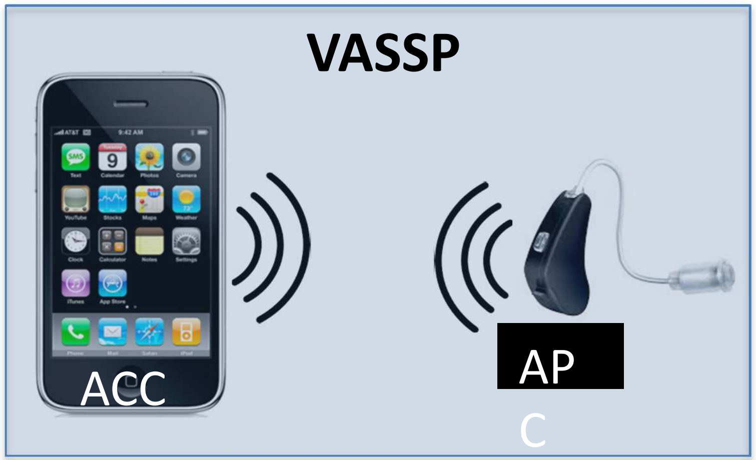



Evaluation setup: TMSC 40nm netlist, Prime Time Power simulations of beamformer algorithms

# 3. Demonstrator platforms

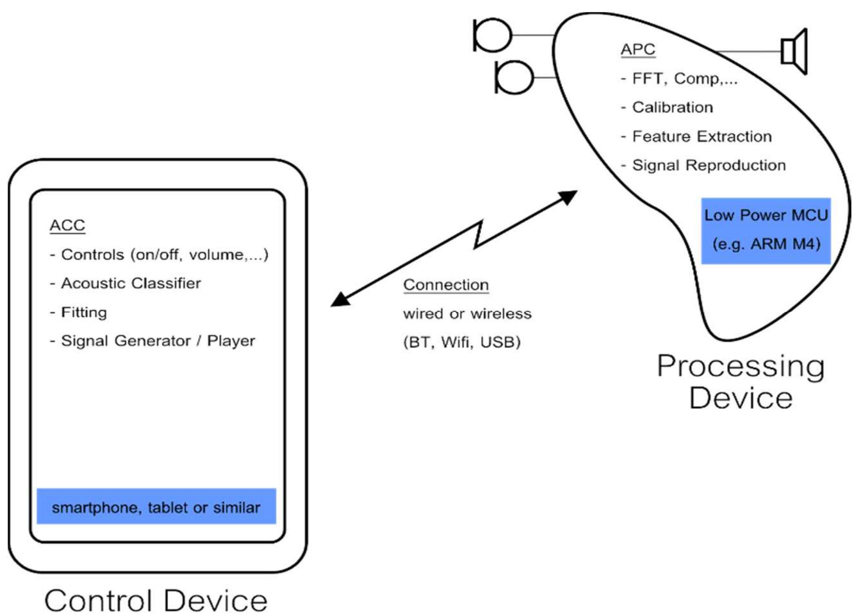
- **Master-Hearing-Aid (MHA)** software development system
- **Scalable hardware platform for mobile testing in the field**
  - PC / Notebooks
  - ARM-system (BeagleBone) with multi-channel AD/DA
  - Accelerated chip-based system (KAVUAKA)
  - Smartphone (iPhone/iPod)





# 3. Demonstrator platforms

- **Versatile Audio Signal Processing Platform – VASSP**
  - Allows hearing aid processing algorithms and split across smartphones and satellite devices
  - Split into *Audio Control Core* (ACC; runs on smartphone) and *Audio Processing Core* (APC; runs on satellite device)

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