Multiresolution Approaches to Fast Time Convolution for Hearing Aids

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Problem Statement

Hearing loss can be described as having a threshold for sounds above the standard speech intensity level.

- Hearing loss typically occurs at higher frequencies
- Hearing level measurements are performed at each octave: 125/250/500/1k/2k/4k/8k Hz

Hearing aids aim to boost sound levels at “weak” frequencies for a given individual such that the sound level meets the individual’s threshold[3].

- Low power and small form factor
- Low computational complexity (CPU and memory)
- Minimize latency in sound delivery

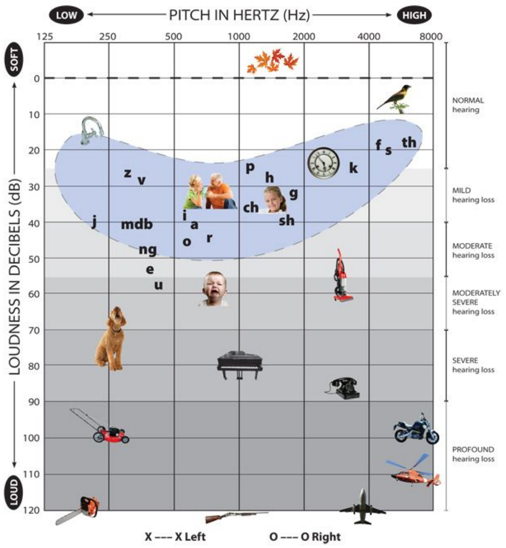


Fig. 1. Familiar Sounds Audiogram [2]

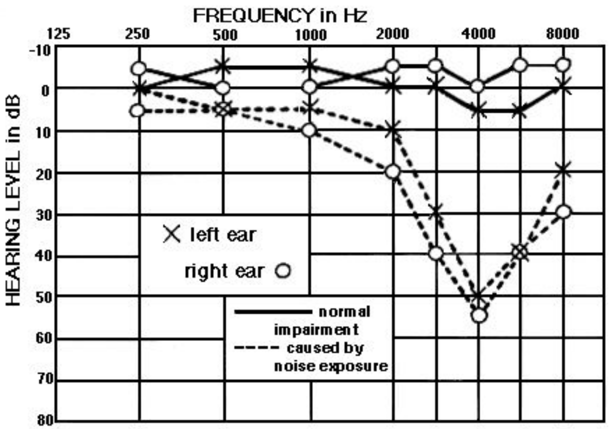


Fig. 2. Audiogram, Normal v. Impaired [1]

Current Approaches

Digital filter banks allow for the amplification at sound levels at specific frequencies to match an individual's audiogram.

- Uniform fixed bandwidth filter bank [Lunner, 4]
  - Successfully amplifies weak sound levels while maintaining low power consumption
  - Latent sound delivery and poor audiogram matching accuracy
- Variable bandwidth filter bank [Ito, 5]
  - Subfilters implemented to closely match the audiogram
  - Filter bank composed of high-pass, low-pass, and band-pass filters was not linear phase causing difficulties in the audio magnitude matching at various frequencies
- Nonuniform filter bank [Onat, 6; Lian, 7]
  - Increased subband resolution supports higher accuracy in audiogram matching from narrow bands at high frequencies
  - Increased subband distribution increases delay in delivered acoustic signal

The focal point of current research lies in the resolution-latency trade off of the filter bank design.

Current Solution

- FIR filter banks derived from two linear phase half-band prototype filters $H(z)$ and $F_m(z)$ presented in Lian et al.
- Upsampled $H(z)$ and $F_m(z)$ and cross-subtracted to generate 8 sub-bands
- Midpoint interpolation vs. MMSE solution explored for optimizing sub-band gain.

	Order	Transition BW
H	18	0.25*Fs
F	38	0.125*Fs

Table 1. Final Filter Orders

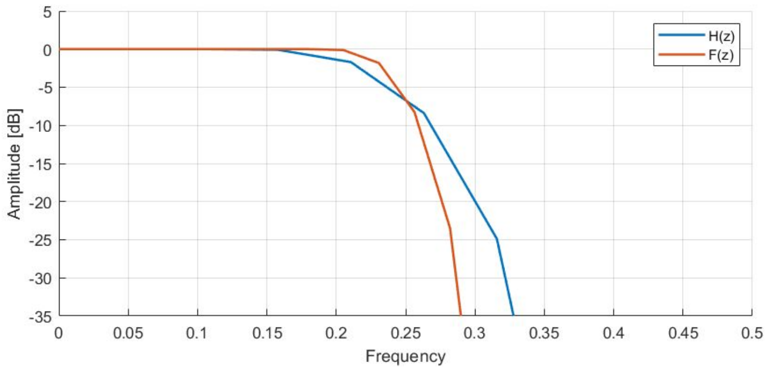


Fig. 3. Half-band Prototype Filters (H,F)

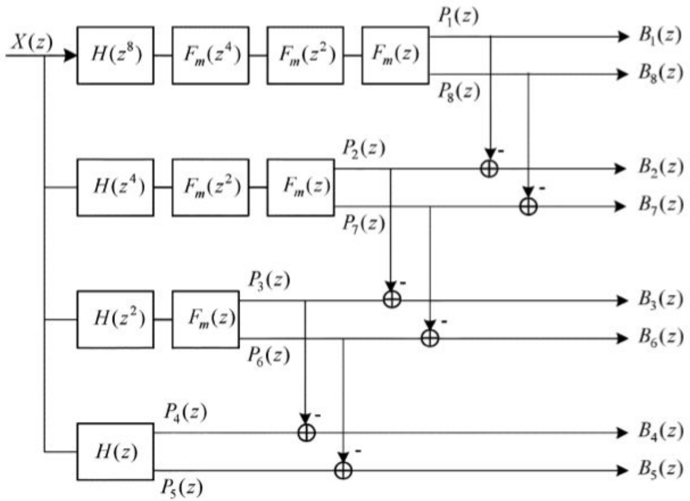
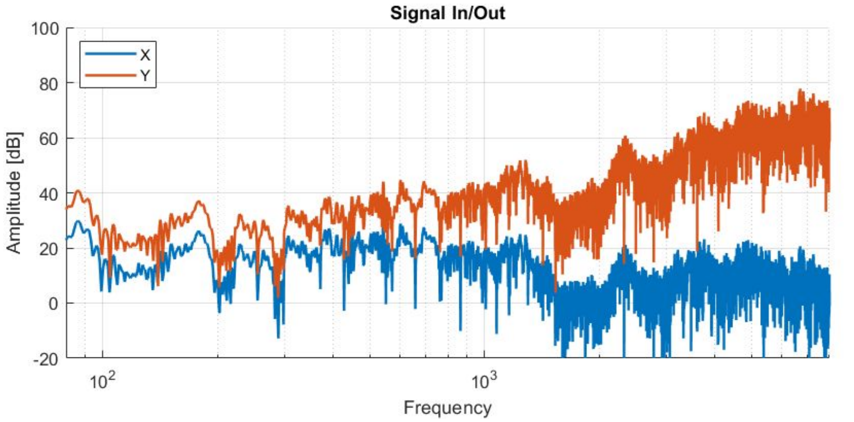
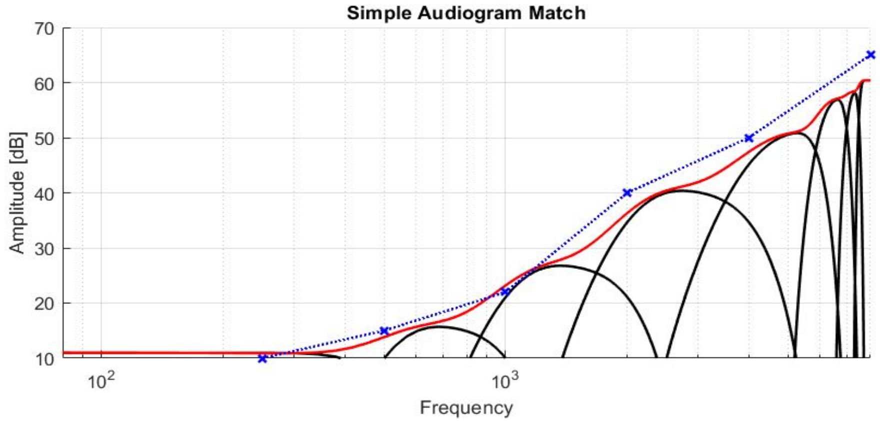
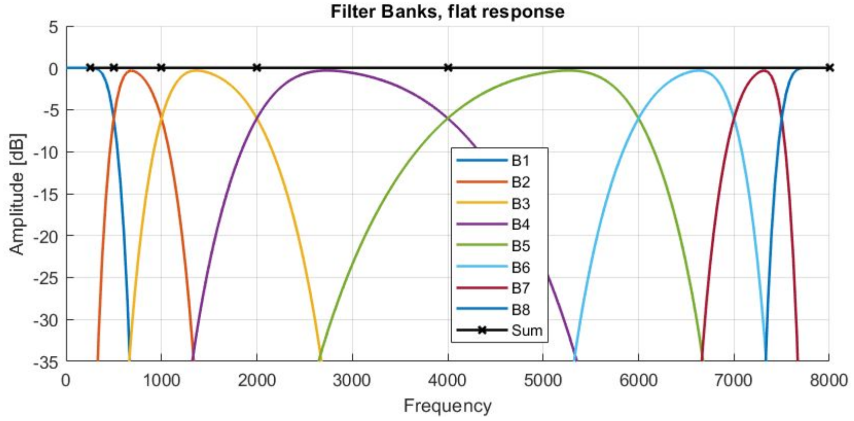


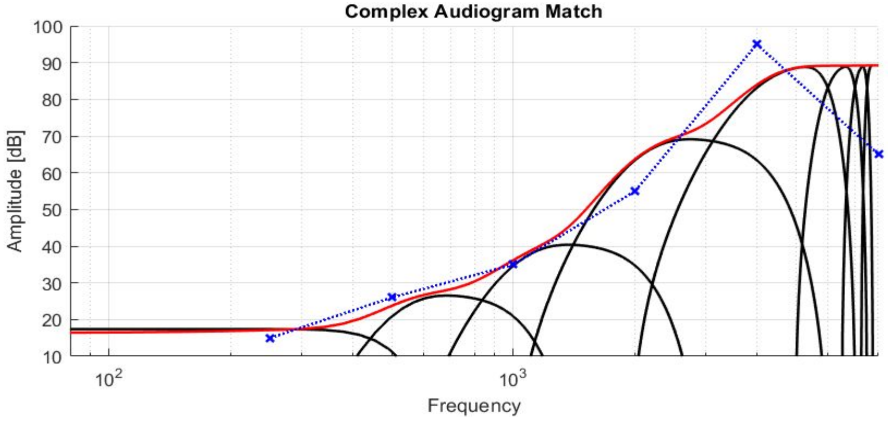
Fig. 4. Filter bank structure

Subband Decomposition



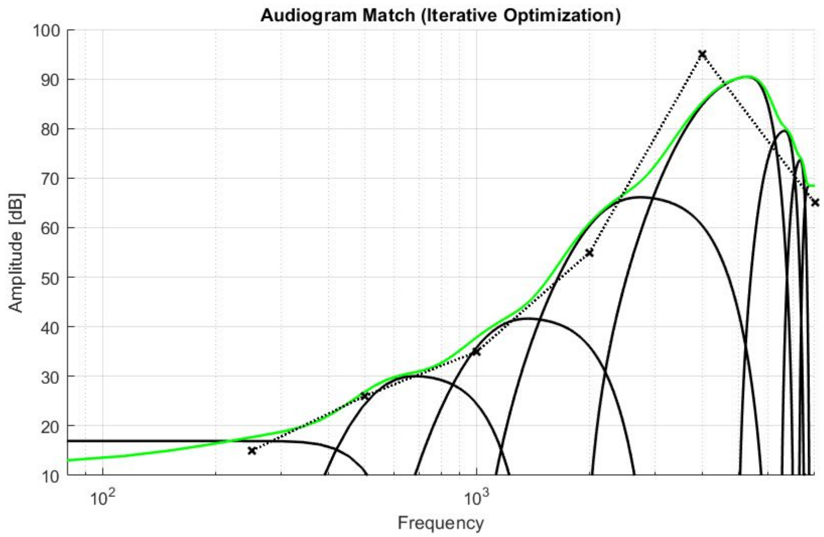


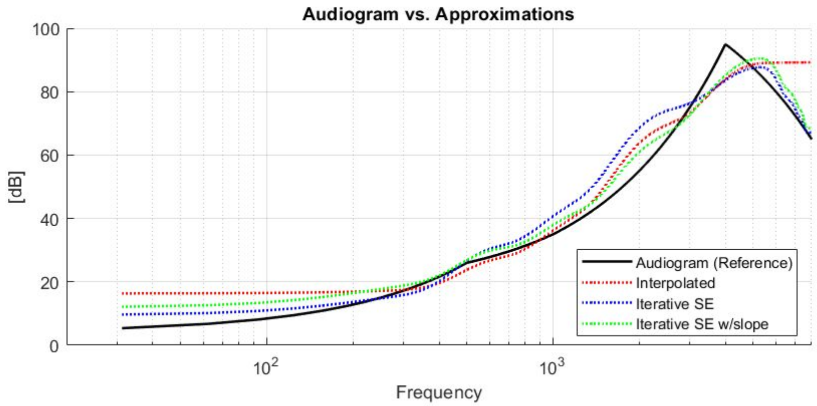


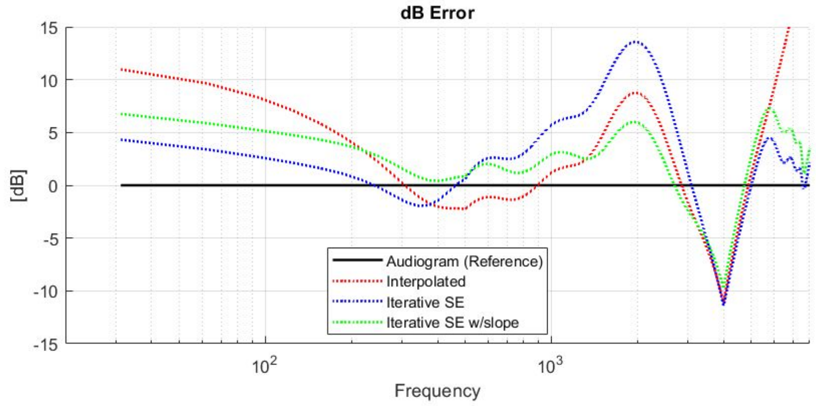


Project Results

- **Linear Interpolation**
  - unable to follow peaks and inflections
- **Iterative MMSE Optimization + Decay:**
  - able to follow peaks
  - poor performance on audiogram slopes
- **Iterative MMSE + Decay + Slope Estimation:**
  - Improves performance with slope while following peaks



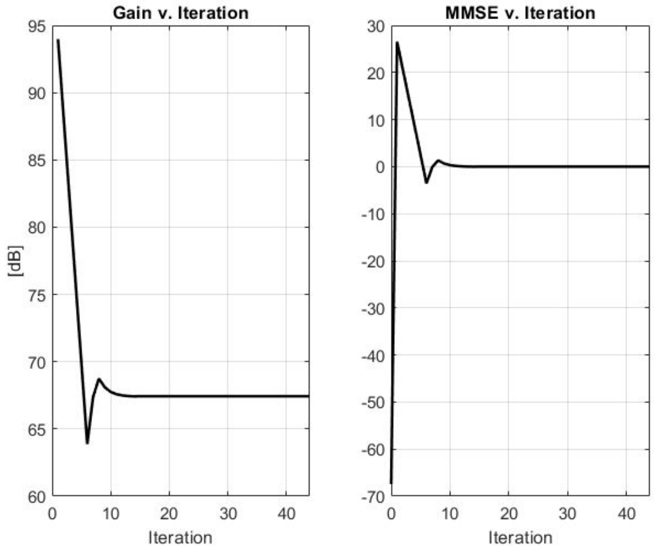


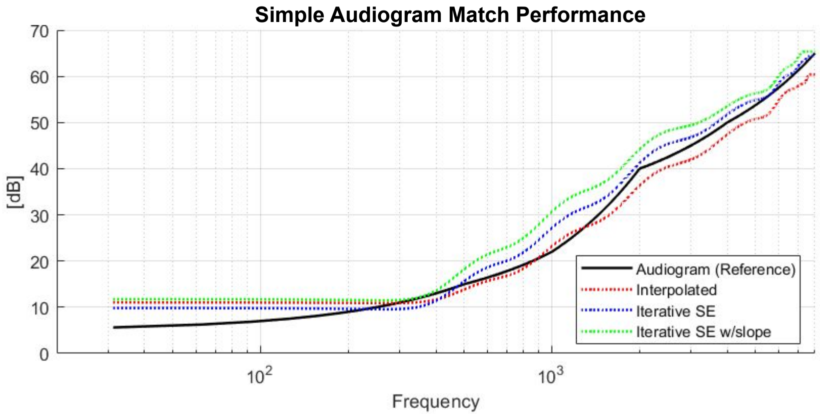


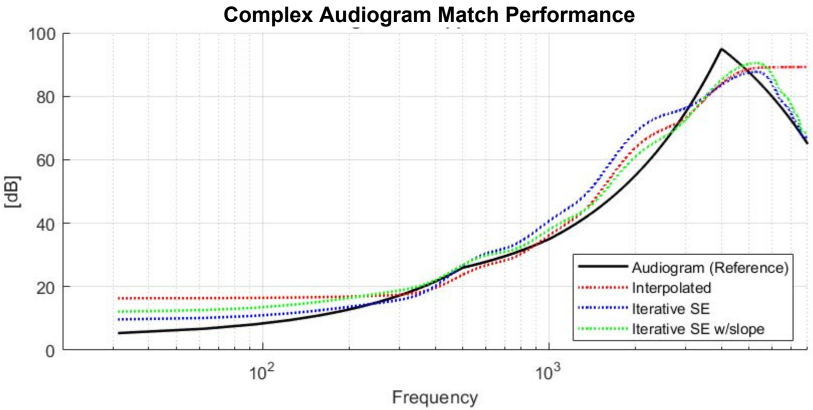
Summary

- Fixed-bandwidth non-uniform filter banks allow for more control at higher frequencies while maintaining low complexity
    - **Linear interpolation:** sufficient for generic prescriptions (e.g. presbycusis)
    - **Iterative methods:** fine-tuning and personalization

- Can apply more methods of statistical regression and training algorithms to fitting process for future filter bank topologies







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