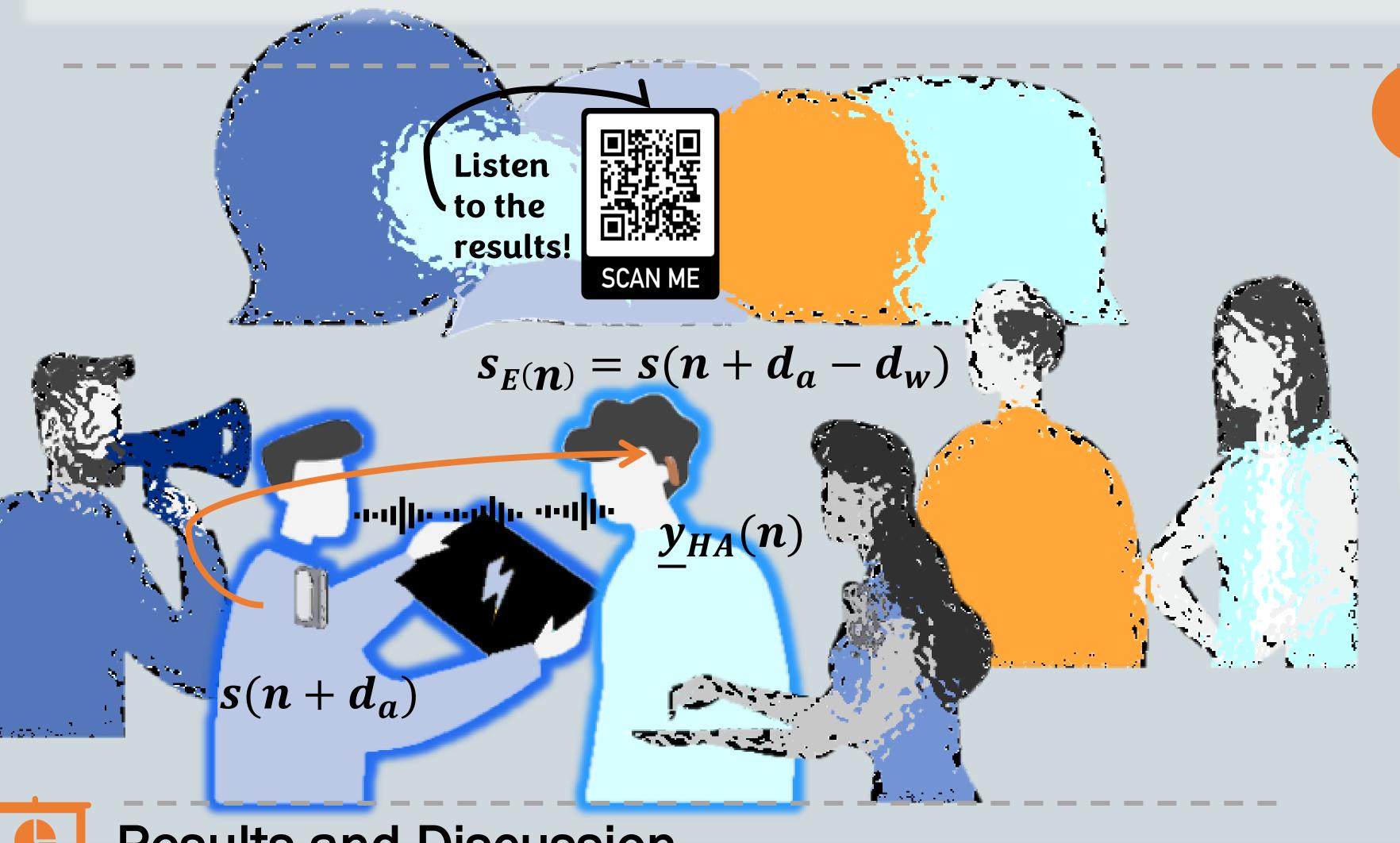
A LINEAR MMSE FILTER USING DELAYED REMOTE MICROPHONE SIGNALS FOR SPEECH ENHANCEMENT IN HEARING AID APPLICATIONS

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- Using remote microphones, placed close to the target source, enhances the noise reduction in hearing assistive devices (HADs).
- Existing literature ignores the time difference of arrival (TDOA) between the acoustically and wirelessly transmitted signals, which leads to poor noise reduction performance, when applied in practice [1],[2].



- To estimate, $\widehat{x}_{ref}(n) = f(y_{HA}(n), s(n-d))$
- Use multiple past frames of the remote microphone signal, with the multi-channel local microphone signals.

 $s(n+d_a)$: anechoic signal

: remote microphone signal received by the HA $S_E(n)$

 $\underline{\widehat{w}}_{EMWF} = \underset{w}{\operatorname{argmin}} E[||\underline{w}^{H}\underline{Y} - X_{ref}||^{2}]$

 \widehat{X}_{EMWF}

: noisy HA microphone signal vector $y_{HA}(n)$

 $d = d_a - d_w : TDOA$

Noisy + clean

statistics

estimation

 $\widehat{\underline{\pmb{C}}}_{YY}$, $\widehat{\underline{\pmb{C}}}_{XX}$

 $\underline{Y}(k,l)$

: acoustic propagation delay

: wireless latency : clock sample at the HA n

 $|\widehat{\underline{C}}_{YY,HA}|, |\widehat{\underline{C}}_{XX,HA}| \Rightarrow |\widehat{\underline{w}}_{MWF-HA}| = |\overset{argmin}{w}E[||\underline{w}^H\underline{Y}_{HA} - X_{ref}||^2]$

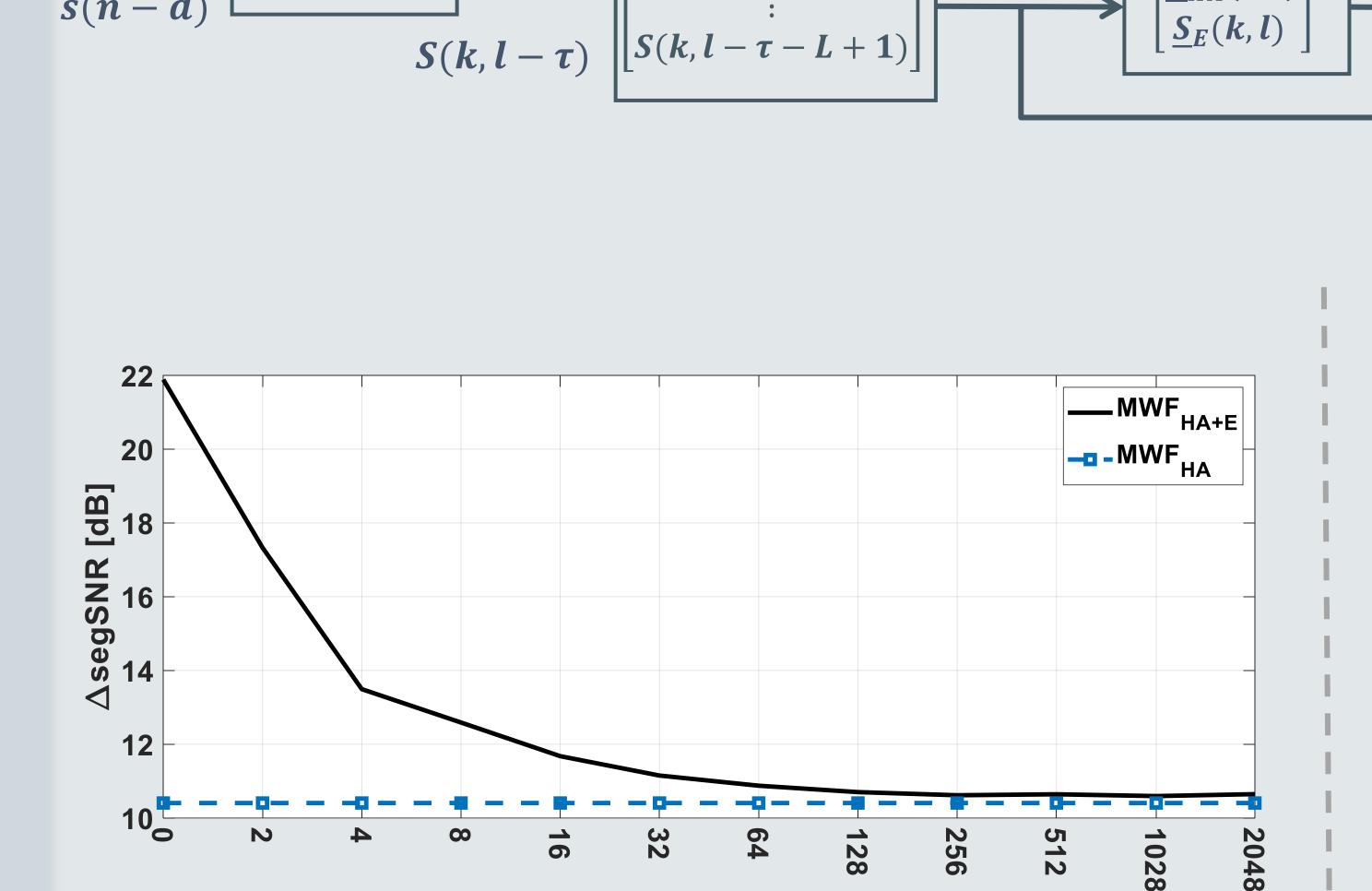
Results and Discussion

STFT

 $\underline{Y}_{HA}(k, l)$

 $y_{HA}(n)$

s(n-d)



 $S(k, l-\tau)$

 $\underline{S}_E(k,l)$

 $[\underline{Y}_{HA}(k,l)]$

 $S_E(k, l)$

Comparing the performance of MWF using HA and remote microphone signals as a function of TDOA (τ).

TDOA [ms]

 $\Rightarrow \widehat{X}_{MFWF-E}$ $\underline{\widehat{w}}_{MFWF-E} = \underset{w}{\operatorname{argmin}} E[||\underline{w}^{H}\underline{Y}_{E} - X_{ref}||^{2}]$ $\widehat{\underline{C}}_{YY,E}$, $\widehat{\underline{C}}_{XX,E}$ It is a linear combination of $\frac{\widehat{w}_{EMWF}}{[\beta.\widehat{\underline{w}}_{MFWF-E}]_{(M+L)\times 1}}$ existing algorithms! $\alpha, \beta = f(SNR_{MFWF-E}, SNR_{MWF-HA})$ 30 **25** segSNR [dB] 10 -20 -10 10 20 SNR_{in, HA} [dB] segSNR as a function of SNRin, HA for the MWF, EMWF and MFWF, at $\tau = 16$ ms and L = 5, estimated for speech signals.

Conclusion

- Resulting beamformer is a simple linear combination of existing multi-channel and multi-frame algorithms.
- Noise reduction and speech intelligibility improves for positive TDOA≤ 16 ms.

References

- [1] Bertrand, Alexander, and Marc Moonen, "Robust Distributed Noise Reduction in Hearing Aids with External Acoustic Sensor Nodes." EURASIP Journal on Advances in Signal Processing 2009, no. 1 (December 2009).
- [2] Gößling, Nico, Daniel Marquardt, and Simon Doclo, "Performance Analysis of the Extended Binaural MVDR Beamformer with Partial Noise Estimation in a Homogeneous Noise Field." In Hands-Free Speech Communications and Microphone Arrays (HSCMA), 1–5. San Francisco, CA: IEEE, 2017.

