Adaptive Feedback Cancellation in Hearing Aids using a Sinusoidal near-end Signal Model

Kim Ngo[1], Toon van Waterschoot[1], Mads Græsbøll Christensen[2], Marc Moonen[1], Søren Holdt Jensen[2] and Jan Wouters[3]

[1]Katholieke Universiteit Leuven, ESAT-SCD, Kasteelpark Arenberg 10, B-3001 Leuven, Belgium
[2]Aalborg University, Dept. Electronic Systems, Niels Jernes Vej 12, DK-9220 Aalborg, Denmark
[3]Katholieke Universiteit Leuven, ExpORL, O. & N2, Herestraat 49/721, B-3000 Leuven, Belgium

Introduction (1)

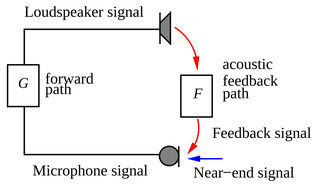
Hearing impairment is becoming more common and can be caused by many reasons, some of them are listed below:

- Daily exposure to excessive noise in the work environment (construction site, factory etc.).
- From listening to loud music (MP3 players, iPod, concerts, night clubs etc.)
- Age-related (high-frequency hearing loss).



Acoustic feedback

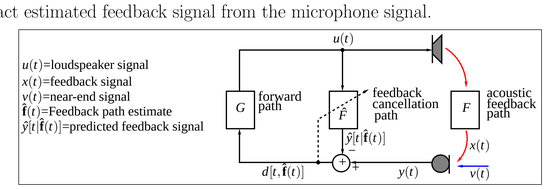
- Acoustic feedback is a well-known problem in hearing aids, which is caused by the undesired acoustic coupling between the loudspeaker and the microphone.
- Acoustic feedback limits the maximum amplification that can be used in the hearing aid without making it unstable.
- In many cases this maximum amplification is too small to compensate for the hearing loss.



Adaptive Feedback Cancellation (2)

Concept:

- Adaptively model the feedback path and estimate the feedback signal.
- Subtract estimated feedback signal from the microphone signal.



Microphone signal: $y(t) = v(t) + x(t) = v(t) + F(q,t)u(t)$

Feedback-compensated signal: $d(t) = v(t) + [F(q,t) - \hat{F}(q,t)]u(t).$

Problem:

- Standard adaptive filtering results in a biased solution.
- Correlation between near-end signal and loudspeaker signal caused by closed signal loop.

Current solutions:

- Prediction error method (PEM)-based AFC with linear prediction (LP) model [2].
- PEM-based AFC with cascaded near-end signal model [1].

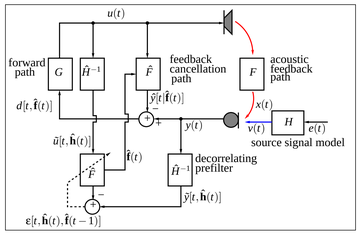
Proposed solution:

- PEM-based AFC with cascaded near-end signal model using pitch estimation.

Prediction Error Method based AFC (3)

Concept:

- Reduce the correlation between the near-end signal and the loudspeaker signal
- Prefiltering of loudspeaker and microphone signals with inverse near-end signal model



- Single all-pole model (Short-term predictor) fails to remove the periodicity
- A cascade of near-end signal models removes the coloring and periodicity
- Cascade of a constrained pole-zero LP (CPZLP) and a LP model [1].

Cascaded near-end signal model: $H(q,t) = \dfrac{B(q,t)}{A(q,t)}\dfrac{1}{C(q,t)}$

Prediction error: $\varepsilon[t,\boldsymbol{\xi}(t)] = H^{-1}(q,t)[y(t) - F(q,t)u(t)]$

Minimize prediction error: $\min_{\boldsymbol{\xi}(t)} = \dfrac{1}{2N}\sum_{k=1}^{t}\varepsilon^2[k,\boldsymbol{\xi}(t)]$

Sinusoidal near-end signal model (4)

- A typical LP model can replaced by a sinusoidal near-end signal model (CPZLP) [3].

Sinusoidal model: $d(t) = \sum_{n=1}^{P} A_n \cos(\omega_n t + \phi_n) + r(t), \quad t = 1, ..., M$

CPZLP model: $d(t) = \left(\prod_{n=1}^{P} \dfrac{1 - 2\rho\cos\omega_n z^{-1} + \rho^2 z^{-2}}{1 - 2\cos\omega_n z^{-1} + z^{-2}}\right) e(t)$

Prediction error filter output: $e(t,\boldsymbol{\omega}) = \left(\prod_{n=1}^{P} \dfrac{1 - 2\cos\omega_n z^{-1} + z^{-2}}{1 - 2\rho\cos\omega_n z^{-1} + \rho^2 z^{-2}}\right) d(t)$

Concept:

- Speech signals are usually considered as voiced or unvoiced
- Voiced sounds consist of fundamental frequency ω_0 and its harmonic components.
- CPZLP estimates all frequencies independently (harmonicity of speech not exploited).

Pitch estimation considered here are: [4]

- Subspace-orthogonality-based pitch estimation.
- Subspace-shiftinvariance-based pitch estimation
- Optimal-filtering based pitch estimation.

Pitch estimation in PEM-based AFC:

- Fundamental frequency ω_0 and its harmonic components are inserted in the CPZLP model
- Sinusoidal components are suppressed by a cascade of notch filters

Experimental Results (5)

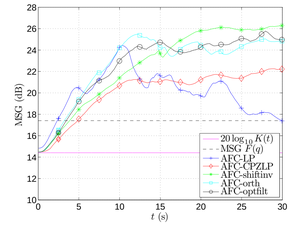
Experimental set-up:

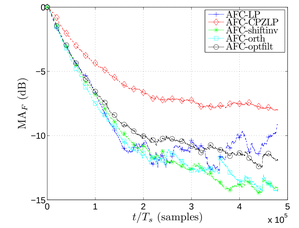
- The near-end sinusoidal model order is set to $P = 15$
- The near-end noise model order is set to 30.
- 50% overlapping data windows of length $M = 320$ samples.
- The NLMS adaptive filter length is set to $n_F = 200$.
- The near-end signal is a 30 s speech signal at f_s= 16 kHz.
- The forward path gain $K(t)$ is set 3 dB below the MSG without feedback cancellation.

Performance measures:

Maximum Stable Gain: $\mathrm{MSG}(t) = -20\log_{10}\left[\max_{\omega\in\mathcal{P}} |J(\omega,t)[F(\omega,t) - \hat{F}(\omega,t)]|\right]$

Misadjustment: $\mathrm{MA}_F = 20\log_{10}\dfrac{||\hat{\mathbf{f}}(t) - \mathbf{f}||_2}{||\mathbf{f}||_2}.$





Conclusion (6)

- Hearing aids typically used a linear prediction model in PEM-based AFC
- A sinusoidal near-end signal model is introduced here in PEM-based AFC
- Different frequency estimation methods in PEM-based AFC have been evaluated
- Performance of a PEM-based AFC with cascaded near-end signal models can be further improved by using pitch estimation methods
- The pitch estimation methods considered here are based on subspace and optimal filtering
- Overall the achievable amplification in terms of MSG is higher and the misadjustment is lower using subspace and optimal filtering methods

References

[1] T. van Waterschoot and M. Moonen, "Adaptive feedback cancellation for audio applications," *Signal Processing*, vol. 89, no. 11, pp. 2185–2201, Nov. 2009.

[2] A. Spriet, I. Proudler, M. Moonen, and J. Wouters, "Adaptive feedback cancellation in hearing aids with linear prediction of the desired signal," *IEEE Trans. Signal Process.*, vol. 53, no. 10, pp. 3749–3763, Oct. 2005.

[3] T. van Waterschoot and M. Moonen, "Constrained pole-zero linear prediction: an efficient and near-optimal method for multi-tone frequency estimation," in *Proc. 16th European Signal Process. Conf. (EUSIPCO '08)*, Lausanne, Switzerland, Aug. 2008.

[4] M. G. Christensen and A. Jakobsson, *Multi-Pitch Estimation*, Morgan & Claypool, 2009.