REAL-TIME IMPLEMENTATION OF HEARING AID WITH COMBINED NOISE AND ACOUSTIC FEEDBACK REDUCTION BASED ON SMARTPHONE

Maxim Vashkevich, Elias Azarov, Nick Petrovsky, Denis Likhachov, Alexander Petrovsky*

Department of Computer Engineering, Belarusian State University of Informatics
and Radioelectronics 6, P.Brovky str., 220013, Minsk, Belarus
vashkevich@bsuir.by*, azarov@bsuir.by, nick.petrovsky@bsuir.by, likhachov@bsuir.by, palex@bsuir.by

* corresponding author

ABSTRACT

The demonstration presents a real-time mockup of smartphone-based hearing aid with combined noise and acoustic feedback reduction. The designed reduction algorithm is based on spectral weighting approach which makes it very robust to rapid changes in feedback path either caused by displacement of the speaker/microphone or room acoustics. The aim of the demonstration is to show potential of the implemented solution especially for devices with unfixed speaker and microphone setup[1].

1. INTRODUCTION AND MOTIVATIONS

Using smartphone as a pocket hearing aid is very convenient and became rather popular recently. One of the tough challenges here is noise and acoustic feedback reduction because along with additive background noise the processed signal leaks back from the speaker to the microphone. Background noise can be reduced by spectral subtraction which is a very efficient and well known technique. For acoustic feedback cancellation there is a classical approach based on adaptive filtering [1–3], which is not as practical in conditions of unstable feedback path caused by changing distance between microphone and speakers. In such conditions adaptive filtering cannot noticeably improve maximum stable gain: when using low adaptation rates the reaction to changes becomes unpredictable, when using sufficiently high adaptation rates the speech signal drastically degrades. After significant amplification feedback path turns from linear mode to non-linear and the system becomes unstable. In order to overcome these drawbacks we developed an alternative acoustic feedback reduction technique which is based on spectral subtraction. The technique provides much higher additional gain compared to adaptive filtering in time-varying conditions. Sharing one approach both noise and acoustic feedback reduction can be merged into mutual processing pipeline which benefits from computational efficiency and improved quality of processed speech. The obtained practical result can be valuable in hearing aid design especially for pocket devices with external headsets.

The demonstration presents a fully functional hearing aid implemented as an application for a smartphone. There is no special space or any equipment-specific requirements.

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2. SCIENTIFIC AND TECHNICAL DESCRIPTION

2.1. Processing framework

The processing framework is shown in figure 1. We use a DFT filter bank implemented as weighted overlap-add structure with K channels and decimation factor $M = K/2$. The input speech signal $x(n)$ is decomposed into complex subbands by the analysis filter bank (AFB). The subbands are decimated resulting in a sequence of subband samples denoted as $X(k,m)$ where k is frequency index and m is time index. Subband samples are multiplied by constant or time-varying hearing compensations gains $G_{HL}(k,m)$ which are calculated according to a desired prescription formula [4]. The processed output signal $y(n)$ is reconstructed using synthesis filter bank (SFB).

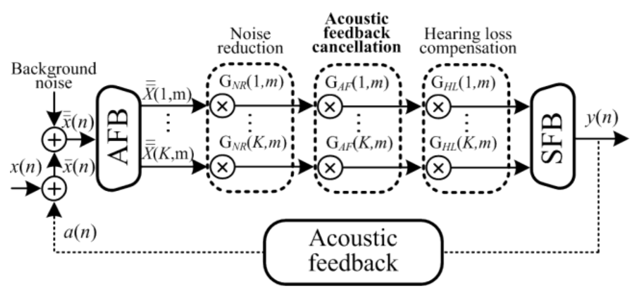

Figure 1 – Implemented processing pipeline

The input is corrupted by background noise and acoustic feedback and we assume that subband sample $\bar{\bar{X}}(k,m)$ is a sum of clean speech signal $X(k,m)$, acoustic feedback $A(k,m)$ and background noise $N(k,m)$:

$$\begin{aligned}\bar{\bar{X}}(k,m) &= X(k,m) + A(k,m) + N(k,m) \\ &= \bar{X}(k,m) + N(k,m).\end{aligned} \tag{1}$$

Minimal amplitude of speech signal can be estimated by subtracting amplitudes of noise and feedback from amplitude of the input signal. Subtracting is equivalent to multiplication by corresponding gains G_{NR} and G_{AF}. The amplitude of speech signal is recovered from noisy input as $|X(k,m)| = |\bar{\bar{X}}(k,m)| G_{NR}(k,m) G_{AF}(k,m)$. The calculation of gains relies on statistical estimation of amplitude spectrum of noise and acoustic feedback.

2.2. Weighting rules

Implemented noise estimation algorithm is based on the minima controlled recursive averaging introduced in [5]. Spectral gain value $G_{nr}\,(k,m)$ is calculated as

$$G_{NR}(k,m) = \max\left\{\sqrt{\frac{\left|\bar{\bar{X}}(k,m)\right|^2 - \nu\left|\hat{N}(k,m)\right|^2}{\left|\bar{\bar{X}}(k,m)\right|^2}}, 10^{-RL/20}\right\}, \qquad (2)$$

where ν – subtraction factor ($1 < \nu < 6$), RL – adjustable parameter that defines desired residual noise level in dB, $\left|\hat{N}(k,m)\right|$ – estimated noise amplitude spectrum which is obtained by averaging past spectral values.

Unlike background noise amplitude of feedback signal grows rapidly and quickly turns the system into non-linear mode long before noise reduction algorithm can react. We introduce the following measure of feedback gain $\chi(k,m)$ based on l previous frames and $2d$ neighboring frequency bins:

$$\chi(k,m) \triangleq \frac{\min\limits_{-l+1\le j\le 0} |\bar{X}(k,m+j)|}{\min\limits_{-d\le i\le d}\left[\max\limits_{-l+1\le j\le 0} |\bar{X}(k+i,m+j)|\right]}, \qquad (3)$$

where numerator is an estimation of $|\bar{X}(k,m)|$ and denominator is an estimation of $|X(k,m)|$. In order to avoid overrating of feedback level we use local minima over previous time samples for estimating $\bar{X}(k,m)$ and local maxima for estimating $X(k,m)$. Suppression gain $G_{AF}(k,m)$ is updated using the following expression:

$$G_{AF}(k,m) = \begin{cases} \tilde{\alpha}_{AF} G_{AF}(k,m-1) + \dfrac{(1-\tilde{\alpha}_{AF})}{\max(\chi(k,m),1)}, & \chi(k,m) < \chi_{th} \\ 1/\chi(k,m), & \chi(k,m) \ge \chi_{th} \end{cases}. \qquad (4)$$

Feedback suppression is smoothly controlled, using smoothing parameter α_{AF} ($0 < \alpha_{AF} < 1$) that specifies averaging time and a time-varying smoothing parameter $\tilde{\alpha}_{AF}$ that depends on feedback intensity $\tilde{\alpha}_{AF} = \alpha_{AF} + (1-\alpha_{AF})\left(\frac{\chi(k,m)}{\chi_{th}}\right)^{\beta}$, where β is equalizing parameter that balances reaction on quiet and loud feedback and χ_{th} is threshold value for hard decision. When $\chi(k,m)$ exceeds χ_{th} an intense acoustic feedback is detected. In this case suppression gain is updated instantaneously and then slowly released.

3. IMPLEMENTATION AND USE

The demonstration mockup is implemented as a smartphone application which processes input microphone signal and renders it back to the speakers or headphones in real-time. The application has in situ audiometry which helps the user to adjust personal hearing loss compensation. The noise and feedback reduction can be separately switched on and off in order to expose additional gain level provided by the implemented algorithm. An example of performance of the algorithm is given in figure 2 where distance between microphone and headphones is approximately 30 cm and additional gain is approximately 12 dB.

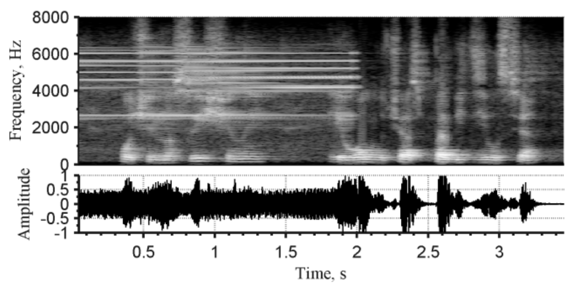

Figure 2 – Output signal recorded by real-time mockup: feedback reduction algorithm is turned on after 2 seconds

4. REFERENCES

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