

Adaptive Feedback Cancellation

Introduction

Figure 1 shows a framework for adaptive feedback cancellation (AFC). The desired input signal is $x(n)$, the actual microphone is $d(n)$, and the hearing-aid output is $s(n)$. When there is feedback, the microphone signal $d(n)$ contains the feedback signal $y(n)$ in addition to the desired signal $x(n)$. The feedback signal is generated by the hearing-aid output $s(n)$ passing through a feedback path $F(z)$. The goal of feedback cancellation is to obtain an estimate of the feedback signal, denoted as $\hat{y}(n)$, and subtract it from the microphone signal $d(n)$ (Kates, 2008). This is achieved by placing a finite impulse response filter $W(z)$ in parallel with the hearing-aid processing $G(z)$. $W(z)$ served as an estimate of the feedback path $F(z)$. Under *adaptive* feedback cancellation, the coefficients of filter $W(z)$ are continuously updated with the goal of emulating the impulse response of the feedback path $F(z)$. Under ideal conditions, the estimated feedback path $W(z)$ matches the actual feedback path $F(z)$. When this happens, the feedback-compensated signal matches the feedback signal, so that $\hat{y}(n) = y(n)$, and $e(n) = x(n)$, i.e., the input to hearing-aid processing will not contain any feedback. In practice, however, AFC is not perfect, and some mismatch remains between $e(n)$ and $x(n)$, resulting in audible distortions.

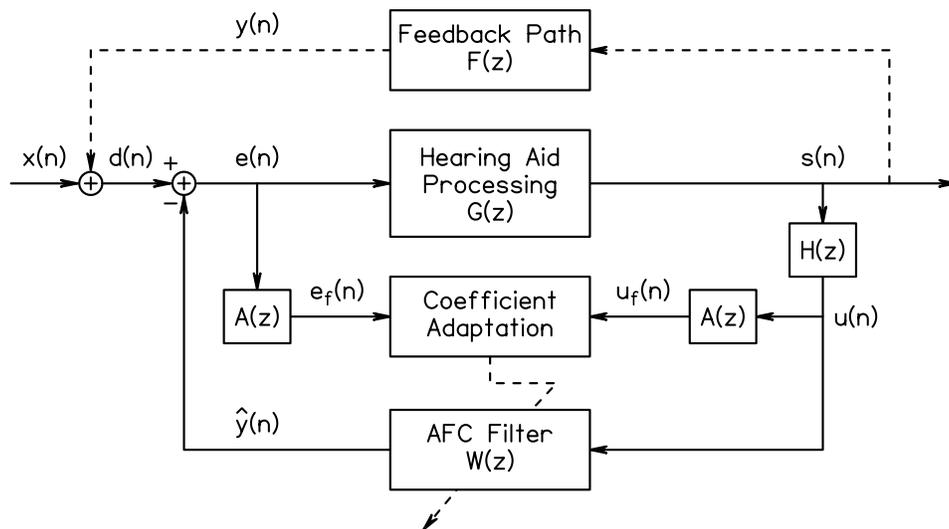


Fig. 1. Framework for adaptive feedback cancellation (Adapted from Lee et al. 2017). The actual and estimated feedback filters are $F(z)$ and $W(z)$, respectively. The whitening and band-limit filters are $A(z)$ and $H(z)$, respectively. The

The most common method for adaptive feedback cancellation is the least mean square (LMS) algorithm. Several LMS-based algorithms have been proposed, all of which include a step size parameter that controls how fast the AFC filter coefficients are updated. In the ordinary LMS algorithm, rapid fluctuations in the input signal can lead to instability. The normalized LMS (NLMS) algorithm mitigates some of the instability problems of the ordinary LMS algorithm by dividing the step size parameter by the signal power. For both NLMS and LMS coefficient adaptation method, the whitening and band-limit filters have unity gain, so $A(z) = H(z) = 1$. The normalized filtered-X LMS (NFXLMS) algorithms address some of the limitations of the NLMS method by (1) reducing non-feedback correlation and (2) concentrating adaptation on the frequency range where feedback is known to occur.

Normalized LMS

Normalized LMS (NLMS) is a variant of the LMS algorithm that improves stability by reducing sensitivity of coefficients to input signal level. The NLMS algorithm is based on cross-correlation between the output and input of the hearing-aid processing. The cross-product is weighted by a time-varying step size

$$w(n+1) = w(n) + \mu_n(n)s(n)e(n).$$

The step is normalized by the signal power to improve convergence rate and stability

$$\mu_n(n) = \frac{\mu}{\wp(n) + \epsilon},$$

where $\wp(n)$ is a signal power estimate at time n and ϵ is a power-threshold parameter that prevents the denominator from becoming too small. In our implementation, the signal power is estimated by a first-order recursive equation

$$\wp(n+1) = \rho[s^2(n) + e^2(n)] + (1 - \rho)\wp(n).$$

Note that the NLMS algorithm has three parameters: μ , ϵ , and ρ . Parameter values are optimized by minimizing the difference between the AFC filter $W(z)$ and the feedback path $F(z)$ in a simulated feedback configuration.

NFXLMS

When the NFXLMS algorithm is used to derive AFC coefficients the same calculations are performed as for NLMS except that $u_f(n)$ and $e_f(n)$ replace $s(n)$ and $e(n)$

$$w(n+1) = w(n) + \mu_n(n)u_f(n)e_f(n)$$

$$\wp(n+1) = \rho[u_f^2(n) + e_f^2(n)] + (1 - \rho)\wp(n).$$

Additional filtering is required to obtain $u_f(n)$ and $e_f(n)$ from $s(n)$ and $e(n)$. Parameter values are optimized by minimizing the difference between the simulated feedback path $F(z)$ and the product of $W(z)$ and $H(z)$. In practice, parameters are optimized by minimizing the mean-squared difference between the impulse responses $f(n)$ and the convolution of $w(n)$ and $h(n)$.

Whiten Filter

The whiten filter $A(z)$ flattens the spectrum of the input signal $x(n)$ to reduce correlation between the input signal and the hearing-aid output $s(n)$. In our implementation, filter coefficients were derived from the long-term average speech spectrum (LTASS) described by Stelmachowicz (2000).

Figure 2 show the LTASS in the left panel and the inverted LTASS in the right panel (blue curve). The frequency responses of 5-tap and 9-tap linear-phase, FIR filters that approximate the inverse LTASS are superimposed (green and red curves). Each of these filters improve the AFC performance with relatively little increase in computational load.

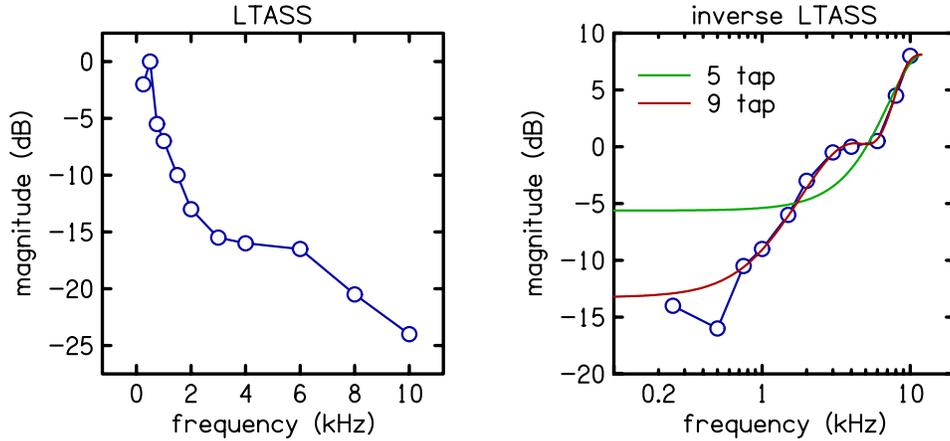


Fig. 2. Long-term average speech spectrum (LTASS) and its inverse. An inverse LTASS was constructed relative to the LTASS value at 4 kHz. The left panel superimposes 5-tap (green) and 9-tap (red) FIR approximations over the inverse LTASS (blue).

Band-limit filter

The motivation for the band-limit filter is to improve the efficiency of coefficient adaptation by focusing on the frequency range where feedback is known to occur. Our strategy is two-fold. First, we move some frequency dependence from $W(z)$ to $H(z)$. Second, we reduce the magnitude of $H(z)$ at low frequencies where feedback is unlikely to occur.

The current implementation of the band-limit filter $h(n)$ has 36 taps and non-adaptive coefficients. The frequency response of the band-limit filter has a high-pass characteristic as shown by the green line in the right panel of Fig. 3. The effect of the band-limit filter on the AFC estimate of the feedback path is demonstrated in Fig 3 by observing that the estimated feedback (red) is closer to the actual feedback (black) in the right panel compared to the left panel.

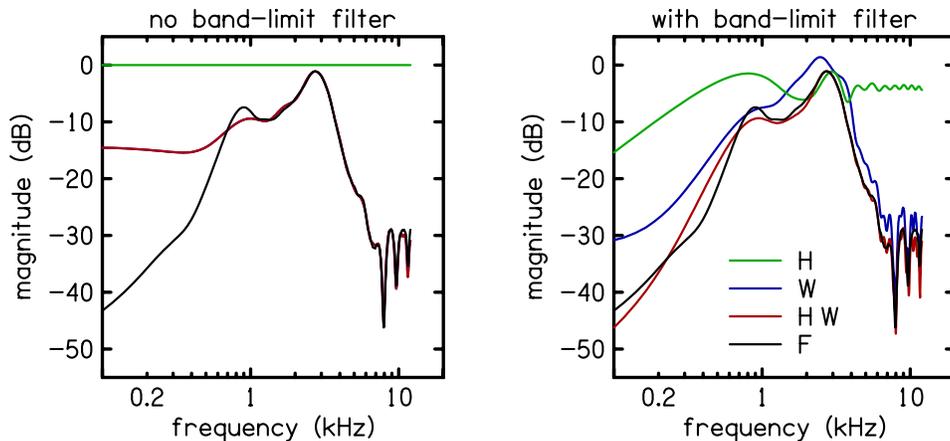


Fig. 3. Spectral magnitudes of the band-limit (H, green), AFC (W, blue), and feedback-path (F, black) filters. Also shown is combined response of the product of the band-limit and AFC filters (HW, red), which approximates the feedback-path filter. For the case without band-limiting (left panel), the band-limit filter has 0 dB gain at all frequencies. When band-limiting is applied (as described in the text), the combined response (red) better estimates the feedback-path (black) below 0.4 kHz and above 8 kHz.

Parameter Values

Optimization of parameter values was based comparison of the actual feedback path with the AFC estimate. An estimation error was defined as the mean-squared difference between the actual and estimated feedback-path impulse response. The estimation error was computed as a function of time for the “carrots” passage, which is 13 seconds long. Parameter values were selected (by a simplex search) to minimize the maximum error over the interval of time between 2 and 13 seconds. Figure 4 illustrates the parameter optimization procedure by comparing an example of actual and estimated feedback paths (left panel) and by showing an example of time course of the estimation error (right panel).

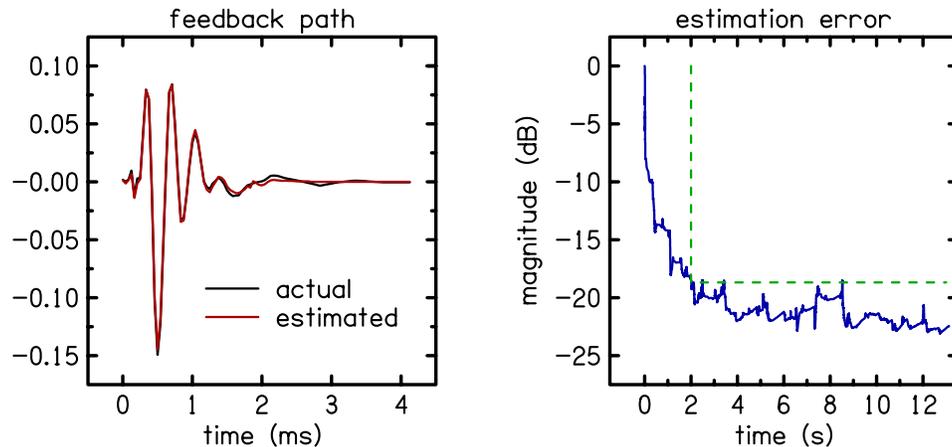


Fig. 4. Comparison of the actual (black) and estimated (red) feedback-path impulse responses (left panel) and the time course of the estimation error (right panel). The dashed (green) line (right panel) indicates the maximum-observed error over the interval of time between 2 and 13 seconds.

The four parameters optimized by this procedure appear in the equations used to calculate the AFC filter coefficients (μ , ϵ , ρ) and the band-limit filter coefficients (α). Filter lengths were also varied independently in a stepwise manner to further the estimation error. The following Table summarizes the results of parameter optimization.

| AFC Filter | Whiten Filter | Band-limit Filter | ρ | ϵ | μ | Error (dB) | Comp time |
|------------|---------------|-------------------|----------|------------|----------|------------|-----------|
| 42 | 0 | 0 | 0.000002 | 0.001395 | 0.000418 | -13.58 | 0.164 |
| 42 | 9 | 0 | 0.006840 | 0.001178 | 0.004693 | -15.27 | 0.165 |
| 42 | 9 | 20 | 0.007219 | 0.000919 | 0.004607 | -18.47 | 0.441 |

The first set of parameter values in the Table represent our NLMS algorithm which has no whiten or band-limit filters. The second set of parameters shows a reduction of 1.7 dB in the estimation error when a whiten filter is added to NLMS. The third set of parameters represents the addition of both whiten and band-limit filters, which represents our NFXLMS algorithm and shows a 5.2 dB reduction in estimation error relative to our NLMS algorithm. We hope to see a corresponding 5.2 dB improvement in added stable gain (ASG).

References

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