SUBBAND ADAPTIVE FILTERING FOR ACOUSTIC FEEDBACK COMPENSATION IN HEARING AIDS

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1. INTRODUCTION

Acoustic feedback occurs in hearing aids when the amplified sound signal played out to the user is coupled back to the input microphones. Low level feedback acts as reverberation, masking the desired signal; high level feedback can lead to system instability and high intensity oscillations that manifest as ``whistling'' or ``howling'' sounds that are very disturbing to the user.

Methods currently used to prevent feedback include limiting the gain of the hearing aid to prevent a positive feedback loop from forming, and applying adaptive notch filters to attenuate the oscillating frequencies when feedback occurs (Hamacher, 2005). Gain limiting reduces the utility of the device, while notch filtering is reactive and can distort the speech signal.

The most desirable method for feedback control is direct signal compensation. If the feedback path is known, a replica of the feedback signal can be created and subtracted from the microphone signal, enabling arbitrary gain in the hearing aid without signal distortion. The difficulty with this approach is obtaining an accurate model of the feedback path. Since the feedback path changes when the hearing aid moves within the ear, or when telephones, hats and other reflecting surfaces are nearby, dynamic modelling of the feedback path with an adaptive filter is required.

2. SUBBAND ADAPTIVE FILTERING

While the adaptive feedback path modelling is typically performed using a fullband, time-domain signal (Hamacher, 2005), subband adaptive filtering has been used in applications such as acoustic echo cancellation to overcome problems of slow convergence for coloured inputs and high complexity for large adaptive filters. Fig. 1 presents a block diagram of a subband adaptive feedback compensation system within a hearing aid. The input to the hearing aid is the microphone signal $z[n]$ and the output is the loudspeaker signal $x[n]$. The input contains the incoming speech signal $s[n]$ as well as the feedback signal $d[n]$, which is formed when the output signal is coupled to the microphone via the feedback path $H(z)$. During system identification periods, a probe signal $p[n]$ is generated inside the hearing aid.

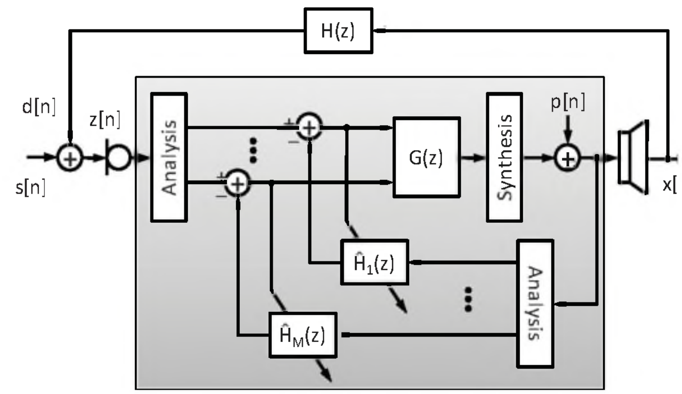

Fig. 1: Subband hearing aid feedback compensation system.

This signal is simultaneously played out through the loudspeaker and passed through a bank of adaptive filters containing estimates of the subband feedback paths. The outputs of the filters are replicas of the subband-decomposed feedback probe signal, which are subtracted from the true feedback probe signal measured at the microphone. The resulting error signal is then used in the normalised least-mean square (NLMS) algorithm to adjust the adaptive filter coefficients to the true feedback path.

Since modern hearing aids currently decompose the input signal to facilitate noise reduction and frequency dependent amplification, adaptive feedback compensation could easily be performed in subbands. However, while acoustic echo paths can require thousands of taps to accurately model, acoustic hearing aid feedback paths are much shorter. The complexity benefits are thus reduced, raising the question of whether subband structures offer any advantages for feedback compensation.

3. SUBBAND/FULLBAND COMPARISON

3. 1 Changing feedback paths

Feedback paths can change with movements of the hearing aid caused by ear or jaw movements, or with changes in the arrangement of reflecting surfaces near the outer ear (Hamacher, 2005). Depending on the nature of the path change, characteristics of the frequency response may remain the same despite the disruption. Feedback paths are typically bandpass, with the 1 - 4 kHz regions experiencing the least attenuation (Puder, 2004). An obstructing object in the feedback path may affect higher frequencies more than lower, because high frequencies are absorbed more easily and because the wavelength of the lower frequencies may be larger than the object. While a fullband filter would have to adapt all tap weights in response to the disruption, a subband adaptive filter would only be required to adjust the weights of the filters in the affected bands. Even if the effects of the feedback path change are uniform across frequency, subband adaptive filters are shorter, so they should be faster to respond to all types of path change.

To compare the performance of feedback compensation structures under controlled conditions, fullband and subband NLMS feedback cancellers were simulated using computer generated white Gaussian noise excitation in a changing acoustic environment at 25 dB measurement SNR. The NLMS adaptation step-size for both algorithms was fixed at μ=0.01. After an initial convergence period an abrupt feedback path change was simulated by changing the feedback path coefficients, and then a more realistic gradual change was simulated by linearly interpolating back to the first response. The mean-squared error (MSE) - the power of the uncancelled feedback signal - averaged over 20 trials, is shown in Fig. 2. The subband structure shows rapid initial convergence and faster re-convergence from the disturbances. During and after the feedback path changes, the subband structure offers consistently better worst-case MSE performance.

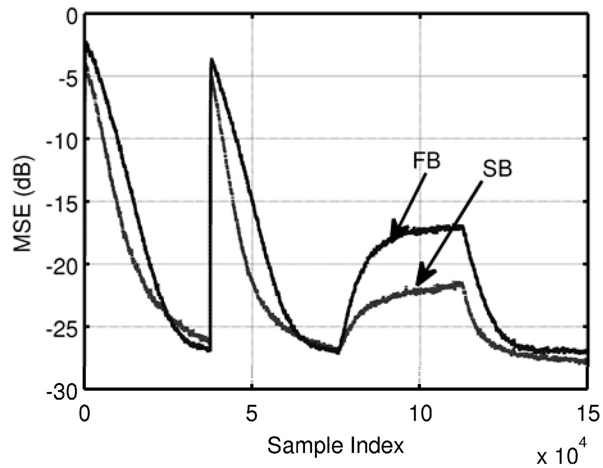

Fig. 2: MSE performance of fullband and subband feedback compensation in changing environments.

3.2 Disturbing speech signal

Since feedback path changes can occur during active speech periods, feedback cancellation systems must be able to adapt in the presence of the incoming speech signal, which acts as a high-level disturbance and can cause the system to diverge. Large step-sizes are desired for faster convergence, but also lead to faster divergence.

Subband adaptive filters can offer more flexibility in dealing with disturbing speech. Since filter divergence occurs when the disturbing signal power is high, and the power of speech signals decays with frequency, a smaller step-size can be used in the low-frequency bands with high incoming speech power, and a larger step-size in the higher frequency bands. Fig. 3 presents MSE curves for fullband NLMS with step-sizes of μ=0.005 and μ =0.025, and for subband NLMS with step-sizes that increase logarithmically with frequency, from 0.005 to 0.025. The large step-size fullband NLMS converges rapidly, but also exhibits MSE fluctuations of up to 15 dB as the interfering speech signal drives the divergence; reducing the step-size greatly reduces the MSE fluctuations, but the rate of convergence is much slower. The subband NLMS with the logarithmically-spaced step-sizes presents a compromise: the initial convergence rate approaches that of the large step-size fullband algorithm, but the divergence during strong speech segments is minimal.

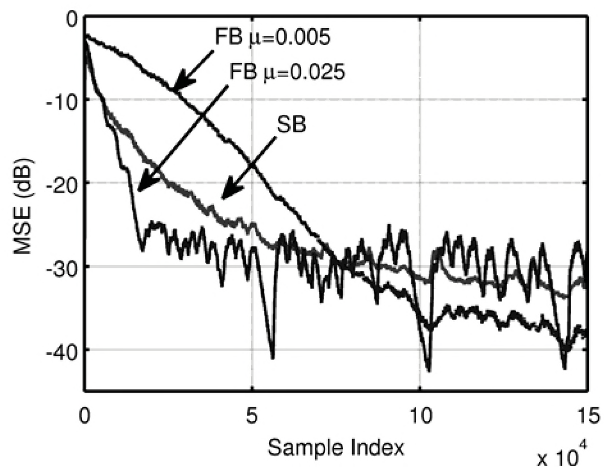

Fig. 3: MSE performance of fullband and subband feedback compensation in the presence of a high-level speech signal.

4. DISCUSSION

Hearing aids use a signal subband decomposition in order to apply frequency-dependent gain and noise reduction strategies. Performing feedback compensation on these subband signals rather than the fullband, allows the system to achieve better tracking of changing feedback paths, and offers additional flexibility in dealing with performance-stability trade-offs.

REFERENCES

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Puder, H and Beimel,B (2004) “Controlling the adaptation of feedback cancellation filters - problem analysis and solution approaches,” in Proc. EUSIPCO’04.