IMPLEMENTATION CHALLENGES FOR FEEDBACK ACTIVE NOISE CANCELLATION

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ABSTRACT

This work describes the factors limiting the performance in practical implementations of digital active noise cancellation (ANC) in commercial systems. This includes the codec delay, the secondary path variability, and the acoustic limitations. The causes for each limiting factor are studied and solutions are proposed that significantly improve system performance and robustness.

Index Terms— Active noise cancellation, delay, analog-to-digital, FXLMS, audio, headphone.

1. INTRODUCTION

Active noise cancellation (ANC) deals with reduction of noise by generating anti-noise that is 180° out of phase with the ambient noise. ANC has many applications like hearing protection, listener fatigue reduction, and vibration control systems. Digital ANC has a rich literature with many tutorials having been published on the subject [1,2]. The primary focus of this work is to highlight design challenges and possible solutions to incorporate active noise cancellation solutions for portable audio applications like mobile communications devices and audio headsets. The emphasis here is on practical implementation issues that need to be considered for commercial-quality ANC systems. First, the problem of codec (i.e, A/D and D/A converters) delay is considered in Sec. 3. The effects of delay on cancellation performance are analyzed and a possible solution using novel predictive FXLMS algorithm is described. The effectiveness of the proposed solution is established by evaluating noise reduction performance in different environments such as car-noise and train-noise. The second issue considered in Sec. 4 is the effect of temporal secondary path variation in headsets. Efficient mechanisms to improve the performance under this variability are proposed. Finally, in Sec. 5 the acoustic limitations of the system are considered and procedures to avoid non-linearities are discussed.

This work provides efficient tools to develop *practical* ANC solutions for headphones and portable audio devices. It aims to bridge the gap between the rich theoretical literature in the subject, and the practical implementation issues for commercial quality ANC systems for practical use.

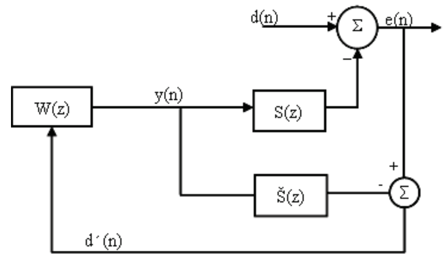

Fig. 1. *An ideal single microphone ANC system.*

2. FB ACTIVE NOISE CANCELLATION

A generic FB active noise canceler is described in fig. 1. For an ideal causal system the limits of the error have been derived in [4]. The optimal Wiener solution $W_{FB}(z))$ is given by,

$$W_{FB}(z)) = \frac{1}{S_{MP}(z)\Gamma_d(z)} \cdot \left\{\Gamma_d(z) \cdot S_{AP}^{-1}(z)\right\}_+ . \quad (1)$$

Here, $S_{MP}(z)$ and $S_{AP}(z)$ are the stable minimum phase and the stable all pass systems. These are deduced from the stable secondary path $S(z)$ when it is decomposed as a product of $S_{AP}(z)S_{MP}(z)$. $\Gamma_d(z)$ is the minimum phase causal spectral factor of the noise $d(n)$ $\left(\Gamma_{dd}(z) = \Gamma_d(z)\Gamma_d(z^{-1}) \right)$. If the secondary path is assumed to be a simple delay $S(z) = z^{-n_0}$, the residual error can be deduced as,

$$e(z) = (1 - \frac{z^{-n_0}}{\Gamma_d(z)} \{\Gamma_d(z)z^{n_0}\}_+) \cdot d(z). \quad (2)$$

This shows that the error is a function of the correlation of the noise signal. As the correlation in the noise increases, it is possible to design better controllers to cancel noise. On the contrary, if the noise is white with unit variance the error can be reduced to,

$$e(n) = (1 - \delta(n)u(n - n_0)) * d(n).e(n) = d(n) \quad (3)$$

This shows that white noise cannot be canceled using a feedback controller with any delay. Under colored noise scenario

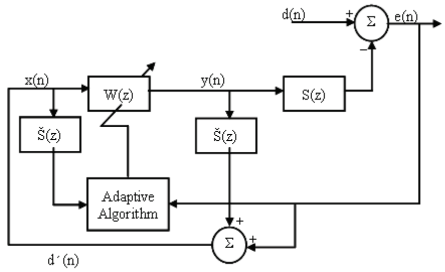

Fig. 2. *Feedback FXLMS system*

the resultant system acts like a whitening filter filtering out the correlated components.

In practice the single-microphone feedback FXLMS algorithm is implemented as shown in fig. 2. Here the adaptive algorithm is the Least-Mean-Square (LMS) algorithm [3, ch. 5]. The adaptive system tries to estimate the reference signal $d(n)$ by observing the error signal $e(n)$. $S(z)$ is the transfer function of the secondary path which accounts for the acoustic system between the canceling speaker and the sensing microphone. Usually, $S(z)$ is estimated off-line using the LMS algorithm with the system actuated by white noise as an input to the speaker. In practice another adaptive algorithm can be used to fine tune it on-line to account for system variabilities. The estimate of this transfer function $S(z)$ includes the inherent codec delays in the secondary path along with the acoustics of the system. The main issues addressed here are the limitations in performance of the FXLMS system due to the secondary path variations and the delays in the digital path. The next section describes the issues associated with designing signal processing systems for wide-band noise cancellation in headphones.

3. CODEC DELAY

The main challenge in implementing practical ANC solutions is the delay associated analog-to-digital and digital-to-analog conversion in the noise canceling systems. This delay is a major concern in systems that use sigma-delta converters which have become dominant due to prevalence of voice-band codecs in the recent years. The problem of codec delay, is usually overlooked in the ANC literature. This may not be critical for narrow-band noise cancellation, but it is critical for wide-band noise cancellation. The model in Fig. 3 includes the codec delays in the secondary path transfer function S(z). The codec delays have been indicated by blue boxes Fig. 3. This delay causes misalignment of the data because the filter coefficients during the LMS adaptation due to which the computed filter will in general have incorrect indices. To overcome this problem, the modifications (red boxes) as shown in fig 3 give a more accurate delay model. Here, Δ_d and Δ_a stand for the digital-to-analog and analog-to-digital delays respectively. Also $\Delta_t = \Delta_a + \Delta_d$. To overcome this misalignment due to delays a Predictive FXLMS is proposed

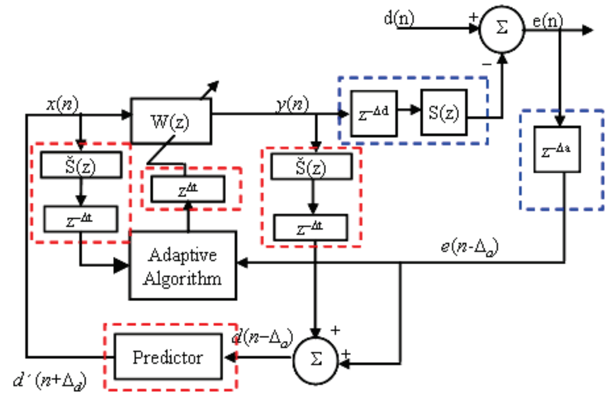

Fig. 3. *FXLMS system with secondary path delays (blue boxes) and a predictor to compensate for delays (red boxes)*

as described in fig. 3. This scheme has two main additional components. The predictor component compensates for the misalignment in x(n) and the shift after the adaptation compensates for the misalignment in the filter coefficients update. In particular, the modified filter coefficient update has the form

$$W(n+1) = W(n-\Delta_t) + \mu e(n-\Delta_a) \cdot x(n). \quad (4)$$

The objective of the predictor is to estimate $E\{d(n+\Delta_d), d(n+\Delta_d-1), .., d(n-\Delta_a+1) | d(n-\Delta a), .., d(0)\}$. If d(n) is assumed to be the output of an auto-regressive (AR) process, then the well known Levinson-Durbin algorithm [3, ch. 3] can be used to estimate the parameters of this AR process given the samples $d(n-\Delta_a), .., d(0)$. Once the AR parameters are estimated, they are used to generate the expected values of all future samples, i.e., if we denote

$$x(r) = d(r+\Delta_d), \text{ for } 0 \leq r \leq n-\Delta_t$$

then,

$$x(r) = \sum_k a_k x(r-k), \text{ for } n-\Delta_t < r \leq n$$

where a_k are the estimated AR parameters from the Levinson-Durbin algorithm. It should be noted that , the AR parameters need not be calculated for each time unit if the statistics of the input signal are stationary or slowly varying. For example, It was found experimentally that a small 0.5-dB degradation is experienced when the AR parameters estimation is run only in 1% of the frames of an train noise. However, in this case the correlation values (which are required by the Levinson-Durbin algorithm) need to be re-evaluated.

Another important feature of the predictive FXLMS that significantly improves the performance is that higher adaptation rates can be used. If it is assumed that the signal d(n) is an AR process with additive white noise, then the correlation matrix of the signal is:

$$R_{dd} = R_{xx} + R_{nn} \quad (5)$$

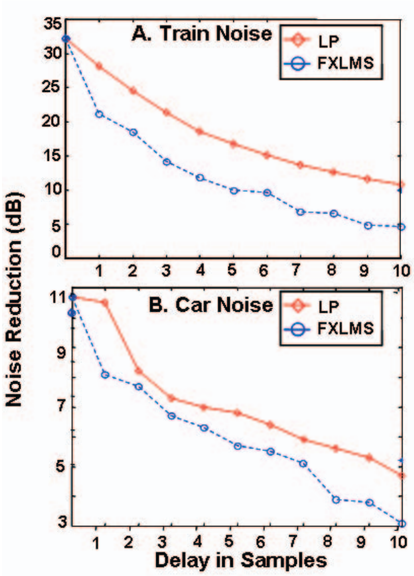

Fig. 4. *Noise cancellation performance for FB systems with and without prediction. With increase in the codec delay, the noise reduction performance decreases, this is compensated by the predictor.*

where R_{xx} is the correlation matrix of the AR process and R_{nn} is the correlation matrix of the additive white noise and it equals $r_n(0)I_m$ where I_m is the identity matrix of size M. In this case the eigenvalues of R_{dd} and R_{xx} are related as:

$$\lambda_d = \lambda_x + r_n(0) \tag{6}$$

i.e., the maximum eigenvalue of R_{xx} is smaller than the maximum eigenvalue of R_{dd}. This enables us to use a bigger step size in LMS as the limits on the step size in the LMS algorithm is given by [3],

$$0 < \mu < 2/\lambda_{max} \tag{7}$$

3.1. Evaluations

The proposed predictive FLXMS algorithm was evaluated using real noise recorded in different environments using a small microphone mounted on a headphone. The sampling frequency in all cases is 22050 Hz. First we evaluate the performance of the basic FXLMS algorithm and the proposed algorithms for train and car noises at different delays. The results are illustrated in Fig. 4. For this experiment, the adaptation step μ was fixed for the best performance within stability constraints. From the Fig. 4, we notice that with an increase in the secondary path delay, the performance of the FXLMS algorithm decreases. Ideally, we would like the FXLMS performance to be robust to any delay in the error path. This is accomplished to a large extent by the proposed predictive FXLMS (with linear prediction) which outperforms the standard FXLMS algorithm by up to 8 dB. The predictive FXLMS algorithm outperforms the basic FXLMS algorithm

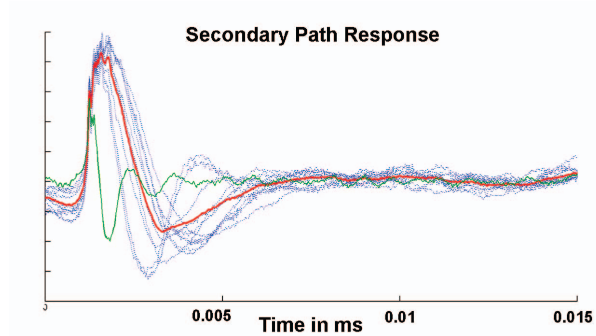

Fig. 5. *Secondary path variation for 10 subjects when the headphone in worn. the blue responses are individual responses for each subject. The red response is an averaged response and the green response is when the headphone is not worn.*

even with signals that have short correlation (as in the case of car noise). With stationary noise types like Train Noise the performance of the FXLMS algorithm drops almost exponentially with an increase in the codec delay and here the predictor exploits the correlation in the noise to maintain a robust noise cancellation performance.

4. SECONDARY PATH VARIABILITY

As mentioned earlier, secondary path for digital headphone active noise cancellation systems is a combination of D/A, A/D, speaker and the microphone responses. The secondary path varies primarily due to the difference in head transfer function and a shift in headphone position. This section discusses the issue of the variation in secondary path and the main approaches to handle it.

In practice, systems have a predetermined secondary path and it is adjusted using adaptive algorithms when the noise canceler is functional (On-line adaptation) [1]. This is effective for small variations that come with small shifts in headphone position or small mismatch across different users. In Fig. 5, the dotted lines show the variation in secondary path across 10 users. These impulse response measurements were taken at the a sampling rate of 48000 kHz using a pair of Audio Technica headsets. When the headphone is worn, the headphone cavity resonances dominates the secondary path and there is a relatively small variation about the mean secondary path response(red). This is different than the case when the headphone is removed (green), this response is a function of the headset and does not depend on subject and usage. As seen from Fig. 5 there is a drastic change in the secondary path due to the absence of the resonating cavity. The magnitude of this temporal change in the secondary path due to removal or wearing of the headphones can cause either the secondary path adaptation or the controller or both to diverge. This divergence causes unpleasant residual noise effects to the listener. In theory, this can be controlled by using a very small value of the adaptation factor μ in the FXLMS adaptation algorithms. This ensures that the feedback response is relatively constant as the secondary path changes. The drawback to this scheme is that the algorithm takes a long time to converge for changes in noise leading to poor noise cancellation.

Another solution is to have a non-linear controller which freezes the adaptation or terminates the noise cancellation loop when the headphone is not on the head. This prevents the adaptation from diverging due to spurious changes is secondary path when the headphone is removed or worn. There are many ways to detect the status of the headphone including:

- Using the adaptation error as a metric such that adaptation takes place when the error is small.
- Using a secondary controller with fewer taps as a headphone on-off detector.

It is important that the time lag in detecting the secondary path change is minimal. This allows for higher values of convergence rates μ if the detection is almost instantaneous. In the two cases presented this is critical as the on-line estimation of the headphone position will be dependent on the system latency. Between the approaches presented above, using a parallel detection scheme is faster than monitoring the error at the cost of higher resource requirements for implementation.

5. ACOUSTIC LIMITATIONS

The last factor considered here that limits the performance of the ANC system is the acoustic capabilities of the speaker and the microphone. Typical microphone responses are between 20-20000 Hz for analog microphones and 300-10000Hz for digital microphones. However, the speaker responses vary greatly depending on the size and make of the speakers. There is a tendency to choose smaller speakers to meet power, cost, and aesthetic constraints in system design. The use of smaller speakers directly limit the effectiveness of the speaker at lower frequencies. The frequency response for a typical speaker used for mobile applications is shown is fig. 6. Usually, with a decrease in size of the speaker, the low frequency response falls sharply. This is an issue for mobile applications where most of the noise observed and are a candidate for ANC are environmental noise. This noise observed typically low frequency noise below 500Hz as seen in fig. 6 for Car and Train Noise. In designing cancellation systems for these noise types, only slight passive attenuation is obtained at lower frequencies doe to the device mechanics. On using FXLMS for such systems, the adaptive controller tries to compensate for this mismatch by increasing the gain of the feedback signal at lower frequencies. This leads to digital clipping of the signal or this drives the speaker to nonlinearity producing undesirable artifacts in the resultant anti-noise. To avoid the nonlinearities due to overdriving the speaker or clipping at lower frequencies, the general idea is to improve the overall secondary path response to extend the response to lower frequencies. In this case, the best solution is to improve the speaker response at lower frequencies by choosing better speakers during design. This provides better signal reproduction without compromising noise cancellation performance. Additionally, the error signal should be filtered using a high-pass filter with the same response as the speaker. This prevents overdriving of the speaker by the generated anti-noise and thus preventing non-linearities.

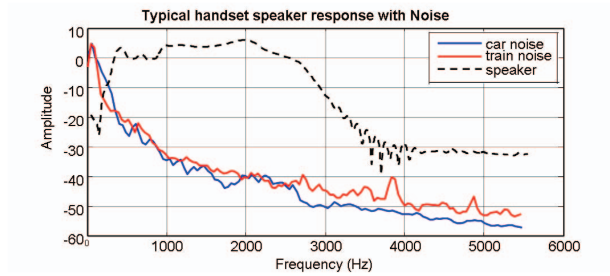

Fig. 6. *Small speakers have poor response at lower frequencies (dotted line) where the noise is predominantly present (solid lines).*

6. DISCUSSION

This work describes the challenges in designing wide-band ANC systems for headphones and portable audio devices using a single microphone. Digital ANC systems have been quite successful for narrow-band ANC solutions but less effective for wide-band solutions. The main implementation issues in designing effective wide-band noise canceling systems with commercial quality have been described. The main challenges considered are secondary path delay, temporal variation in the secondary path, and the transducer limitations in the context of FXLMS algorithm. These issues were analyzed in detail, and efficient solutions to reduce their effect on the system performance are discussed.

Future work will consider the secondary path variations in depth and approaches to overcome temporal variations. These studies will be key to developing practical ANC systems with commercial quality.

7. REFERENCES

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