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Headphone with Active Noise Control using Analog Adaptive Filters

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Abstract Traditional methods of acoustic noise control are based on materials that absorb sound waves. These methods employ relatively bulky materials, and present some deficiencies that become more important at low frequencies. An active noise control (ANC) system may be used to efficiently reduce noise with frequencies below 500Hz. Most ANC systems for headphones are based on fixed analog controllers with feedback, that usually reduce the noise up to 15dB for frequencies below 500Hz, independently of the noise spectrum. Changes in the environment, in the positioning of the phone, in the ageing of the components, and different users, modify the transfer function of the secondary path, reducing the system performance. Given this limitation, the use of feedback digital adaptive filters has been investigated. The main factor limiting the performance of feedback digital controllers applied to headphones is the group delay of the secondary path. The group delay of an analog system is smaller than that introduced by a digital system (due to the anti-aliasing, reconstruction filters, analog-to-digital and digital-to-analog converters used for digital processing). In order to reduce the contribution of the group delay due to the electrical system and to improve the correlation between the reference signal and the noise, we propose the use of a headphone with active noise control using analog adaptive filters with a FIR-like structure, using instead of a delay line a cascade of gamma filters.

1. INTRODUCTION

Active noise control follows the principle of the destructive wave interference, reducing an unwanted acoustic noise generated by a primary source through an anti-noise produced by a secondary source. Figures 1 and 2 show a headphone with feedback ANC and its electrical block diagram [1] using digital adaptive filters, where $d(n)$ is the primary noise, $e(n)$ is the residual noise measured by the error sensor, $y'(n)$ is the anti-noise produced by adaptive filter $W(z)$, $x(n)$ is the estimate of the primary noise obtained as

$$x(n) \equiv \hat{d}(n) = e(n) + \sum_{k=0}^{K-1} \hat{s}_k y(n-k) \ , \tag{1}$$

where $\hat{s}_k$ is an approximation to the impulse response of the secondary path transfer function $S(z)$. The secondary path is composed by the transfer functions of the error microphone, the pre-amplifier, the anti-aliasing filter, the analog-to-digital converter (ADC), the digital-to-analog converter (DAC), the reconstruction filter, the power amplifier, the loudspeaker, and the acoustic path from loudspeaker to error microphone. The feedback ANC system may be viewed as a feedforward ANC system able to create its own reference signal.

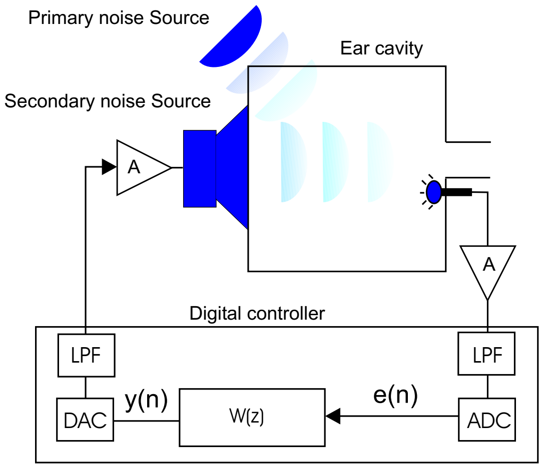

Figure 1: Headphone with ANC system

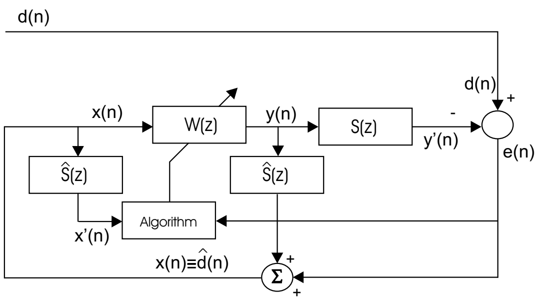

Figure 2: Feedback ANC system

In order that $y'(n)$ exactly cancels $d(n)$, the adaptive filter W(z) would need to model the inverse transfer function of $S(z)$, that is

$$W(z) = \frac{1}{S(z)} \ . \tag{2}$$

Real-time ANC requires that the processing time of the samples be less than than the sampling period, and that $W(z)$ be causal. Unfortunately, in general $S(z)$ is not minimum-phase, so its inverse is non-causal, and hence unrealizable. If the noise were periodical (such as a pure tone), this would not be a problem, but for broadband or white noise $W(z)$ would be unable to produce an exact anti-noise to completely cancel the primary noise.

In order to have a causal $W(z)$ close to the ideal solution, the group delay of $S(z)$ must be decreased as much as possible.

2. SECONDARY PATH

In this section we present the characteristic delays introduced by electrical, electroacoustic, and acoustic transfer functions for headphones with ANC.

The ADC, DAC, and two low-pass filters (LPF) of the digital secondary path generate an eletrical delay τ_d given approximately by

$$\tau_d = \frac{1}{f_s}\left(1 + \frac{3.2.n}{8}\right) , \tag{3}$$

where f_s is the sampling frequency, and n is the number of poles of each LPF [2]. The delay of the acoustic path is given by

$$\tau_a = \frac{r}{c} , \tag{4}$$

where r is the distance between the loudspeaker and microphone, and c is the speed of sound. The condenser microphone we used has a negligible delay.

The total secondary path delay can be express as

$$\tau_T = \tau_d + \tau_a + \tau_l , \tag{5}$$

where τ_l is the delay, due to loudspeaker.

If instead of the digital system, an analog system were applied, the group delay of the secondary path would be decreased to

$$\tau_T = \tau_a + \tau_l , \tag{6}$$

due to not use of the LPF´s and converters.

It should be emphasised that τ_d could be reduced in two ways. The first would be to reduce the number of poles, but there is a lower limit to the filter order that still avoids aliasing. Another possibility would be to increase the sampling frequency, but at a high sampling rate the adaptive control algorithm would have some problems that we discuss in the next section.

3. ADAPTIVE CONTROL

In order that the secondary noise source cancels the primary noise, we need both a reference signal correlated with the primary noise source and a filter able to transform the reference signal into an anti-noise. Changes in the transfer functions due to changes in the environment, in the positioning of the error microphone, the ageing of the components, and different users, degrade the performance of fixed (non-adaptive) controllers.

Adaptive filters can be realizated using FIR or IIR structures. Unlike IIR adaptive filters, FIR adaptive filters have the advantage of having an unimodal error surface. FIR adaptive filters are also simpler to implement, and their convergence is usually faster. However, for systems with long impulse responses, IIR structures may lead to better modeling. In general, an IIR filter will require less coefficients for a given level of performance than an FIR filter.

In the case of ANC systems with a single microphone, the secondary path has a relatively short impulse response, so the simpler FIR structure should be the best choice.

The least-mean-square (LMS) algorithm for a FIR digital adaptive filter $W(z)$ of order M is [3] given by

$$w_m(n+1) = w_m(n) + \mu x'(n)e(n) \ , \tag{7}$$

where $w_m(n)$ is a coefficient of the vector $\mathbf{W}(n)$, $x'_m(n)$ is a filtered version of $x_m(n)$ express by

$$x'_m(n) = \sum_{k=0}^{K-1} \hat{s}_k x_m(n-k) \ , \tag{8}$$

$m = 0, 1, \cdots, M-1$, and μ is the digital adaptive gain whose upper bound is given by

$$\mu < \frac{2}{3} \frac{1}{M\sigma^2_{x'}} \ , \tag{9}$$

where $\sigma^2_{x'}$ is the variance of $x'(n)$.

The performance of an ANC system for non-periodical noise depends on group delay introduced by the secondary path, which in principle favours the use of the analog system.

The group delay of a digital system could be diminished if a high sampling frequency were used. The advantage of increasing the sampling frequency is the possibility of reducing the order of the LPF, and/or of keeping the group delay low in the frequency range of interest (up to approximately 500Hz) by pushing up the LPF's cut-off frequency. However, a higher sampling rate means less time to compute every new output, and also requires the use of more parameters for the digital adaptive filter. Adaptive filters using the LMS algorithm have a misadjustment [3] given by

$$\sigma^2_{\text{misad}} = \sigma^2_o \left(1 + \mu \frac{M\sigma^2_{x'}}{2}\right) \ . \tag{10}$$

We can note in (10) that a high value of M results in higher misadjustment. Increasing M would also require a reduction in the speed convergence, since the condition for stable performance of the filter must be as (9).

In order to implement analog adaptive filters with a FIR structure without the need of analog delay lines (all-zero structure), cascades of all-pass or low-pass filters can be used. With all-pass filters (such as Laguerre or Kautz filters), the estimation problem is orthogonal, and better-conditioned [4], however, for the sake of simplicity we used a cascade of low-pass (gamma) filters [5]. The use of a cascade of filters instead of a delay line, may also lead to a smaller number of parameters to estimate, if the (fixed) low-pass filter parameters are properly chosen.

The block diagram of the digital and analog adaptive filters used to minimize the square error between $y'(n)$ and $d(n)$ are shown in figures 3 and 4.

Analog adaptive filters implemented via gamma filters have a structure similar to that of a transversal filter, with the delay line replaced by a cascade of low-pass filters with transfer function

$$H(s) = \frac{a_0}{s + w_0} \ . \tag{11}$$

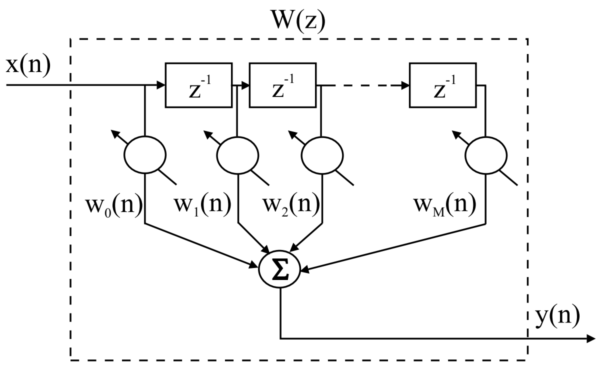

Figure 3: Digital adaptive filter

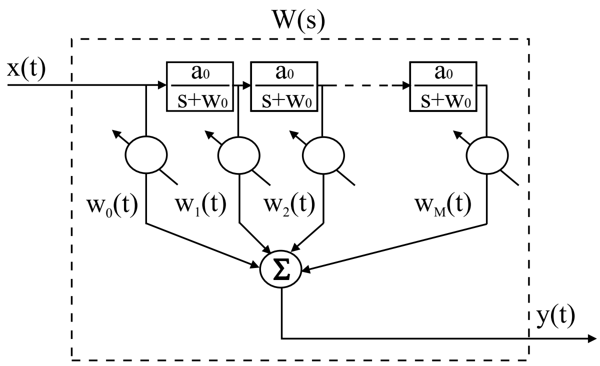

Figure 4: Analog adaptive filter

The filtered-X LMS algorithm for an analog adaptive filter $W(s)$ of order M is [6, 7] given by

$$\frac{\mathrm{d}\, w_m(t)}{\mathrm{d}\, t} = \rho x'_m(t) e(t) dt \ , \tag{12}$$

where $w_m(t)$ is the m-th coefficient of the vector $\mathbf{W}(t)$ and $x'_m(t)$ is a filtered version of $x_m(t)$, expressed by the convolution operation below

$$x'_m(t) = \hat{s}(t) * x_m(t) = \int_0^t \hat{s}(\varphi) x_m(t - \varphi) \ , \tag{13}$$

Unlike the case of digital adaptive filters, the analog adaptive gain ρ can be arbitrarily large without affecting the stability of the algorithm, but a faster convergence increases the misadjustment [8].

4. RESULTS

The transfer function of the digital secondary path was estimated through the off-line method [9], with a pseudo-noise as input (a maximum-length sequence) [10]. A sucessive-approximation analog/digital converter was used, due to its group delay that is, in general, significantly lower than the group delay introduced by sigma-delta converters.

In our experiments, we used f_s = 8000 Hz, $n = 7$, $r = 5$ mm, and $c = 343$ m/s, that result in $\tau_d = 0.781$ ms, and $\tau_a = 0.015$ ms. The loudspeaker has a significative delay, especially at low frequencies; for 100 Hz the delay was $\tau_l = 2.055$ ms, and for 500 Hz, $\tau_l = 0.957$ ms. The total digital secondary path delay was $\tau_T = 2.851$ ms for 100 Hz and $\tau_T = 1.753$ ms for 500 Hz.

In order to estimate the analog secondary path a sine sweep (50 to 4000 kHz) was used. With the frequency response, the Matlab function `invfreqs` was used to find an approximate transfer function $S(s)$. For 100 Hz the total delay was $\tau_T = 2.070$ ms, and for 500 Hz $\tau_T = 0.972$ ms. Figure 5 shows the magnitude and phase of the digital (solid line) and analog (dashed line) secondary paths.

We analyzed the performance of the digital and analog systems through computer simulations. A signal composed by sinusoids with frequencies of 200 Hz, 300 Hz, and 400

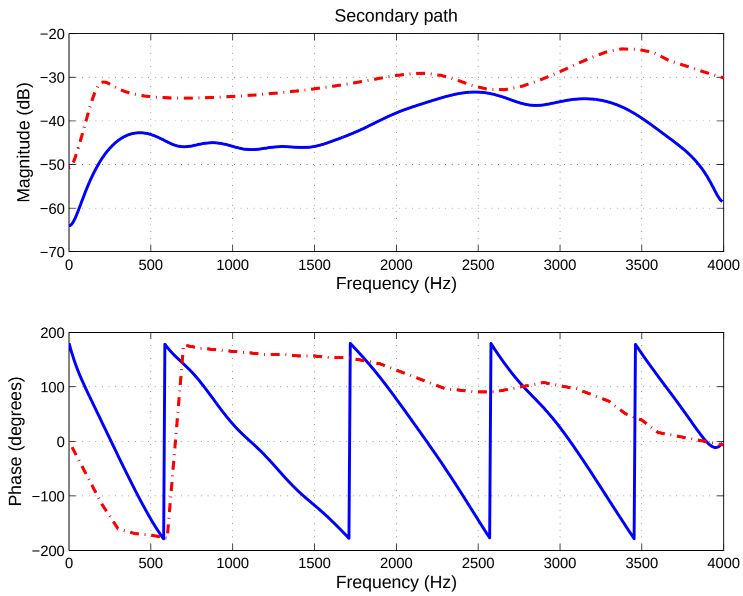

Figure 5: Digital secondary path (solid line) and analog secondary path (dashed line)

Hz, added to a white noise with zero mean and unity variance band-limited between 120 Hz and 600 Hz, was used as primary noise source.

The digital adaptive filter $W(z)$ used to model the inverse of $S(z)$ was an FIR filter with 128-taps and $\mu = 0.1$. For the analog system, $W(s)$ was implemented with a 10-taps gamma filter cascade and $\rho = 10^4$ (these values of parameters μ and ρ were chosen by trial and error, and achieved the best results for each filter). Since the variance of the filtered version of $x(n)$ is low, the parameters μ and ρ result in high values. The gamma filter was implemented with the transfer function

$$H(s) = \frac{2512}{s + 2512} \,.$$

Figures 6 and 7 show the noise at the error microphone without (solid line) and with (dashed line) the ANC system. For the digital ANC system, the primary noise was reduced by aproximadely 10 dB, while the feedback ANC system implemented with the analog adaptive filter achieved a reduction of about 20 dB.

5. CONCLUSION

We proposed a method to improve the performance of ANC headphones, by using an analog adaptive filter. This approach allows the use of adaptive filters with a reduced total delay in the ANC system's secondary path, compared with digital filters, thus keeping the advantages of both analog (low group delay) and digital (adaptiveness) ANC systems. We have tested the performance of our analog adaptive filter through simulations, using transfer functions measured in a comercial headphone. Our simulations show the potential performance improvement that can be achieved by using analog adaptive filters.

In future works we intend to implement our system, and compare its results with digital

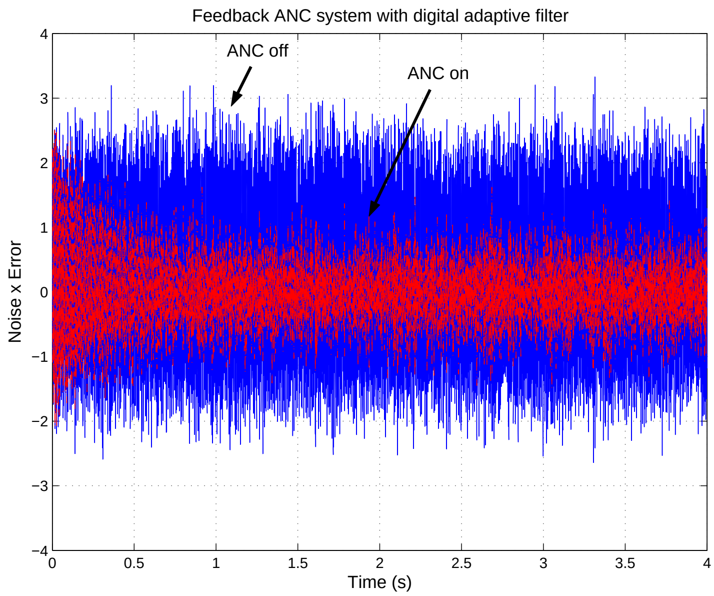

Figure 6: Feedback ANC system with digital adaptive filter without (solid line) and with (dashed line) using the ANC system

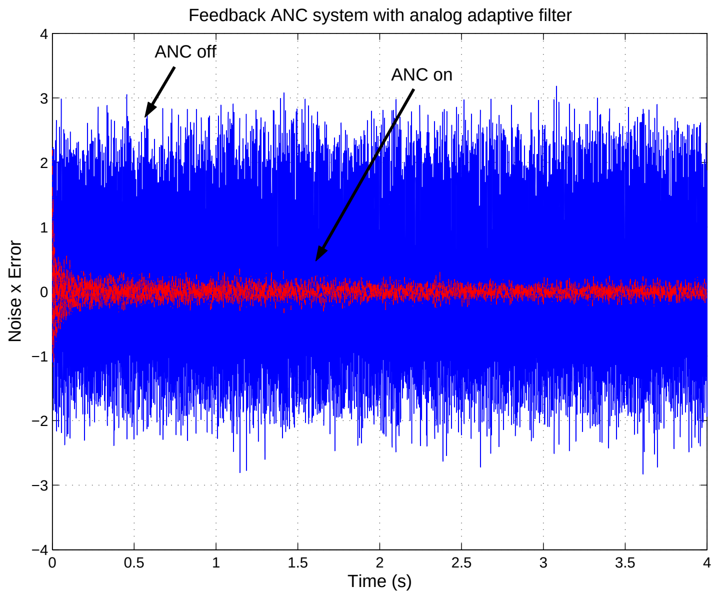

Figure 7: Feedback ANC system with analog adaptive filter without (solid line) and with (dashed line) using the ANC system

adaptive and fixed analog headphone ANC systems. We also intend to investigate which are the best choices for the gamma filter parameters, and experiment with orthogonal filters, such as Laguerre and Kautz filters [4].

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