PERFORMANCE OF LOCAL CONTROL EQUALIZATION OF PERIODIC NOISE

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ABSTRACT

Several works have been carried out concerning active cancellation and equalization of tonal and periodic noise. However little attention has been given to the performance of such systems working on real environments. Therefore, this work addresses the problem of evaluating the performance of real active equalizers, which belong to the multichannel feedforward type, for local control. The aim of this work is to improve the comfort sensation that the active controller produces on listeners. Equalization zones around listener ears have been measured for different configurations and different algorithmic variants. The effects over the performance of the bad estimation of the error paths have been also studied and evaluated.

1 INTRODUCTION

Advances in digital signal processing and the versatility of the active control techniques have allowed to develop new applications using active noise control (ANC) techniques. Concretely, the well-known adaptive noise canceller applied to repetitive noise, firstly proposed by Widrow *et al.* [1], has evolved to provide not only a reduction of sound field level, creating a zone of quiet around the controlled points, but to create an specific acoustical sensation retaining some residual noise [2, 3]. For example inside a vehicle, driver could prefer to hear some audible information instead of quiet to improve safety or to feel more comfortable. In order to reshape the sound field, Kuo *et al.* [4] proposed an adaptive noise equalizer (ANE) which uses the filtered-X LMS (FXLMS) algorithm to adapt the coefficients of a two-weights filter, minimizing a pseudo-error signal instead of the residual noise. That ANE was implemented in a real single channel active system. The multichannel extension of this ANE, which is called classical ANE, is developed in this work. This algorithm together with an algorithmic variant

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named novel ANE [5] have been used to test the ability of real multichannel active controllers to achieve some given specifications such as: a given attenuation and/or amplification of sound at some chosen frequencies (usually harmonically related), an equalization zone around listener ears and a desirable subjective acoustic sensation.

The performance of the real controllers concerning the mentioned aspects is measured carrying out the following analysis and experiments:

- An steady state analysis of the transfer function between the error (residual or after control) signal and the primary (under control) signal. This way the theoretical performance of the ANE in terms of attenuation and/or amplification of a given single frequency can be observed. Also some undesirable effects as the overshoot effect [6] can be determined for different algorithms and for accurate and non accurate secondary path estimation.
- An objective measure of the residual acoustic field at listener ears and therefore an objective measure of the achieved cancellation. This objective measure has to be repeated for multiple points inside the controlled (equalized) area to consider displacements of the listener head inside this controlled zone.
- Finally a subjective study about the effects that the active control causes on listeners has also to be carried out. An analysis of the psychoacoustic parameters [7] and its relation to the subjective impression that ANE produces on humans. This subjective impression can be evaluated by means of tests [8]. An example of this kind of study can be found in [3].

2 STEADY STATE ANALYSIS

To actively control periodic signals Kuo *et al.* [2, 4, 9, 10] proposed the single and multiple frequency adaptive noise equalizers, which minimize a pseudo-error signal instead of the residual error. Those equalizers use the FXLMS algorithm to adapt the coefficients of two-weight filters, one for each reference signal (single tone). In [5] the multichannel version of the multiple frequency ANE was developed and called multichannel ANE, which uses the multiple error LMS algorithm to adapt the filters' coefficients. The steady state transfer function of the ANEs can be obtained by analyzing the signal propagation from the primary path output $d[n]$ to the system output $e[n]$ [1, 2, 11]. The transfer function of the multiple frequency ANE is described below. The aim to develop the extension of the ANE to a multichannel system is that it results much more efficient for local control applications.

Multiple frequency active noise equalizer. The multiple frequency ANE with filtered-X proposed by Kuo in [2, 4, 10] allows to control K different single frequency reference signals, which are usually generated from a synchronization signal, using K single frequency ANE in parallel [4]. We will consider the following expressions to calculate the close-loop transfer function $H_I(z)$ [2] between the primary input signal $d[n]$ and the noisy output of the system $e[n]$, taking into consideration the secondary path (system response $S(z)$ and impulse response $s[n]$), and its estimation modeled with the filter $\hat{S}(z)$. Figure 1 shows the signals involved in the computation of the transfer function. In particular, $y_k[n]$ is the k-th cancelling signal, which feeds two branches: the balancing branch and the cancelling branch. Signals pass through the cancelling branches, being affected by the $(1-\beta_k)$ factor, and the corresponding secondary paths in order to cancel the primary signal, $d[n]$. The cancelling result, called error signal $e[n]$, is fed back to the cancelling system and then combined with signals from the balancing branches to generate the pseudo-error signals, $e'_k[n]$. Parameter β_k represents the gain parameter at the frequency ω_k and its value determines the attenuation or amplification of the primary signal at a given frequency achieved by the ANE. Four different work modes to actively control the primary signal at a given frequency, ω_k, exist: $\beta_k = 0$ indicates cancellation mode, $0 < \beta_k < 1$ indicates attenuation mode, $\beta_k = 1$ keeps the signal intact, and $\beta_k > 1$ amplifies the signal by β_k.

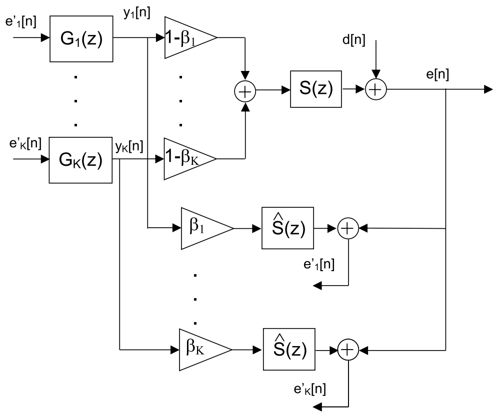


Figure 1: *Block diagram of the multiple frequency ANE depending on open-loop transfer functions* $G_k(z)$*.*

Considering the open-loop transfer function $G_k(z)$ between the k-th pseudo-error signal $e'_k[n]$ and the k-th filter output signal $y_k[n]$, defined in [1, 2] and showed in Figure 1, we can derive the transfer function $H_I(z)$ of the multiple frequency ANE with the FXLMS algorithm. This result can be extended to the multichannel case, see [5].

$$H_I(z) = \frac{E(z)}{D(z)} = \left\{ 1 - S(z) \sum_{k=1}^{K} \frac{G_k(z)(1-\beta_k)}{1-\beta_k G_k(z)\hat{S}(z)} \right\}^{-1}, \quad (1)$$

where the open loop transfer function is expressed as, see [10]:

$$G_k(z) = -2\mu_k A_k^2 \hat{A}_{sk} \frac{z cos(w_k - \hat{\phi}_{sk}) - cos\hat{\phi}_{sk}}{z^2 - 2z cos w_k + 1}, \quad (2)$$

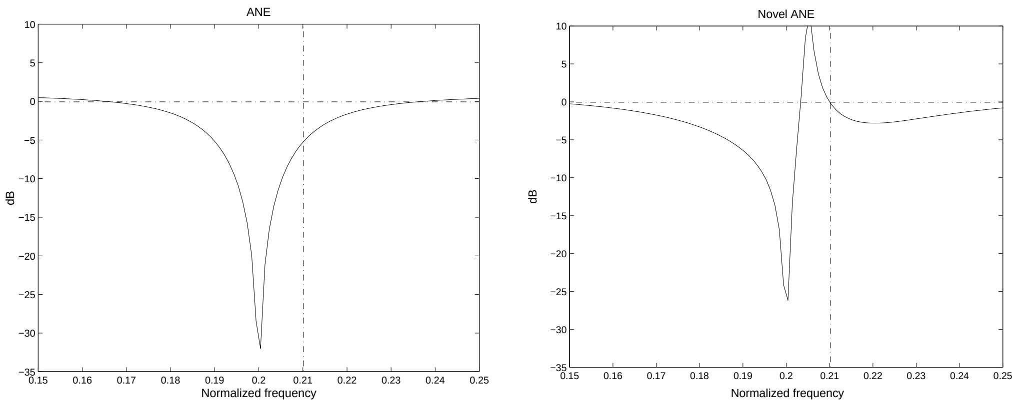
$\hat{A}_{sk}$ and $\hat{\phi}_{sk}$ denote respectively the amplitude and phase of $\hat{S}(z)$ at frequency ω_k.

The ANE described above can be modified to achieve meaningful computational savings and better control of each single frequency. A new version of the adaptive equalizer named novel ANE has been developed [5]. A common pseudo-error signal is used for each secondary path in this algorithm. A reduction in computations is achieved, concretely $K-1$ convolutions are saved and even more in the multichannel case.

2.1 Simulation results

Performance of the ANEs can be examined by means of the relative positions of the transfer functions poles and zeros, and the frequency responses of the systems. Several simulations have been carried out considering distinct system parameters. Some chosen examples are shown in the following.

Ideal secondary paths ($S(z) = \hat{S}(z) = 1$). In case we want to cancel f_1 and allow to pass f_2, $\mathcal{B} = [0\ 1]^T$, the behaviour of both equalizers differs as it is shown in Figures 2.(a) and 2.(b). Primary signal component of frequency f_1 is completely eliminated, but the amplitude of the component of frequency f_2 is only unchanged using $H_{II}(e^{j2\pi f})$, see Figure 2.(b). On the other hand, the $H_{II}(e^{j2\pi f})$ transfer function, which corresponds to the novel ANE, could



(a) (b)

Figure 2: *Amplitude response in dB of (a) the ANE transfer function $H_I(e^{j2\pi f})$ and (b) the novel ANE transfer function $H_{II}(e^{j2\pi f})$. Parameter settings: $\mathcal{B} = [0\ 1]^T$, $f_1 = 0.2$, $f_2 = 0.21$, and $\mu = 0.1$.*

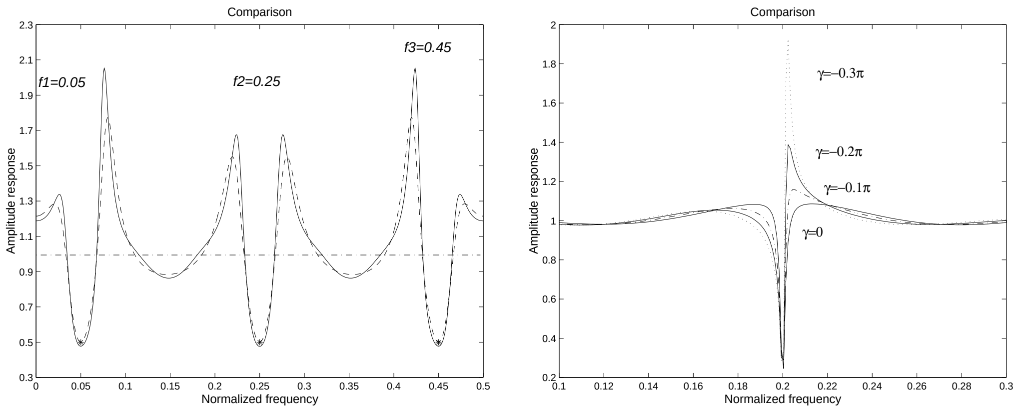
generate an undesirable overshoot at the intermediate frequencies.

Perfect secondary path estimation ($S(z) = \hat{S}(z) \neq 1$). The existence of a secondary path in the system slightly modifies the transfer functions analyzed varying the position of poles and zeros and the amplitude responses. The secondary path filters will shift the angle of the poles away from $\pm\omega_k$, introducing an undesired out-of-band overshoot in the neighborhood of ω_k, which is negligible when μ_k is small [10][6]. Simulations show that the size of the overshoot varies as a function of reference signal frequency. In some cases, novel ANE shows a slightly better performance, see Figure 3.(a). It must be noted that the amplitude at overshoot peaks and the deep of notches (marked with an $'*'$ at the correct value) are better for the novel ANE.

Phase errors on secondary path estimation. When there exists a mismatch between the secondary path and its estimated filter $\hat{S}(z)$, the overshoot is larger than in the previous examples as is illustrated in Figure 3.(b). The amplitude responses of both equalizers with $f = 0.2$, $\beta = 0$, $\mu = 0.1$ and different phase errors between actual and estimated secondary path model are shown for phase errors of values $\gamma = 0$, -0.1 π, -0.2π and -0.3π. It is easily verified in Figure 3.(b) that both transfer functions exhibit the same behaviour and that the overshoot increases with γ. Another characteristic of the overshoot is that it is not symmetric with respect to f. Finally, it should be also noted that simulations using phase errors (γ) larger than $\pi/2$ lead to the instability of both systems providing poles outside the unit circle. Kuo *et al.* analyzed theoretically this point in [6].

2.2 Experimental results

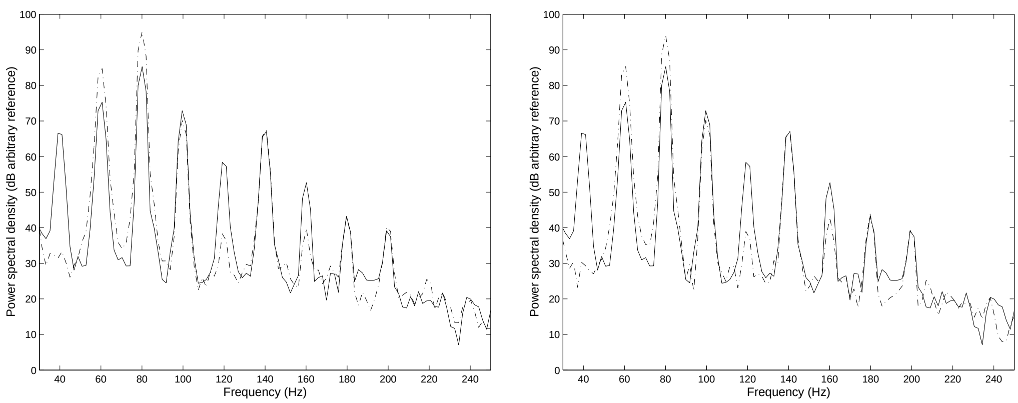
An active noise control system was installed inside a rectangular enclosure of $1.80\ m$ wide, $2.30\ m$ long and $2.40\ m$ in height. The practical multichannel system implemented consisted of: two error sensors array, two secondary loudspeakers and the primary noise source. A head and torso simulator with two calibrated microphones at the ear canals allows to measure the acoustic signals at different points inside the room. A mobile platform controlled by a computer has been designed to move the mannequin and carry out measurements at different points of the space. These points are located in a plane of the space around the listener ears. Synthesized repetitive noises with harmonics of 15 and 20 Hz, and a 80 Hz single tone were



(a) (b)

Figure 3: *Comparison of ANE and Novel ANE.* $S(z) = z^{-10}$*. (a) Amplitude response when* $f_1 = 0.05$*,* $f_2 = 0.25$*,* $f_3 = 0.45$*,* $\mathcal{B} = [0.5\ 0.5\ 0.5]^T$*,* $\mu = 0.1$ *and* $\hat{S}(z) = z^{-10}$*. ANE solid line and Novel ANE dashed line. (b) Amplitude responses when* $f = 0.2$*,* $\mu = 0.1$*,* $\gamma = 0$*,* -0.1π*,* -0.2π *and* -0.3π*,* $\beta = 0$*.*

used to test the performance of the equalizers implemented. The sampling rate used were 500 Hz and the cut-off frequency of filters 200 Hz.



(a) (b)

Figure 4: *Power spectral density of synthesized periodic noise with harmonics of 20 Hz measured at the right mannequin's microphone with gain parameter vector* $\mathcal{B} = [1\ 0\ 1.5\ 1.5\ 0.5\ 0\ 1\ 0\ 0]^T$ *before the active system operation (solid line) and after the active system operation (dashed line) using: (a) the ANE or (b) the novel ANE.*

Figure 4 illustrates a measurement obtained at the right mannequin's microphone before (solid line) and after (dashed line) ANC system operation using the ANE (Figure 4.(a)) and the novel ANE (Figure 4.(b)) for a gain parameter vector $\mathcal{B} = [1\ 0\ 1.5\ 1.5\ 0.5\ 0\ 1\ 0\ 0]^T$. From these values of the β gain parameter, the system should cancel ($\beta = 0$): 40 Hz, 120 Hz, 160 Hz and 180 Hz. It is observed that the 180 Hz frequency remains the same, this is due to the effect of the proximity of the cut-off frequency of filters. It must be noted that the frequency components corresponding to a β parameter higher than 1 increase their spectrum amplitude as it was expected. The power spectral densities obtained for both equalizers show a very similar performance.

Equalization zones. The possibility to measure the sound field around the head of

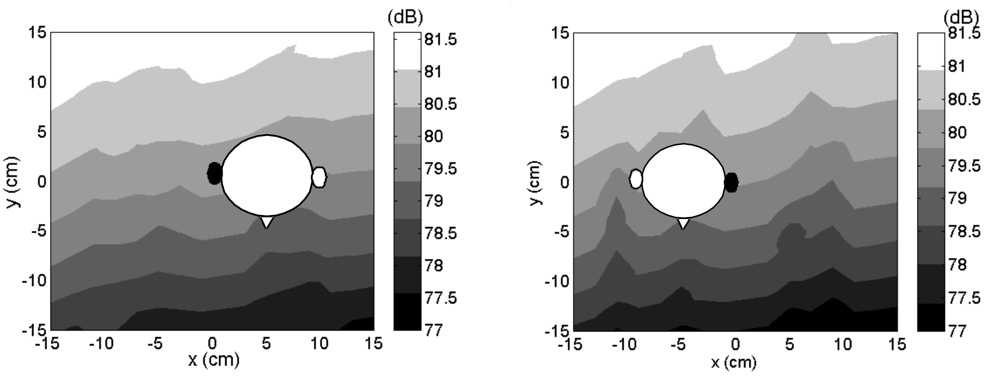


Figure 5: Sound field for a primary signal of 80 Hz. Head relative position is shown. Measurement microphone is colored in black: (a) right microphone and (b) left microphone.

the listener is provided by a mobile platform. The equalization zones will be measured in the plane of the mannequin ears so that we can record what a listener seated there might hear. Figure 5 shows the acoustic field for a 80 Hz single tone signal. Sound field level changes inside the measurement area are less than 5 dB. Attenuation levels between 33 and 38 dB were achieved. Several experiment were carried out and the equalization areas were measured. Different error sensor configurations, periodic primary signals and equalization parameters, β, were also tested.

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