Active Noise Control of Speech in Headphones using Linear Prediction

Mikkel Krogh Simonsen, Kasper Kiis Jensen, Maxime Démurger, Oliver Palmhøj Jokumsen, Christian Claumarch
Aalborg University, Acoustics and Audio Technology - 1st Semester

Active Noise Control (ANC) is a widely used technique. However most solutions are targeted at attenuating static noise with limited bandwidth. This paper focusses on using prediction to improve performance for non stationary signals. The solution proposed uses an FIR-filter adapted by an Filtered-x Least Mean Square (FXLMS)-algorithm combined with Linear Prediction to attenuate a speech signal. This solution uses the quasi-periodic properties of speech to estimate the upcoming content of a speech signal. Thereby decreasing the sample delay in the system which increases bandwidth, yielding better attenuation.

1 Introduction

Active Noise Control (ANC) is a used technique in a wide variety of applications. Research is being done in creating silents zones[1], Improving room acoustics using multiple loudspeakers[2] and is already available in various headphones, for noise cancellation, both for professional and consumer use.

Noise canceling headphones are used in various environments which have different noise characteristics. These characteristics vary from periodic low frequency noise (X-Y Hz), e.g. from machinery and plane turbines, med-range frequency noise (Y-Z Hz), e.g. speech, to high frequency noise (Z-Q Hz), e.g. X. The high frequency noise is naturally attenuated by using a closed headphone cup and static low frequency noise are attenuated by the present consumer headphones[1].

Here is going to be a graph of different headphones tested with non stationary signals (Speech)

Figure 1: Figure in the making

Assuming speech as the primary noise source, the solution used in this paper as a reference is a digital feedforward system using the Filtered-X Least Mean Squares (FXLMS). The solution is chosen due to it being the optimal system for non stationary signals. The general solutions in ANC are described thoroughly by Hansen et al. [3], [4].

The problem of a feedforward system is the dependency of noise having a propagation delay from the noise source the source.

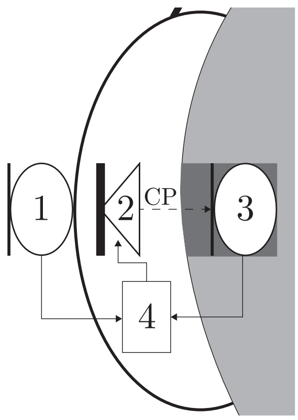

Figure 2: Simplified ANC Headphone on ear. Showing the outer reference microphone (1), a headphone loudspeaker (2), an error microphone (3) and a DSP (4).

A parameter in ANC is the bandwidth which is determined by the system delay, introduced by sampling and processing the signal in the feedforward system. Consumer ANC headphones has a bandwidth that does not cover the entire frequency area of speech. This paper examines how to extend the bandwidth.

To increase bandwidth a prediction algorithm is proposed as a potential solution. In order to do

[1]Numbers and sources will be added to the statements

prediction, certain signal characteristics must be known. Prediction of speech is described by Wai C. Chu [5]. In this paper we will combine the reference solution with the prediction of speech to increase the bandwidth of a real time system.

The paper is split into two parts. The first part describes the method used in the reference solution and how to predict speech using Linear Prediction (LP). The second part describes simulations and the implementation. Performance of the reference and the combined solution are determined and the increase in performance of the LP ANC system is verified.

2 Methods

Figure 2 shows a head fitted with an ANC headphone using a reference microphone (1), a headphone loudspeaker (2), an error microphone (3) and a DSP (4).

2.1 Feedforward ANC using FXLMS

Figure 3 shows an expansion of the DSP-block from Figure 2. This consists of converters and a control filter which is a FIR-filter where the coefficients are adapted by the FXLMS-algorithm.

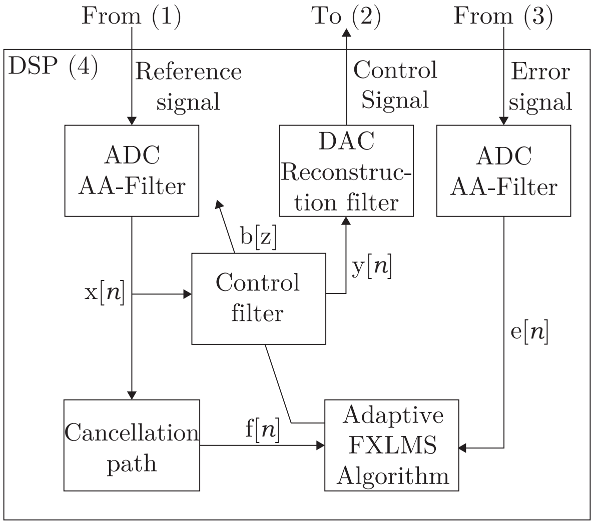

Figure 3: Adaptive feedforward ANC system.

Control Filter is the filter representing the transfer function from the reference microphone to the headphone loudspeaker. The filter, shown in equation 1, is initialized with the inverse of the first 256 samples from the impulse response of the transfer function found by measurements in **??**.

$$y[n] = \sum_{n=0}^{L-1} b_j[n]x[n-j] \tag{1}$$

Where:

$b_j[n]$ is the weight coefficients written as $b[n] = [b_0[n], b_1[n], \cdots, b_{L-1}[n]]^T$

FXLMS is the optimization algorithm which updates the control filter coefficients using the FXLMS method shown in Equation 2 and derived in **??**.

$$b_j[n+1] = b_j[n] - 2\mu e[n]f[n-j] \tag{2}$$

Where:

μ is the convergence factor

$e[n]$ is the error

$f[n]$ is the reference convolved with the Cancellation Path

Cancellation Path (CP) is the transfer-function from the headphone loudspeaker to the error microphone. In the literature [4] the CP is adaptively adjusted, but it is assumed constant in this setup because the headphone position is fixed.

When implementing the system, delays exist due to the anti-aliasing and reconstruction filters. The delays of the system exceeds the propagation time of sound from the reference microphone to the headphone loudspeaker resulting in poor performance. Therefore an LP-algorithm is proposed to predict future samples in order to decrease the effect of the time delays.

2.2 Characteristic of Speech

Speech can be split into two main classes, voiced and unvoiced. Voiced sounds are characterized by a strong periodicity, with the fundamental frequency referred to as the pitch frequency (50 - 500 Hz). Unvoiced sounds are characterized as random. Speech is a non stationary signal and can only be assumed wide sense stationary (WSS) for periods of 20 - 30 ms.

2.3 Linear Prediction of Speech

In order to predict future samples the auto correlation function (ACF), which is used in LP, of speech must be estimated in frames. The outlines of the prediction system is shown in figure 4.

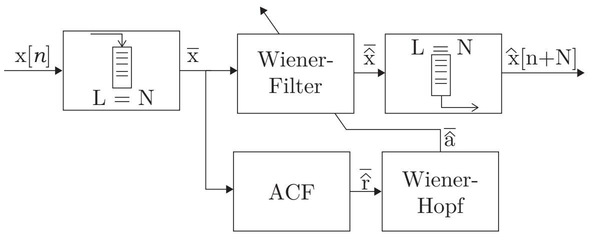

Figure 4: Linear prediction overview

Utilizing a nonrecursive estimation of the ACF, shown in Equation 3, it is possible to determine the linear prediction coefficients (LPCs) which are used for predicting future samples. When estimating the ACF of the input (x) a Hamming window (w) is applied.

$$r_x[l] = \sum_{n=|l|}^{N} x_l[n] w_l[N-n] \tag{3}$$

Where:
$x_l[n] \ = x[n]x[n-l]$

$w_l[n] \ = w[n]w[n+l]$

l is the lag

m is the frame index

N is the frame size

The LPCs are determined using Equation 4, known as the Wiener-Hopf equation.

$$R \cdot \bar{a} = -\bar{r_x} \tag{4}$$

Where:
R is the covariance matrix C_{xx}

$\bar{a}$ is the LPCs, $\bar{a} = [a_1, a_2, \cdots, a_N]^T$

$\bar{r_x}$ is the ACF, $\bar{r_x} = [r_x[1], r_x[2], \cdots, r_x[N]]^T$

Equation 4 can be rewritten as shown in Equation 5 yielding the LPCs directly.

$$\bar{a} = -R^{-1} \cdot \bar{r_x} \tag{5}$$

Calculating R^{-1} is computationally heavy on a DSP. To estimate the LPCs the Levinson-Durbin method is used. Prediction using wiener filtering, shown in equation 6, can then be applied to the current frame for prediction of the next frame.

$$\hat{x}[n+1] = -\sum_{i=1}^{N} a_i x[n-i] \tag{6}$$

Using equation 6 in cascade where $\hat{x}[n+2]$ is estimated using $\hat{x}[n+1]$ and $x[n]$. The predicted frame is then used as input for the ANC system.

Results

#404 Error..

- Prediction gain
- comparison of ref and predict ref
- comparison of ref and predict ref
- comparison of frequency response between:
  - no cancel
  - ref cancel
  - ref predict cancel

The system

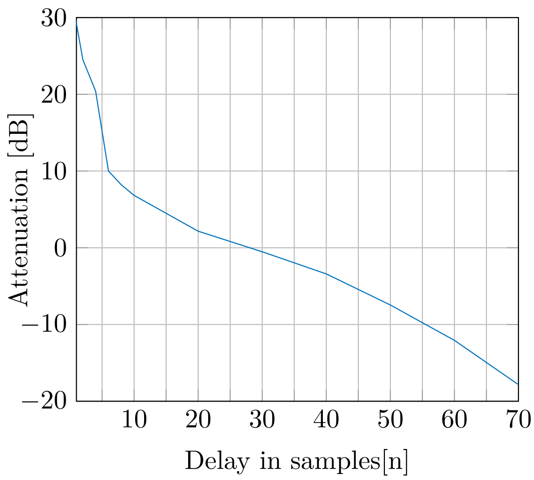

Figure 5: Attenuation achieved by the system for different system delays.

3 Dicussion

4 Conclusion

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