Human Voice Filtering System for Hearing Aid Devices to Utilize in Learning Environments

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***Abstract*—Human speech, pivotal in communication, encounter challenges due to unwanted noise in speech signals. The proposed system employs adaptive filtering techniques to eliminate extraneous noise, optimizing incoming auditory input and enhancing the clarity of speech. By mitigating these obstacles, the system aims to facilitate improved learning experiences, especially benefiting individuals with hearing impairments, in educational environments. This technology, crucial for addressing the needs of individuals facing hearing challenges, enhances the quality of transmitted speech signals and fosters a more conducive learning environment.**

***Index Terms*—active noise cancellation, adaptive filtering, hearing aid devices, background noise**

I. Introduction

Human speech holds an unmatched significance in human communication, serving as the bedrock of our interactions and a conduit for conveying emotions, knowledge, and thoughts. This significance extends to educational environments, particularly within university settings, where the seamless transmission and reception of spoken information are vital for effective learning and academic success. Nonetheless, the comprehension of human speech can face impediments, primarily due to the presence of unwanted noise in speech signals.

This extended abstract is dedicated to exploring the development and significance of a Human Voice Filtering System tailored for integration into Hearing Aid devices. The central objective of this system is to enhance learning environments by ensuring uninterrupted comprehension of spoken language, especially in educational contexts.

To address these challenges effectively, the proposed system employs an adaptive filtering mechanism meticulously designed to eliminate extraneous noise from speech signals. Leveraging adaptive filtering techniques, this system optimizes incoming auditory input, resulting in a clearer and more intelligible signal. This enhanced audibility fosters an improved learning experience, particularly benefiting individuals with hearing impairments. The relevance of this technology within the realm of hearing aid devices cannot be overstated, as it directly addresses the critical needs of individuals contending with hearing challenges, enabling their more effective engagement in educational settings. A pivotal component of this system is adaptive filtering, which dynamically adjusts its parameters to minimize unwanted noise, significantly elevating the quality of transmitted speech signals and enhancing the learning environment for individuals facing hearing challenges.

II. Active noise cancellation utilization

This research paper focuses on the implementation of active noise cancellation within hearing aid devices, employing an innovative approach involving adaptive filtering to enhance the quality of captured auditory signals [1]. The proposed mechanism utilizes two microphones: one capturing the combined human voice and background noise (Signal 01) and the other solely acquiring the background noise (Signal 02). The core concept involves the estimation and subtraction of the background noise component from the primary captured signal to achieve an enhanced, noise-reduced auditory output for hearing aid applications.

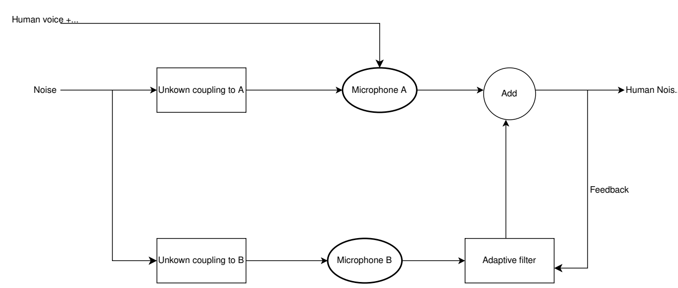


Fig. 1. Active noise cancellation

A. Active noise cancellation and Adaptive filtering

The methodology involves the application of an adaptive filtering system to deduce the background noise from the combined Signal 01, which includes both human voice and ambient noise. Leveraging adaptive filtering principles, the system accurately estimates the background noise content solely from Signal 01. The secondary microphone dedicated to capturing the background noise (Signal 02) serves as a reference for the adaptive filter, enabling it to discern and isolate the unwanted noise component from the primary signal.

Subsequently, the estimated background noise component is subtracted from Signal 01, resulting in a processed signal that effectively mitigates the background noise, thereby enhancing the clarity of the captured human voice. The derived signal after this subtraction represents the desired output for hearing aid devices, ensuring an improved auditory experience by reducing the interference caused by ambient noise.

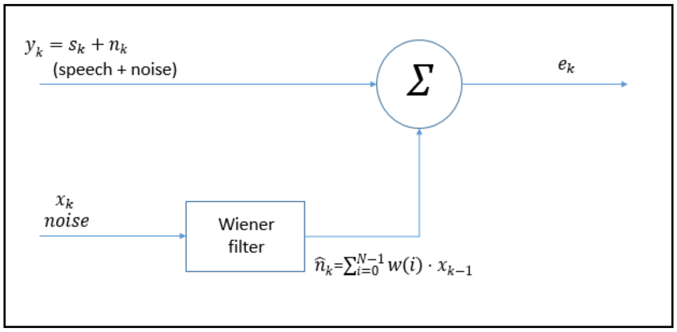


Fig. 2. wienier filter

III. Understanding Adaptive Filtering in the System

The core attribute defining an adaptive filter lies in its dynamic and self-modifying nature over time. Typically structured as a Finite Impulse Response (FIR) filter, an adaptive filter incorporates an algorithm that iteratively adjusts its filter coefficients. This continuous adaptation aims to minimize an error signal based on a specific criterion. The adaptive algorithm ensures that the filter coefficients evolve in response to input variations, thereby optimizing the filter's performance and enhancing its ability to accurately capture and process changing signals. Understanding "Wiener filtering theory" helps to figure out the function of the "Adaptive filtering block" we used in Simulink model.

IV. Wiener Filter Theory

Referring to Figure 2, y_k represents the kth sample of the signal y. This y_k consists of two distinct components: s_k denoting the principal signal and n_k representing the noise component. The fundamental objective of the Wiener filter is to acquire the most suitable estimation for n_k, denoted as $\hat{n}_k$.

The Wiener filter of interest to us is an FIR (Finite Impulse Response) filter characterized by N coefficients [2]. The estimation of the error signal is obtained as follows:

$$e_k = (y_k - \hat{n}_k) = y_k - \sum_{i=0}^{N-1} w(i) \cdot x_{k-1}$$

Here, $w(i)$ represents the ith coefficient of the filter. The calculation is simplified by the application of matrices as follows:

$$\mathbf{x}_k = \begin{bmatrix} x_k \\ x_{k-1} \\ \vdots \\ x_{k-(N-1)} \end{bmatrix}$$

$$\mathbf{w} = \begin{bmatrix} w(0) \\ w(1) \\ \vdots \\ w(N-1) \end{bmatrix}$$

With the substitution of matrix notation, we arrive at:

$$e_k = y_k - \mathbf{w}^T \mathbf{x}_k = y_k - \mathbf{x}_k^T \mathbf{w}$$

The squared error of the signal is calculated as follows:

$$e_k^2 = y_k^2 - 2\mathbf{w}^T(y_k \mathbf{x}_k) + \mathbf{w}^T \mathbf{x}_k \mathbf{x}_k^T \mathbf{w}$$

By obtaining the expected value of this expression, the Mean Square Error (MSE) is derived:

$$\text{MSE} = \mathbb{E}[e_k^2] = \mathbb{E}[y_k^2] - 2\mathbf{w}^T \mathbb{E}[y_k \mathbf{x}_k] + \mathbf{w}^T \mathbb{E}[\mathbf{x}_k \mathbf{x}_k^T] \mathbf{w}$$

This MSE can be further simplified using the "Autocorrelation Matrix" $\mathbf{R}_{xx}$. By leveraging this matrix, the Least Mean Square Error (LMSE) can be determined. However, as it is not the primary focus of this paper, we refrain from elaborating further on this topic. Nevertheless, this MSE is utilized to continually adjust the filter weights, ultimately producing the optimal estimation for the background noise.

V. Implementation through MATLAB Simulink

We utilized the MATLAB Simulink environment to simulate our model. The primary approach involved the application of an adaptive filtering block to estimate the background noise signal, as discussed in the context of the "Wiener Filter." The adjustment of filter weights was conducted using a step size of 0.001. At the "Input" port of the filter, we provided the background noise signal captured through a dedicated microphone in the hearing aid device. Simultaneously, at the "Desired" port of the filter, we input a combination of human speech along with background noise. This amalgamated input enabled the filter to compute the estimation for the background noise at the "Output" port of the filter.

The expected signal of the model was derived by subtracting the estimated background noise $\hat{n}_k$ from the "desired" signal y_k. Subsequently, the amplified expected signal $(y_k - \hat{n}_k)$ was directed to the output speaker of the hearing aid device.

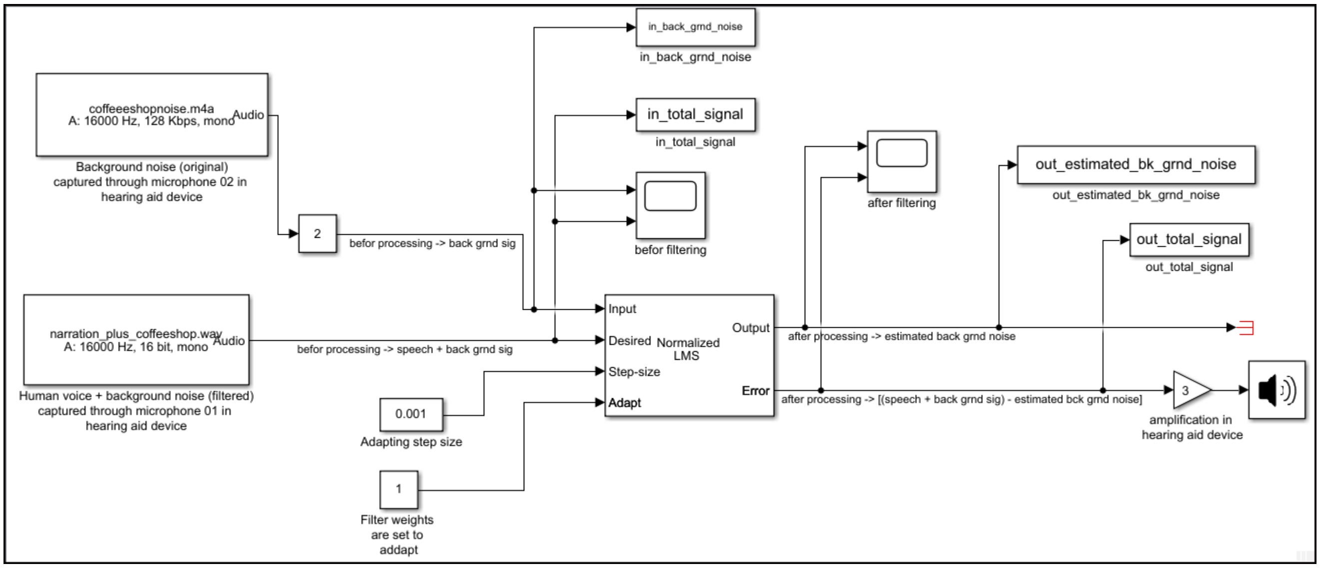


Fig. 3. Simulink model

A. Results

We generated time domain plots representing the desired signal (depicted in blue) and the background noise (illustrated in yellow) as in Fig. 4. The time domain signals of the output (yellow - representing the estimated noise) and the error signals (blue - derived by deducting the estimated noise from the desired signal) are plotted as in Fig. 5. A noticeable reduction in background noise is evident when Figure 4 is compared with Figure 5.

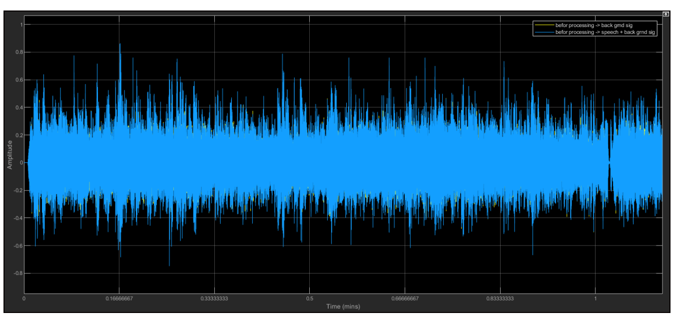


Fig. 4. desired signal (blue) and background noise (yellow)

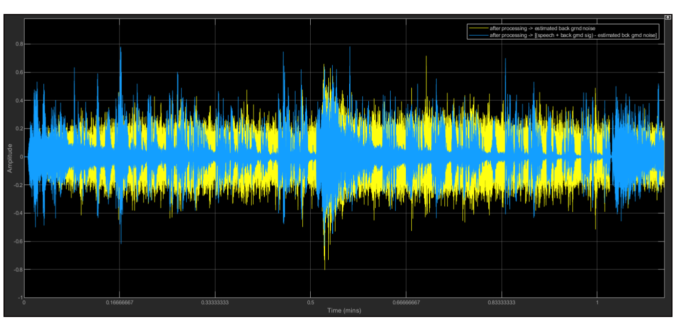
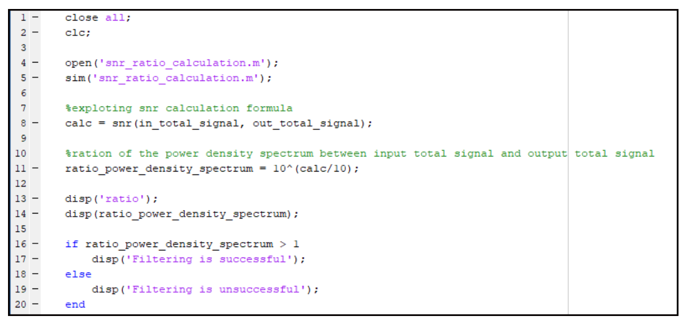


Fig. 5. error signal (blue) and estimated background noise (yellow)

We applied this methodology (Fig. 6) to quantitatively evaluate the quality of the final output of the model $(y_k - \hat{n}_k)$. Specifically, this assessment investigates the ratio of the "Power Density Spectrum" between the "Desired signal" and the "Error signal". An outcome exceeding 1 signifies the level of success achieved in the filtering process.



```
close all;
clc;

open('snr_ratio_calculation.m');
sim('snr_ratio_calculation.m');

%exploting snr calculation formula
calc = snr(in_total_signal, out_total_signal);

%ration of the power density spectrum between input total signal and output total signal
ratio_power_density_spectrum = 10^(calc/10);

disp('ratio');
disp(ratio_power_density_spectrum);

if ratio_power_density_spectrum > 1
    disp('Filtering is successful');
else
    disp('Filtering is unsuccessful');
end
```

Fig. 6. code

VI. Future Works

In our current model, we utilized two multimedia files to simulate the input audio samples used in the adaptive filter. However, in real-world applications, the plan is to utilize two microphones for this task. One microphone will record both human speech and background noise, while a second dedicated microphone will capture solely the background noise during pauses in speech. Our intention is to further explore this methodology for future investigative endeavors.

VII. Conclution

Apart from advanced technological hearing aids, the majority of hearing aids available in the market do not integrate this form of background noise filtering. Our objective is to investigate an efficient hearing aid device that incorporates not only amplification but also background noise filtering. This advancement aims to enhance hearing capabilities within educational environments.

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