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

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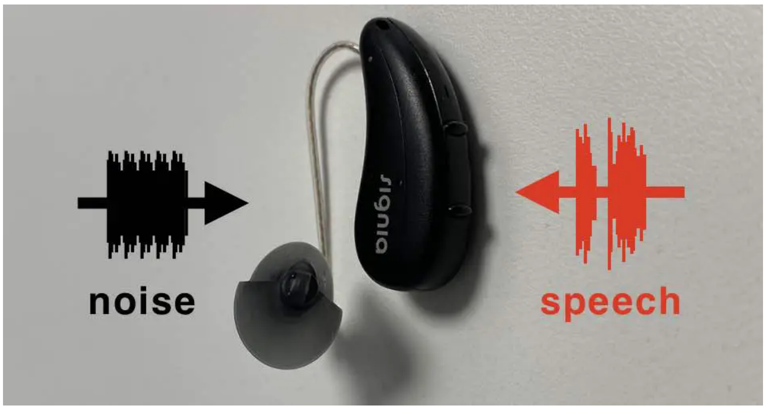
Abstract

Human speech, a pivotal aspect of communication, often faces challenges arising from unwanted noise in speech signals. In this project, we introduce a proposed system that leverages adaptive filtering techniques to effectively eliminate extraneous noise. By optimizing incoming auditory input, the system enhances the clarity of speech, thereby overcoming obstacles posed by ambient noise. The primary objective is to facilitate improved learning experiences, with a particular focus on benefiting individuals with hearing impairments within educational environments. The incorporation of adaptive filtering technology is crucial for addressing the unique needs of individuals facing hearing challenges, ultimately elevating the quality of transmitted speech signals. The positive impact extends beyond mitigating noise, creating a more conducive learning environment. This technology not only addresses communication barriers but also contributes significantly to fostering inclusive educational spaces. Through the elimination of auditory impediments, our system stands as a valuable tool for advancing accessibility and communication in diverse learning settings.

1 Introduction

Human speech is a fundamental aspect of communication, serving as the cornerstone of our interactions and a conduit for expressing emotions, sharing knowledge, and conveying thoughts. This significance is especially pronounced in educational environments, particularly within university settings, where the seamless transmission and reception of spoken information are paramount for effective learning and academic success. However, the comprehension of human speech can encounter obstacles, primarily stemming from unwanted noise in speech signals. This extended abstract delves into the development and significance of a Human Voice Filtering System tailored for integration into Hearing Aid devices. The central objective of this innovative system is to enhance learning environments by ensuring uninterrupted comprehension of spoken language, a critical factor in educational contexts. To effectively address these challenges, the proposed system employs an adaptive filtering mechanism meticulously designed to eliminate extraneous noise from speech signals. Adaptive filtering techniques are utilized to optimize incoming auditory input, resulting in a clearer and more intelligible signal. This enhanced audibility contributes to an improved learning experience, with particular benefits for individuals with hearing impairments. The relevance of this technology within the realm of hearing aid devices cannot be overstated. 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. This technology is poised to redefine the landscape of educational accessibility. By mitigating the challenges posed by extraneous noise in speech signals, the Human Voice Filtering System stands as a transformative tool for creating inclusive learning environments. The adaptability of the filtering mechanism ensures its efficacy across diverse auditory environments, making it a versatile solution for individuals facing varying degrees of hearing challenges. In conclusion, the Human Voice Filtering System represents a groundbreaking advancement in addressing communication barriers within educational settings. By enhancing the clarity of speech signals, this technology not only facilitates improved learning experiences but also reinforces the importance of inclusivity in education, paving the way for a more accessible and equitable academic landscape.

2 Utilization active Noise Cancellation in Hearing Aid Devices

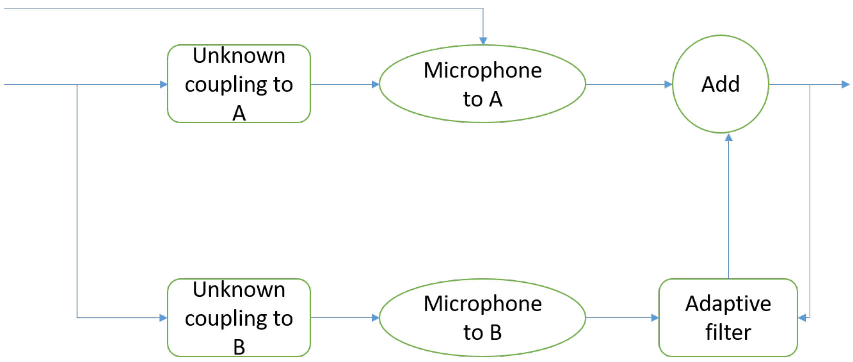


Figure 1: Simulink model

This research paper is dedicated to exploring the implementation of active noise cancellation (ANC) within hearing aid devices, presenting an innovative approach that leverages adaptive filtering to augment the quality of captured auditory signals [1]. The proposed mechanism adopts a dual-microphone setup: one microphone captures the amalgamated human voice and background noise (Signal 01), while the second microphone exclusively captures the background noise (Signal 02). The fundamental concept revolves around the estimation and subsequent subtraction of the background noise component from the primary captured signal. This process is orchestrated to achieve a refined, noise-reduced auditory output specifically tailored for hearing aid applications. By discerning and eliminating unwanted noise, the system ensures that the wearer receives a clearer and more intelligible representation of the intended auditory information. This research contributes to the evolving field of hearing aid technology by introducing a sophisticated method that goes beyond mere amplification, actively addressing the challenge of ambient noise. The utilization of adaptive filtering techniques adds a dynamic dimension, enabling the system to adapt to changing acoustic environments and providing a tailored solution for individuals with varying degrees of hearing impairments. The integration of active noise cancellation into hearing aid devices showcases promising potential for significantly improving the auditory experiences of users, marking a notable advancement in the quest for more effective hearing solutions.

3 Active Noise Cancellation and Adaptive Filtering Methodology

The methodology employed in this study integrates active noise cancellation (ANC) with an adaptive filtering system to enhance the quality of auditory signals captured by hearing aid devices. The key focus is on deducing and mitigating background noise from the combined Signal 01, which encompasses both human voice and ambient noise.

The adaptive filtering system plays a pivotal role in this process by accurately estimating the background noise content solely from Signal 01. This estimation is facilitated by utilizing principles of adaptive filtering, allowing the system to dynamically adjust its parameters based on the incoming auditory input. The secondary microphone, capturing exclusive background noise (Signal 02), functions as a reference for the adaptive filter. This reference signal enables the system to discern and isolate the unwanted noise component from the primary Signal 01 effectively.

Following the estimation of the background noise, a subtraction operation is executed, wherein the estimated noise component is subtracted from Signal 01. This results in a processed signal that effectively mitigates the background noise, thereby enhancing the clarity of the captured human voice. The derived signal, post-subtraction, represents the desired output for hearing aid devices.

The significance of this methodology lies in its ability to provide an improved auditory experience for users of hearing aid devices. By actively reducing interference caused by ambient noise, the derived signal ensures a clearer and more intelligible representation of the human voice. This advancement is particularly crucial for individuals with hearing impairments, as it addresses the common challenge of background noise interference in real-world environments. The synergy between active noise cancellation and adaptive filtering not only marks a technological breakthrough in hearing aid design but also holds the potential to significantly enhance the overall quality of life for individuals reliant on such devices.

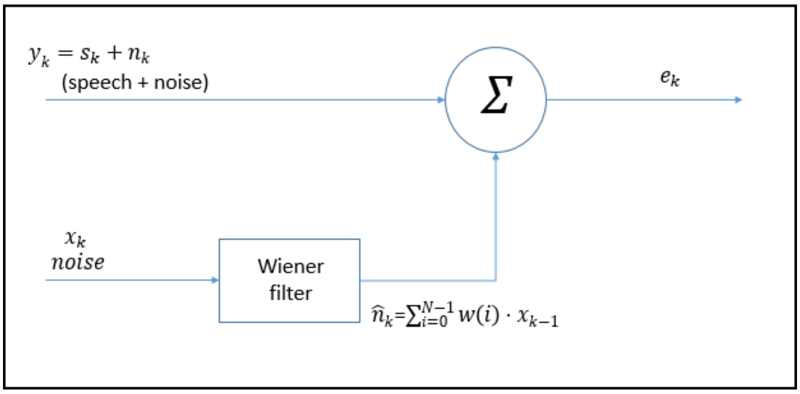


Figure 2: wienier filter

4 Understanding Adaptive Filtering in the System

The pivotal attribute characterizing an adaptive filter is its dynamic and self-modifying nature over time. Typically configured as a Finite Impulse Response (FIR) filter, the adaptive filter incorporates an algorithm that iteratively adjusts its filter coefficients. This continual adaptation process aims to minimize an error signal based on a specific criterion.

Structured as an FIR filter, the adaptive filter ensures that its coefficients evolve in response to variations in input signals. This dynamic adjustment optimizes the filter's performance, enhancing its ability to accurately capture and process changing signals. The adaptability of the filter is particularly crucial in scenarios where input signals are subject to fluctuations, such as the dynamic and unpredictable nature of background noise in speech signals.

Understanding the "Wiener filtering theory" is instrumental in deciphering the function of the "Adaptive filtering block" utilized in the Simulink model. Wiener filtering theory, a well-established concept in signal processing, provides the theoretical foundation for adaptive filtering. It involves the minimization of the mean square error between the desired signal and the filtered output, guiding the iterative adjustment of filter coefficients in the pursuit of optimal signal processing.

In the context of the proposed system, the adaptive filtering block in the Simulink model harnesses these principles to dynamically refine its coefficients, ensuring effective noise reduction in the captured auditory signals. This understanding of adaptive filtering theory, rooted in Wiener filtering principles, is paramount for comprehending the intricacies of the system's functionality and its capacity to adapt in real-time to optimize the clarity of transmitted speech signals.

5 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 [1]. 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.

6 Implementation through MATLAB Simulink

Our model's simulation was conducted in the MATLAB Simulink environment, a powerful platform for designing and testing dynamic systems. The primary focus of our implementation was on incorporating an adaptive filtering block, aligning with the principles discussed in the context of the "Wiener Filter."

The core of our approach involved utilizing the adaptive filtering block to estimate the background noise signal. The adjustment of filter weights was achieved through an iterative process, employing a step size of 0.001. At the "Input" port of the adaptive filter, we fed the background noise signal captured by a dedicated microphone in the hearing aid device. Simultaneously, at the "Desired" port of the filter, we input a combination of human speech and background noise, simulating real-world conditions.

This amalgamated input allowed the adaptive filter to compute an estimation for the background noise at the "Output" port of the filter. The adaptive filtering mechanism dynamically adjusted its parameters, ensuring accurate estimation of the evolving background noise in real-time.

The expected signal of our model was derived by subtracting the estimated background noise ($\hat{n_k}$) from the "desired" signal (y_k). This subtraction operation resulted in an enhanced signal that effectively mitigated the background noise, emphasizing the system's noise reduction capability.

Subsequently, the amplified expected signal ($y_k - \hat{n_k}$) was directed to the output speaker of the hearing aid device. This final output represented the processed auditory signal, demonstrating the successful implementation of our adaptive filtering system within the Simulink model.

Our MATLAB Simulink implementation serves as a virtual testing ground, allowing us to observe the real-time performance of the adaptive filtering system in reducing background noise and enhancing the clarity of the captured human voice. This simulation framework provides valuable insights into the efficacy of our proposed system before actual deployment in practical settings, ensuring its reliability and effectiveness in real-world scenarios.

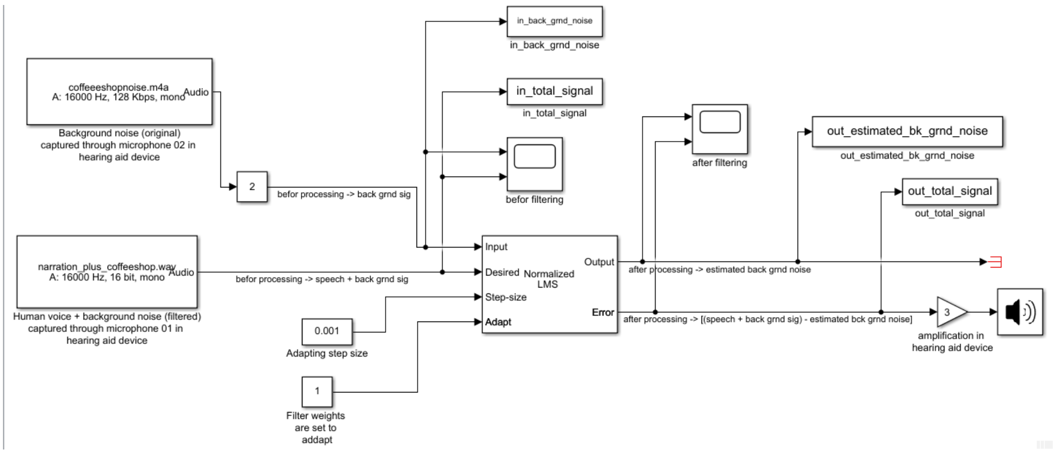


Figure 3: Simulink model

7 Results

In our study, we visualized the effectiveness of the adaptive filtering system through time domain plots, providing a clear representation of the desired signal (depicted in blue) and the background noise (illustrated in yellow) 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 further illustrated in Fig. 5. A visual inspection of these plots reveals a noticeable reduction in background noise, indicating the efficacy of our adaptive filtering approach.

To quantitatively assess the quality of the final output of the model ($y_k - \hat{n_k}$), as depicted in Fig. 6, we employed a methodology that evaluates the ratio of the "Power Density Spectrum" between the "Desired signal" and the "Error signal." This assessment serves as a robust metric for gauging the success of the filtering process.

An outcome exceeding 1 in this assessment signifies a successful reduction in unwanted noise, indicating an improved signal quality. This quantitative analysis complements the visual observations from the time domain plots, providing a comprehensive evaluation of the adaptive filtering system's performance in enhancing the clarity of the captured human voice.

These results collectively affirm the efficacy of our adaptive filtering system in mitigating background noise, demonstrating its potential to significantly enhance the overall auditory experience in hearing aid devices. The combination of visual and quantitative analyses reinforces the reliability of our approach, paving the way for further advancements in noise reduction technologies for improved hearing aid applications.

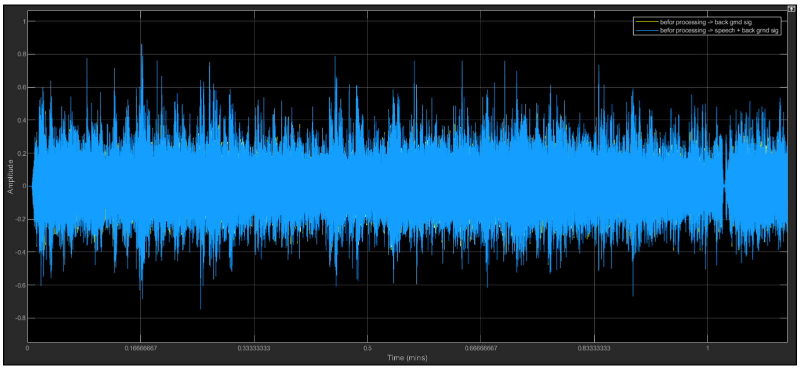


Figure 4: wienier filter

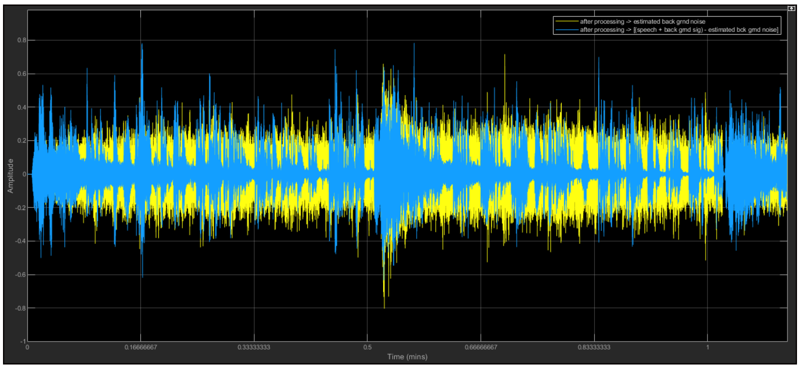
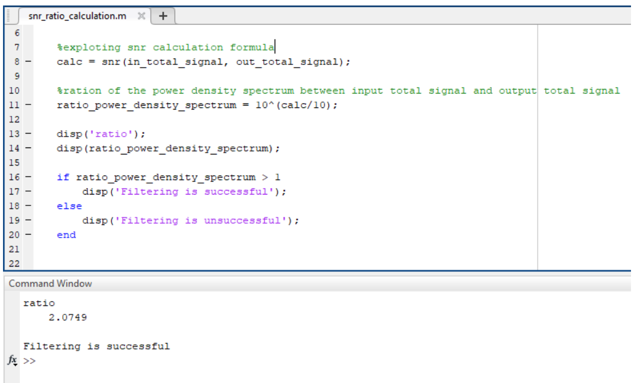


Figure 5: wienier filter




```
%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
```

```
ratio
    2.0749

Filtering is successful
>>
```

Figure 6: Matlab code to find the quality of the signal

8 Future Works

While our current model effectively utilized two multimedia files to simulate input audio samples for the adaptive filter, the next phase of our research will transition towards real-world applications. Specifically, our plan involves the integration of two microphones in the system. The first microphone will record a composite signal containing both human speech and background noise, while the second dedicated microphone will capture background noise exclusively during pauses in speech.

This shift from simulated multimedia files to a dual-microphone setup aligns more closely with practical scenarios, enhancing the authenticity and applicability of our adaptive filtering system. Real-world conditions, characterized by dynamic and unpredictable changes in speech and ambient noise, present a more complex and challenging environment. By employing two microphones, we aim to enhance the adaptability and robustness of our system to varying acoustic conditions.

The exploration of this methodology in future investigative endeavors holds the promise of refining our adaptive filtering system for even greater effectiveness in addressing the complexities of background noise interference. This progression marks a crucial step towards the development of more advanced and practical solutions, ensuring the continuous improvement of technologies designed to enhance auditory experiences, especially for individuals with hearing impairments.

9 Conclusion

In conclusion, the majority of hearing aids in the market lack advanced background noise filtering capabilities. Our research has been dedicated to exploring and implementing an efficient hearing aid device that goes beyond mere amplification, integrating sophisticated background noise filtering. This technological advancement is geared towards significantly enhancing hearing capabilities, particularly within educational environments. By addressing the challenge of unwanted noise interference, our objective is to contribute to the development of more inclusive and effective hearing solutions, ensuring that individuals with hearing impairments can experience improved auditory clarity in various learning settings.

10 Individual works

Table 1: Project Details

ID	Name	Work
210502E	RAJAKARUANA R.H.M.N.D	Active noise cancellation utilizing in the model (Part II intro). Active noise cancellation and adaptive filtering in the system (Part II.A). Conclusion writing (VII)
210506U	RAJAPAKSHA R.L.P	Abstract. Identifying how to align adaptive filtering in the proposed system. Matlab Simulink model design. Final results analyzing
210507A	RAJAPAKSHA R.W.M.A.B	Introduction (Part I). Designing Matlab code to determine the quality of the signal. Winer filter theory accommodated in the system (Part IV)

References

[1] Ziming Qi and Tom Moir. *An Adaptive Wiener Filter for Automatic Speech Recognition in a Car Environment with Non-Stationary Noise*, pages 299–315. Springer Berlin Heidelberg, Berlin, Heidelberg, 2008.