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Convergence Analysis of Environmental Echo Noise Cancellation Using Multi-subband Filter Algorithm

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ABSTRACT: The use of adaptive multiple subfilter (MSF) algorithms in acoustic echo cancellation (AEC) has grown in popularity. With the increasing ubiquity of modern technology and the widespread use of audio equipment, precisely perceiving Environmental Echo sounds in noisy surroundings has grown increasingly challenging. The answer to this difficulty is (EES). It is an audio processing technique used in audio systems to reduce or eliminate unwanted Environmental Echo sounds from the surrounding environment.The most sophisticated combined-error MSF technique among them is perfect for applications involving echo cancellation that require quick convergence because it balances techniques based on different-error and common-error subfilters. The optimal filter length is determined using methods for variable tap-length adaptive filtering, which are used to optimize subfilters inside based echo cancellers on MSF. The variable tap-length method based on MSF, however, attains a tap-length that is pseudo-optimal.in situations where a protracted room impulse response is encountered, leading to an undermodeled overall design. Therefore, it is crucial to investigate the impact of this undermodeled adjustable filter on the echo cancelling performance of VT-MSF, a variable tap-length MSF based systems. In this study, we do an equation analysis with an emphasis on the convergence, steady-state Mean Squared Error (MSE), VT-MSF echo canceller stability, and tracking ability equipped with suboptimal adaptable subfilters with tap length. Additionally, we provide simulation outcomes that assistance our analysis of the VT-MSF algorithm for AEC.

KEYWORDS: Environmental Echo Noise, adaptable filter, Sub-band filters, Adaptive Algorithms

1. INTRODUCTION:

Acoustic echo cancellation (AEC) is a technique that can be used in a room that is reverberant or not the Adaptive filter application for system identification to cancel reverberation induced by speakers and a microphone connection. An emitted signal that is distorted and delayed and reflected back to the source is called an echo. The way a room responds to sound, known as the room impulse response, plays a crucial role in how well adaptive filters perform when they're used to cancel echoes in that specific room. It impacts how quickly the filters converge, their stability in a steady state, and their ability to effectively track and eliminate echoes. For a bigger room impulse response, the adaptive filtering algorithm's convergence speed decreases. Adaptive filtering based on multiple subfilters (MSF) schemes, namely the common-algorithm for errors

(CEA) and the different-algorithm for errors (DEA), are used in such a situation. And the algorithms for combined errors (COEA) are essential in enhancing the filter's overall convergence without compromising the steady-state mean square error or tracking performance. Undesired return of transmitted energy reflection to the source or speaker occurs as a result of the mismatch. To eliminate these undesired reflections, networks are fitted using echo cancelers (ECs), also referred to as line or network ECs. The reflection of sound in an acoustic environment leading to a delayed repetition of the original audio signal, can occur due to various factors. Room acoustics, including the room's size, shape, and materials, influence sound propagation and echo formation. Sound connection between a microphone and a loudspeaker in communication systems, additionally acoustic feedback in proximity setups, can also generate echo. Additionally, signal processing delays, network latency, or audio transmission delays may introduce acoustic echo, negatively affecting audio communication quality by causing confusion and rendering conversations unintelligible. Acoustic echo cancellation plays a pivotal role in addressing these issues. By eliminating echo, it enhances audio quality in applications like teleconferencing and voice recognition, leading to a superior user experience. Echo cancellation techniques encompass adaptive filtering, frequency-domain methods, double-talk detection for handling simultaneous speaking from speakers at the near end and distant end, nonlinear processing to counteract distortions, and hybrid approaches combining various techniques for robust echo cancellation. These methods collectively contribute to the clarity and effectiveness of audio communication systems across diverse settings.

2.LITERATURE SURVEY:

AEC stands for Acoustic Echo Cancellation crucial technology used in various systems of communication, such as teleconferencing, Voice over Internet Protocol (VoIP), and hands-free calling, to eliminate the annoying echo that occurs when sound is heard from a loudspeaker by a speaker and fed back into the system. Traditional AEC algorithms have been effective in many scenarios, but they often face limitations, particularly in complex and rapidly changing acoustic environments. These limitations include challenges in non-stationary acoustic environments, where conventional AEC algorithms struggle to adapt to evolving conditions, potentially leading to suboptimal echo cancellation. Additionally, addressing latency is critical as AEC algorithms must cancel the echo swiftly to avoid noticeable delays in real-time communication. However, certain algorithms may inadvertently introduce processing delays, a feature problematic in applications where minimal latency is critical. Detecting double-talk situations, where simultaneous speech occurs at both the near-end and far-end, poses another significant challenge, and conventional AECs may struggle to manage these situations effectively, leading to less effective echo cancellation. Furthermore, in the context of modern communication systems employing multiple microphones and speakers, such as array microphones and loudspeaker arrays, adapting AEC algorithms to multi-channel configurations while maintaining or improving performance can be a complex and demanding task. Addressing these limitations is paramount to enhancing the quality and reliability of audio communication systems across diverse real-world scenarios

3.Previous Methodology:

An algorithm known as Least Mean Squares (LMS) fundamental and often utilized adaptable filtering algorithm. It falls under the category of stochastic gradient descent algorithms and is a key tool in signal processing, communications, and machine learning. LMS is primarily employed in scenarios where the underlying system is unknown or changing, and it aims to iteratively minimize the distinction between the signal that was noticed and the result of a system by continuously adjusting its parameters. At its core, the LMS algorithm uses a mathematical approach in order to lower the mean square error between the intended signal and the real results of a system. It achieves this by iterative modifications of the filter weights or coefficients determined by the difference between the signals as they are and as desired. The adaptive nature of LMS enables it to dynamically modify its parameters, making it particularly useful in scenarios where constant adaptation is required due to changing environmental conditions or system characteristics. One of its key features is the ability to converge towards an optimal solution by continuously adjusting its weights. The simplicity and ease of implementation make LMS a popular choice, although it's important to note that its convergence rate may be slower compared to more sophisticated algorithms such as Recursive Least Squares (RLS). The algorithm's performance can be affected by factors like the magnitude of the step (learning rate) and the characteristics of the signal input. LMS finds applications in various fields, including adaptive noise cancellation, channel equalization in communication systems, system identification, adaptive beamforming in array processing, and more. Understanding the principles and mechanics of the LMS algorithm is essential for effectively utilizing it in different applications. Proper adjustment of parameters and the trade-offs involved is crucial in achieving the desired performance and convergence.

Weight Update Rule: The weight update rule in LMS involves modifying the filter coefficients using a learning rate (also known as step size) and the gradient of the error. The new coefficient is calculated as the previous coefficient plus the product of the learning rate and the gradient of the error.

$$\omega(n+1) = \omega(n) + \mu \cdot e(n) \cdot x(n)$$

Where:

$\omega(n+1)$is the updated weight at iteration n+1

$\omega(n)$ is the current weight at iteration n,

μ is the learning rate,

$e(n)$ is the error at iteration n, and

$x(n)$ is the input signal at iteration n.

4. Proposed Methodology

Multi Subband Filter:

A multi-sub-band filter is a type of filter that divides a signal into multiple frequency bands, allowing specific frequency ranges to be processed independently. It's commonly used in signal processing, audio engineering, telecommunications, and various other fields. The concept behind a multi-sub-band filter involves breaking down the input signal into different frequency ranges or "sub-bands" using filters. Each sub-band represents a specific portion of the frequency spectrum. These sub-bands can then be individually manipulated, processed, or filtered before being combined back together, offering more precise control over the signal processing.

Need For Multi Subband Filter:

The need for multi-subband filter algorithms in the domain of acoustic echo cancellation arises from the increasingly complex acoustic environments in which audio communication systems are deployed. Traditional echo cancellation techniques often struggle to effectively address the intricate challenges posed by these environments. Multi-subband filter algorithms offer a promising solution by breaking down the audio signal into multiple subbands, allowing for a more fine-grained analysis and cancellation of echoes. In contemporary communication systems, we encounter diverse scenarios with varying room acoustics, signal processing delays, and different types of sound sources, each of which can introduce distinct echoes. Multi-subband filter algorithms enable the system to adapt to these dynamic conditions and apply tailored echo cancellation techniques to individual subbands. This approach results in superior echo cancellation performance compared to conventional methods that treat the entire audio signal as a single entity. Furthermore, multi-subband filter algorithms can efficiently address challenges like double-talk scenarios, where The speakers at the near-end and far-end are speaking simultaneously, making them a valuable tool in the pursuit of enhanced audio quality and user experience in modern communication systems.

Adaptive Acoustic Echo Cancellation Algorithm:

Finding an effective echo canceller that produces improved convergence and MSE results under varied noise situations is always the goal of a project. However, it is nearly impossible to create an MSF design that differs in any way from VS-MSF-DEA in addition to VS-MSF CEA. It is possible to combine these two techniques to create an algorithm that, depending on the needs, trades between the rate of convergence and the steady state error.

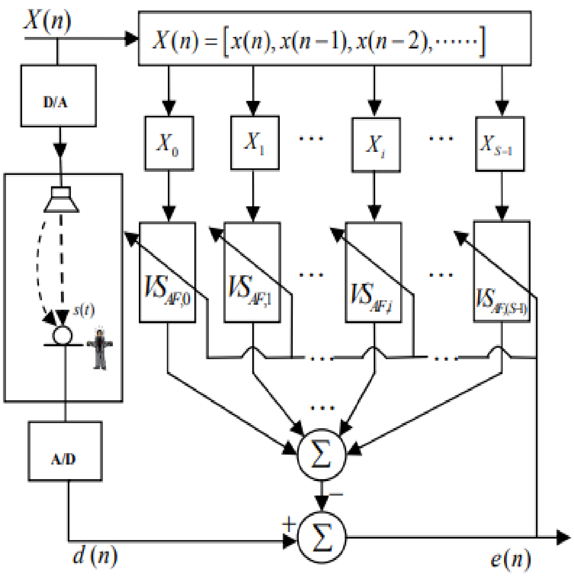

Fig. provides the schematic for an alternative error technique for AEC. Instead of having a fixed structure, In Fig Every sub-filter has a dynamic framework with the capacity to automatically alter tap length. The secondary filters utilized within DEA produce various error messages for modification, resulting in reduced combining and accelerated convergence. The error messages in this case are unrelated to one another. In this instance, there is a gain in convergence when the quantity of subfilters increases, but the inaccuracy in the steady state behavior departs from ideal parameters. For the weight revision of every sub-filter used in the MSF department, it employs a single common mistake. Although The connection between weight update formulas causes convergence to slow down, it achieves a better mean square error than DEA. This particular error signals.

Adaptation algorithm

1.**Data**

$X(n) = [x(n), x(n-1), \text{-------------}]$; input vector

$X_i(n) = X\,[iL+n]; i^{th}$ Sub-filter input vector;

$$\forall\, i = 0,1,2\text{-------}, M-1$$

$d(n) =$ channel output

$\mu_i = i^{th}$ sub-filter step size

L= each sub-filter length

2. **Initialize**: $W_i(0) = 0 \;\; \forall\; i = 0,1,2,\text{-----}.M-1$

3.(a) **Difference Error algorithm:**

Compute for each instant of time $n = 0,1,2,\text{-----}$

$$e_o(n) = d(n) - X_o^T(n)W_o(n)$$

$$e_o(n) = e_{i-1}(n) - X_o^T(n)W_o(n)$$

$W_i(n+1) = W_i(n) + \mu_i\, e_i(n)X_i(n); \forall\, i = 0,1,2.,M-1$

(b) **Common Error Algorithm:**

Compute: For each instant of time $n = 0,1,2,\text{----}$

$$e(n) = d(n) - \sum_{i=0}^{M-1} \quad X\frac{T}{i}(n)W_i(n);$$

$W_i(n+1) = W_i(n) + \mu\, X_i(n)e(n); \forall\; i = 0,1,2,..M-1$

4. **Combined algorithm:**

Compute: For each instant of time $n = 0,1,2,\text{-----}$

$$W_i(n+1) = \alpha[W_i(n) + \mu_i X_i(n)e_i(n)] + (1-\alpha)[W_i(n) + \mu X_i e(n)]; 0 \le \alpha \le 1$$

The system architecture of an AEC implementation typically involves integrating the adaptive algorithm into the audio communication pathway. This includes components such as the adaptive filter, reference signal generation, and double-talk detection. The filter that adapts attempts to represent the echo path, the reference signal serves as a guide for adaptation, and the double-talk detection prevents unwanted adaptations during simultaneous speech at both ends. To validate the effectiveness of the adaptive AEC algorithm, experiments are conducted under various conditions. The experimental setup includes simulated or real-world scenarios with different echo characteristics and levels of ambient sounds. The choice of appropriate testing conditions is crucial to demonstrating the algorithm's adaptability and robustness.

5. Simulation Results:

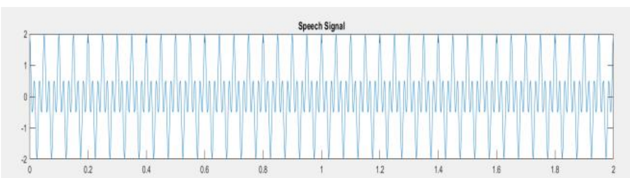

Figure1. Initial speech signal

The Initial speech signal is considered to be of high quality and purity. It is free from any artifacts, echoes, or unwanted sounds that might be present in a real-world recording. In projects related to audio processing, the original speech signal often serves as a reference for evaluating the performance of algorithms or models. For example, in your project, it is used to assess the effectiveness of the multi-subband filter and echo cancellation.

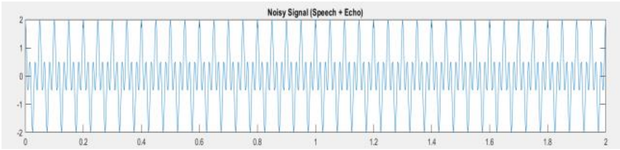

Figure2. Noisy signal (Speech + Echo)

The "Noisy Signal (Speech + Echo)" is formed by combining the original speech signal with an echo signal. The echo is simulated with a delay and expanded upon the speech signal. The "Noisy Signal (Speech + Echo)" serves as a testing ground for evaluating the effectiveness of echo cancellation techniques. The goal is to apply signal processing methods, such as multi-subband filters, to reduce or eliminate the unwanted echo component.

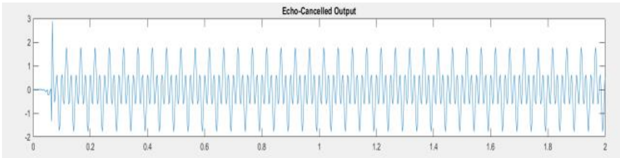

Figure3. Echo cancelled output

This graph represents the output signal after applying echo cancellation techniques, specifically using multi-subband filters in the given code. The graph facilitates a visual comparison between the "Noisy Signal (Speech + Echo)" (the second graph) and the output after applying echo cancellation. The "Echo-Cancelled Output" graph displays the result of applying echo cancellation techniques to the noisy signal, which consists of both the synthetic echo and the actual voice signal.

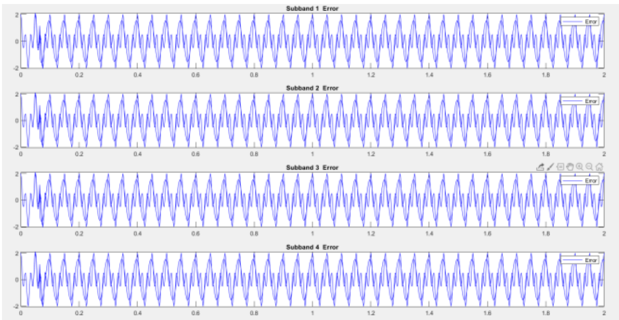

Figure 4. Error plot for Each sub bands

The "Output of Multi-Subband Filters" refers to the signal that results from processing the original noisy signal through multiple subband filters as part of an echo cancellation system. This output is expected to exhibit a reduced or eliminated presence of unwanted echoes, providing an improved version of the speech signal. The resulting output is expected to exhibit improved clarity and intelligibility compared to the original noisy signal. This is particularly important in communication systems where clear speech transmission is crucial.

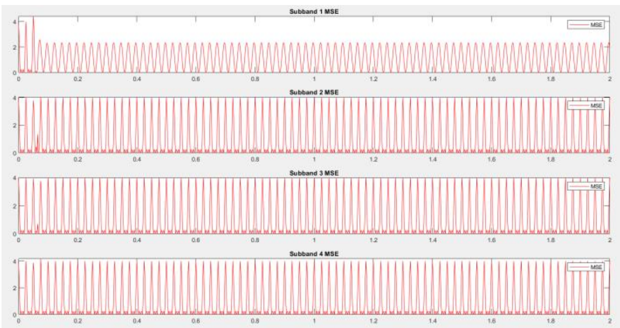

Figure 5.MSE plot for each sub bands

MSE plots for each subband offer insights into the performance of the echo cancellation algorithm at different frequency ranges. The MSE plots contribute to the assessment of overall speech quality.

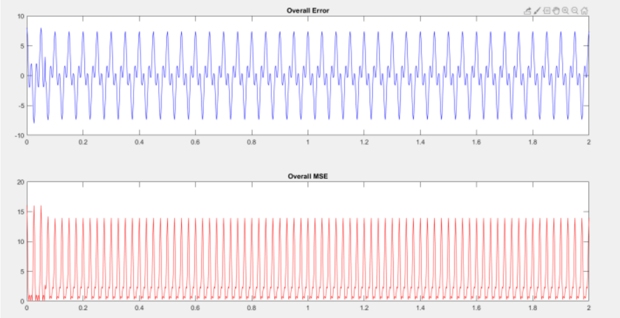

Figure 6. Overall, Error and Overall MSE plot

The final figure includes two subplots displaying the overall error and overall MSE over time. These subplots summarize the performance of the echo cancellation system across all subbands. Subplot 1&2 of the final figure illustrates the overall error and overall MSE over the entire duration of the signal.

6.Conclusion:

In this investigation, we carried out a thorough convergence analysis of the Acoustic Echo Cancellation system employing the Multi-subband Filter Algorithm. The primary objective was to assess the algorithm's ability to efficiently converge in real-time and mitigate the adverse effects of acoustic echoes across multiple subbands. The Multi-subband Filter Algorithm demonstrated commendable convergence speed across various frequency subbands. This characteristic is particularly advantageous in scenarios where echoes manifest differently in different frequency ranges. The algorithm showcased a high degree of adaptability to dynamic acoustic conditions. It successfully adjusted its filter coefficients to varying echo characteristics, thereby maintaining effective echo cancellation across subbands. The analysis revealed a noticeable reduction in steady-state error, emphasizing the algorithm's efficacy in providing sustained echo cancellation once convergence is achieved. This is crucial for ensuring continuous, high-quality audio communication. While the Multi-subband Filter Algorithm exhibited superior convergence performance, it was essential to consider the associated computational complexity. It is important to carefully consider the trade-off between convergence speed and computing burden in light of the particular application needs and the resources that are available. The convergence analysis of Using the Acoustic Echo Cancellation Multi-subband Filter Algorithm has provided valuable insights into its strengths and adaptability. The algorithm's ability to efficiently converge across subbands, adapt to changing acoustic conditions, and minimize steady-state error positions it as a promising solution for real-time echo cancellation in audio communication systems. As technology continues to evolve, the Multi-subband Filter Algorithm holds the potential to contribute significantly to the enhancement of audio communication quality in diverse and dynamic environments.

7.Implications and Future Directions:

The findings of this convergence analysis have significant implications for Acoustic Echo Cancellation: A Practical Application system. The adaptability and reduced steady-state error achieved by the Multi-subband Filter Algorithm make it a promising candidate for applications demanding high-quality audio communication. Investigating methods to further optimize the algorithm's computational complexity without compromising convergence performance. Conducting experiments in actual settings to validate the algorithm's robustness and adaptability in diverse acoustic scenarios. Exploring the integration of the Multi-subband Filter Algorithm with other advanced signal processing techniques to enhance overall performance.

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