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Analysis of Signal Distortion Remodeling in Loud Environments

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Abstract: *Typically, disturbances during conversation will influence speech signals. In digital hearing aids, a Wiener filter is developed to reduce the noise signal that is mixed with the speech stream. By predicting the relationship between the power spectra of the speech signal that is influenced by noise and the noise signal, the Weiner filter plays a significant role in noise suppression and augmentation. The main issues with integrating Weiner filters into large communication systems are power consumption and hardware requirements. In this work, a fast Fourier transform (FFT) and inverse FFT procedure was used in conjunction with an effective Wiener filter to suppress noise. The iteration issues in the traditional Wiener filter were addressed by the proposed Wiener filter. In the suggested design, an effective inverse and multiplication operation took the place of the division operation. By modifying the suggested approach for speech signal noise degradation, an effective reduction in power and area was attained. Additionally, we looked into uniform quantization, quantization based on the A-Law and Mu-Law transformations, and quantization following noise addition. To eliminate noise, we used a variety of filtering methods.*

Keywords: *Noise Filtering, Wiener Filter, Sampling, Quantization.*

I. Introduction

A speaker, a listener, and several communication devices are used in every instance of speech communication. Speech communications occur everywhere, including in private residences, places of employment, educational conferences, seminars, professional settings, and cocktail events. Random sounds frequently obstruct the speaker's ability to communicate with the listener. These sounds may cause speech to be misheard. It is hard to name every type of noise because they are produced by several different things and can be heard anywhere. Although the features of these noises are either known or unidentified, all of them have the ability to muddle, obstruct, or alter the nature of speech signals. As a result, background noise and loud environments are likely to have an impact on a large population, particularly those who have hearing loss. Speech processing is a field of study that examines how different signal processing techniques can be used to clean up noisy speech. voice processing can take many different forms, including voice synthesis, speech coding, recognition of speech, and augmentation.

Speech communications are in demand these days thanks to the development of various broadband and multimedia applications that are being used quickly in a variety of settings. In these situations, maintaining the information in the speech signal is crucial for effective communication.

Furthermore, speech processing applications, voice communication systems, and voice recognition systems are widely used in a variety of real-time application situations thanks to inexpensive digital signal processors and memory chips. Mobile phones, microphones, and other communication gadgets that can be used both indoors and outside have also been produced. Background noise is present in these settings, which adds to the original signal's loss of speech quality and could affect the effectiveness of voice recognition systems [10].

The occurrence of diverse noises in the background in outdoor settings like busy streets, bus terminals, malls, etc. contaminates the original signal and causes inappropriate communication in immediate applications like speaker identification systems, cellphone communication, hearing devices, etc. Therefore, it is necessary to evaluate and lower noise in order to enhance speech quality. Various improvement strategies for speech signal augmentation have been introduced over the past ten years. These methods can be broadly divided into single-channel and multi-channel speech augmentation. Typically, single channel enhancement of speech can be used to identify background noise attributes, and multichannel speech augmentation can be used to lessen the impacts of reverberating. Even while multichannel methods perform far better for voice augmentation than single-channel schemes, the treatment must be performed through every microphone separately.

The goal of the speech enhancement method is to improve communication connection performance by repairing any damaged input or output signals. The distorted speech signal causes enormous problems, especially for speech recognition software. The noise present in the speech signal degrades speech quality and understandability. The phrase "intelligibility" refers to how easily the voice signal may be understood as a whole. Quality is defined as the precision of the actual content of the spoken signal. Adaptive or non-adaptive, frequency or temporal domain speech enhancement algorithms are some of the numerous methods that are employed to remove noise.

Over the past few decades have seen the introduction of a number of strategies for improving speech signals, including spectral subtraction, wavelet-based methods, model-based approaches, and methods for filtering. Additionally, spectral and temporal processing methods are subcategories of speech enhancement techniques.

In contrast to the temporal processing technique, which employs time-domain analysis to improve the overall quality of the speech signal, the spectral domain procedure processes damaged signals using the transform domain methodology.

II. RELATED WORK

Multiple filter designs have been used in communication systems in recent research to lessen and ultimately remove the impacts of ambient background noise and to improve voice quality [1-2]. Frequency Response Masking (FRM), a method based on creating low complexity, narrow transition bandwidth, linear phase Finite Impulse Response (FIR) filters, has been developed to remove high frequency noise for voice enhancement [1]. When comparing the initial and filtered speech signals, a FIR filter's impulse responses are linked to different cut-off frequencies, which reduces Mean Square Error (MSE) [3].

Adaptive filtering based on employing the Least Mean Square (LMS), Normalised Least Mean Square (NLMS), and Sign-Data Least Mean Square (SDLMS) algorithms has been applied [4] in applications where both the speech and the noise signals vary continually. Both hard and soft thresholding were utilised with the Discrete Wavelet Transform (DWT) method to denoise speech signals, with soft thresholding outperforming hard thresholding at all input SNR levels [5]. Scaling factors and weighted functions have been used to minimise residual musical noise produced by the spectral subtraction technique.

In order to address memory-based FFT computers with concurrent arithmetic processing units composed of radix multi-path delay commutators, authors [9] introduced a generalised conflict-free memory addressing system. Their suggested addressing method takes into account continuous-flow operation with the least amount of shared memory needed. Parallel high-radix processors are used to increase throughput. They demonstrate the existence of a non-conflict access to memory solution that satisfies the requirements for constant flow, changing size, increased radix, and simultaneous activities.

Traditional multiband improved speech involves two processes: first, the spectrum is divided into frequency bands, and second, speech enrichment is carried out separately within each band. Due to the impact of formants in nearby bands, the polointeraction problem in the spectral domain suppresses a small number of coefficients in the estimate of clean speech, which results in poor quality ratings. The evaluation of clean speech is conducted in the temporal domain to lessen the dominance of stronger formants over the surrounding bands. Unsuppressed speech is filtered into a number of equivalent rectangular bandwidth-based subbands, and then spectral speech augmentation using the Discrete Cosine Transform (DCT) and Minimum Mean Square Error (MMSE) is applied to each band [11].

III. Spectral Subtraction

Speech enhancement techniques have been developed since Boll S.'s [6] proposal of a noise reduction technique based on Spectral Subtraction. A widely used methodology for reducing noise is spectral subtraction [7]. Using the fast Fourier transform (FFT), the noisy voice signal is first converted in this manner from the time domain to the frequency domain. The noisy speech signal is then transformed from the frequency domain to the time domain by using the inverse FFT (IFFT), after which the noise spectrum is computed in the speech pauses and deducted from the noisy speech signal's spectrum of frequencies.

IV. Weiner Filter

Weiner filter makes use of the signal characteristics. It is also simple to construct and controls output error. The ideal Wiener filter [8] is one of the most fundamental noise reduction strategies among the many techniques that have been created. It has been defined in several ways and used in a variety of contexts. For the purpose of creating the Wiener filter, information of the spectral characteristics of the original signal and the noise is required. The Wiener filter is created to take use of the accessibility of specific statistical data like the mean and correlation properties of the original voice signal and unwelcome additional noise. The noisy speech signal is provided as input to the Wiener filter to lessen the effect of noise signal in accordance with some statistical criterion.

The Wiener filter can be used to eliminate the undesired noise signal by minimising the mean square error. There have been few attempts to demonstrate the underlying connection between noise mitigation and speech distortion, however the Wiener filter may have some negative impacts on the speech signal. The goal of changing the noise estimations using the Wiener filter and the rules for converting noisy speech signals from the time domain into the frequency domain and the reverse were taken into consideration from the described drawbacks of the noise mitigation method using a Wiener filter in order to allow an adaptation to the non-linear transmission behaviour of the eardrum of humans. In our research, we have constructed it within a single processor to render the system more effective for the domain transformation rather than using two distinct processors for FFT and IFFT.

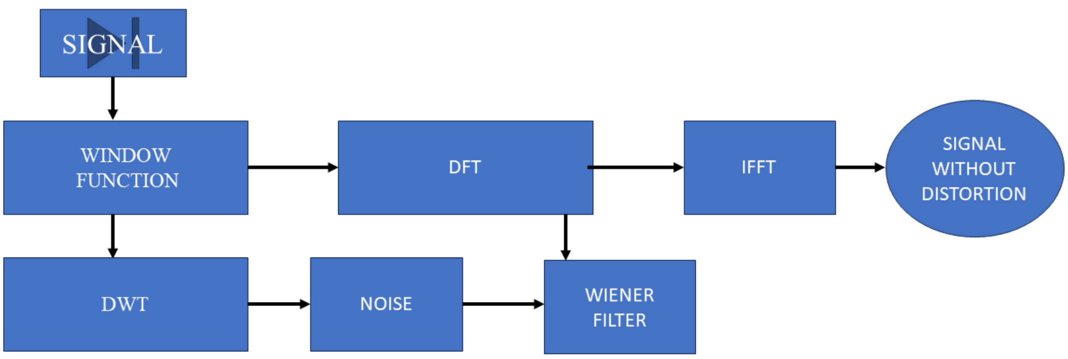

Fig. 1: Signal Distortion Removal Using Wiener Filter

V. PROPOSED METHOD

Voice signals in modern communications are tainted by a variety of sounds, which reduces voice quality and negatively affects speech recognition ability. To address these problems, modified Wiener filtering is used in a unique approach for voice augmentation, and power spectrum computation is used for damaged signals to extract the noise characteristics from a noisy spectrum.

The Gaussian distribution of each signal, including the original and noisy signals, is analysed in the following phase using the MMSE technique. For the creation of probabilistic models, the Gaussian distribution offers spectral coefficient parameters and spectrum estimation. The conversion of input signals from the time domain to the frequency domain is the first step in the recommended noise abatement methodology. The traditional FFT design for domain translation can be substituted with a revised low-power pipelined design because speech and noise signals are real-valued signals, making the entire hardware structure effective in regard to space and power usage. The FFT/IFFT processors is designed here using the primary distinction between the FFT and IFFT.

VI. DISCRETE FOURIER TRANSFORM (DFT)

The (DFT) converts discrete-time data sets into discrete-frequency visualisations, as its name indicates. The DTFT, in contrast, converts discrete time to continuum frequency. Since the produced frequency information is discrete, computers typically implement DFT designs when frequency information is needed. On modern computers, there is a method for calculating the DFT that is very quick by utilising a number of mathematical tricks and generalisations. This method, also known as the (FFT), computes far more quickly than conventional computations of DFT while still delivering the same conclusions.

VII. RESULTS

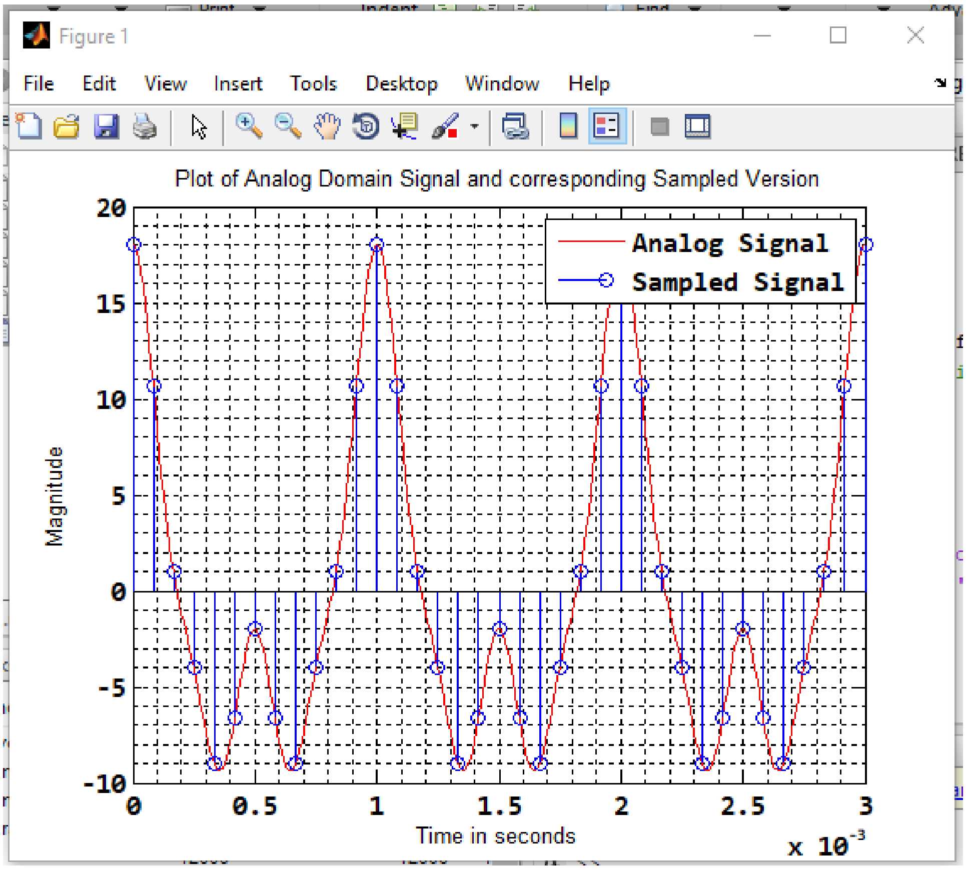

Fig.1 : Sampling of an Analog Signal

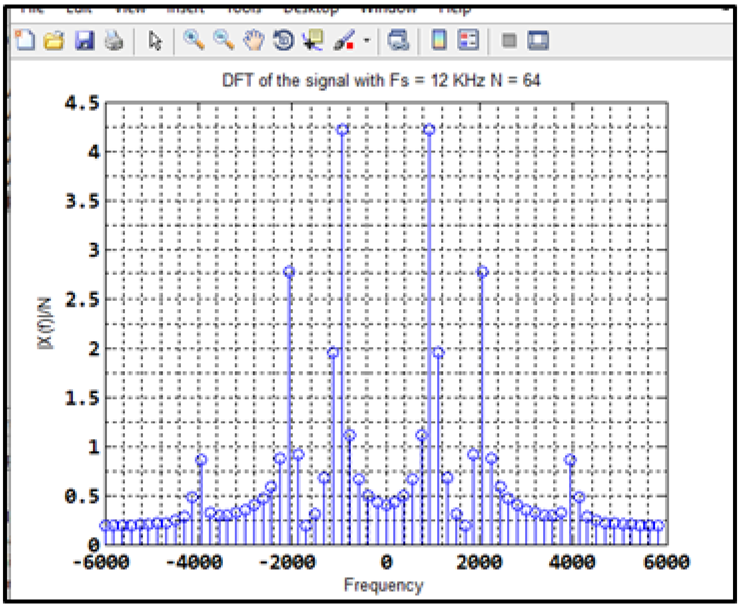

Fig.2 : Sampling of an Analog Signal Using Different Frequencies

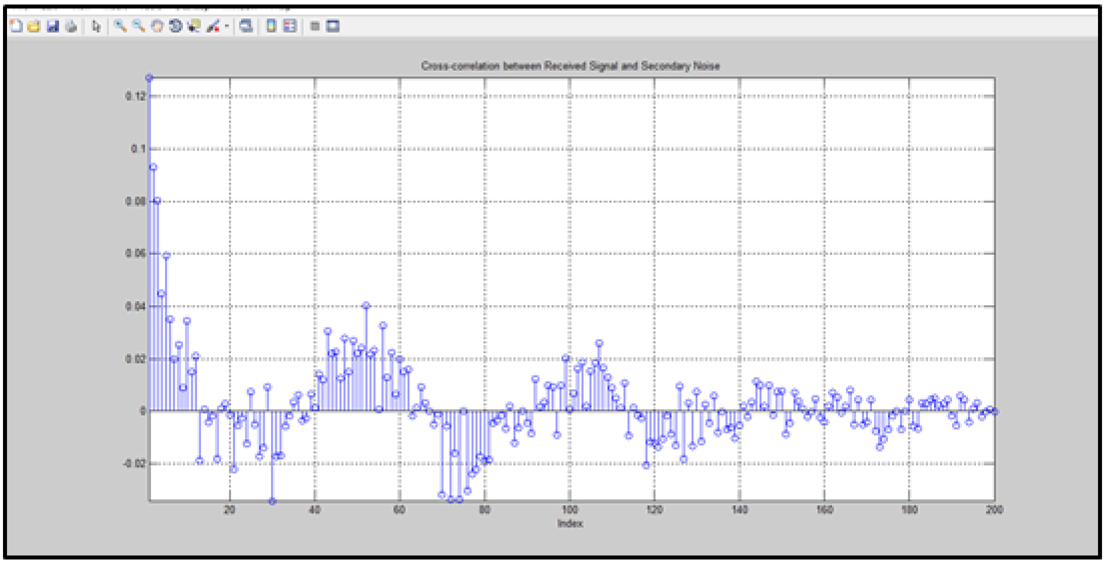

Fig.3: Relation between Signal and Noise

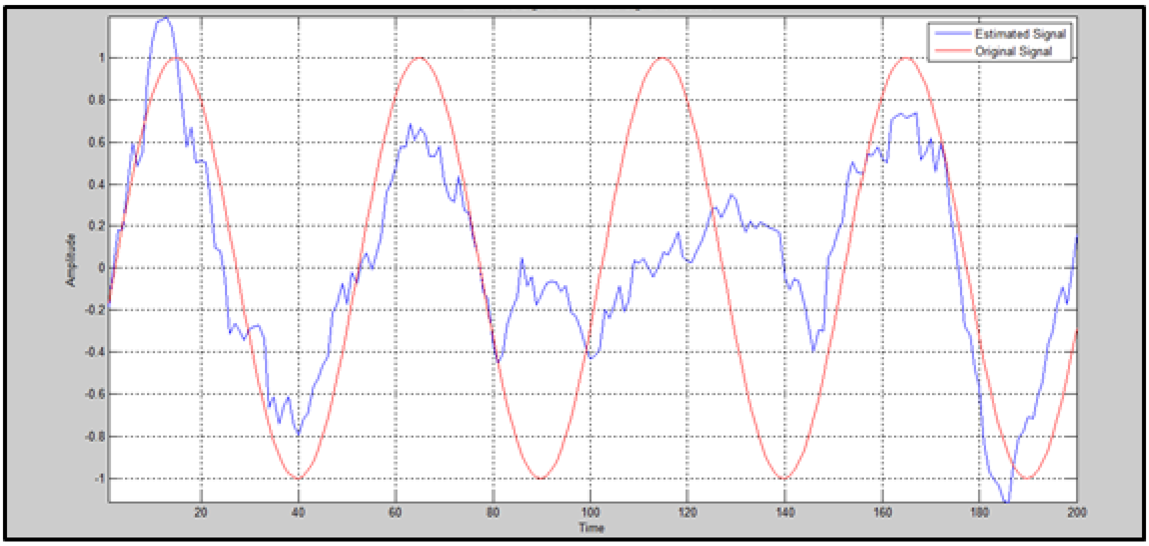

Fig. 4: Difference between Estimated and Source Signal

VIII. CONCLUSION

The proposed technique is found to have a higher objective intelligibility measured for various voiced and unvoiced signals. In order to enhance speech quality, Wiener filtering, spectral coefficient, and probability distribution models are used. Phase coefficients are saved and used again after the reconstruction while the noise characteristics are extracted. When the noise power is estimated locally, it performs better in low SNR situations. Additionally, MMSE-based noise tracking is employed when there is speech present, and SPP is used to make a soft judgement when identifying noise statistics. This enhances the performance at high SNR and provides improved noise estimation. Recursive smoothing is then utilised to produce an effective single-channel speech augmentation.

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