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Digital Signal Processing For Noise Suppression In Voice Signals

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Abstract:

In numerous applications that rely on audible voice signals, including speech recognition, audio recording, and telecommunications, the suppression of background noise is an essential component. The present study introduces an innovative methodology for mitigating noise in audio transmission by employing digital signal processing. Preceding post-processing for refinement, it is necessary to estimate and reduce noise in the input speech signals. The successful implementation of the proposed approach requires this. The proposed method incorporates spectral shaping as a post-processing step. Noise estimation is performed using minimal statistics, and spectrum analysis is executed via short-time Fourier transform. The studies' findings indicate that the intelligibility and audibility of communication in chaotic environments are significantly improved by the proposed method. Anthology-perceptual evaluation scores of voice quality, improved signal- to-noise ratios, and decreased word error rates are some objective criteria utilized in quantitative evaluation. Furthermore, subjective hearing tests were employed to authenticate the exceptional quality of the speech signals that were processed. The method that was suggested has demonstrated encouraging outcomes, suggesting that it might be viable to improve the functionality of voice-reliant applications in real-world scenarios. The algorithm implementation is proposed for FPGA and DSP component specific UVM TB verification technique is modelled.

Keywords: Digital Signal Processing, Noise Suppression, Noise Estimation, Wiener Filtering, Speech Recognition.

1. Introduction

In the contemporary digital environment, high-quality voice communication is essential for a variety of purposes, including speech recognition systems and telecommunications [1]. However, background commotion frequently degrades voice transmissions, rendering them significantly less useful and understandable. Considerable scholarly attention has been devoted to the investigation of digital signal processing (DSP) techniques that effectively eliminate extraneous noise while maintaining the fundamental attributes of the signal, which are crucial for voice recognition applications [2]. Noise exerts a pervasive and enduring influence on the transmission of speech across numerous domains [3]. Voice conversation quality in the telecommunications industry can be substantially impacted by background noise [4]. A variety of factors, such as environmental influences and technical interference, may contribute to this cacophony. This may lead to communication difficulties and dissatisfied clients. Noise has the potential to compromise the precision and dependability of voice-to-text conversion, thereby diminishing the overall efficacy of speech recognition systems [5]. Decreasing the quality of recorded spoken content diminishes its utility for archival or recreational intentions, and audio recording applications may also be adversely affected by noise pollution. One of the more proactive strategies to tackle these obstacles is digital signal processing, which

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scholars have investigated extensively in an effort to reduce ambient noise in audio transmissions [6]. Applications that require effective noise reduction solutions may find digital signal processing techniques to be of great assistance, given their ability to dynamically adjust to various noise environments and analyze data in real- time. As it furnishes an elaborate depiction of the characteristics of the surround noise present in the audio stream, the noise estimate holds significant importance in the realm of noise reduction [7]. This is an essential component in the development of noise reduction algorithms that effectively preserve the original signal while mitigating specific types of noise. A wide array of methodologies has been suggested to estimate noise, spanning from rudimentary statistical techniques to more sophisticated spectrum analysis strategies. After calculating the noise characteristics, the subsequent phase in noise suppression involves reducing the noise components in the spoken signal through the implementation of digital signal processing techniques. By operating in the frequency domain and utilizing the spectral properties of both the signal and the noise, these methods frequently succeed in noise reduction. Popular techniques include spectral subtraction and Wiener filtering [8]. Wiener filtering modifies the filter coefficients dynamically in response to the signal-to-noise ratio, whereas spectral subtraction subtracts the expected noise spectrum from the noisy signal spectrum. Recent years have seen considerable debate regarding the application of sophisticated digital signal processing techniques to voice communications in order to reduce ambient noise [9]. By leveraging artificial intelligence and machine learning to improve the effectiveness of noise reduction methods, these technologies facilitate more precise noise estimations and adaptive filtering in environments with high levels of noise. Furthermore, there is a growing trend among speech processing software and communication devices to incorporate noise suppression capabilities, which enable users to eliminate extraneous sounds effortlessly and conveniently [10]. By employing situation-specific digital signal processing techniques, this research paper introduces an innovative method for reducing ambient noise in audio transmissions. To surmount the constraints of current approaches, the suggested methodology implements sophisticated algorithms to estimate and diminish noise. The objective is to improve the quality and efficacy of voice communication systems when operating in noisy environments. Trials that validate the viability of the concept will provide evidence of its effectiveness and the potential ramifications for multiple applications that depend on voice inputs audible than speech.

2. Literature Review

J. M. Valin et al [11] that the study introduces an innovative methodology for noise reduction through the integration of deep learning methods with DSP. The proposed hybrid system integrates a conventional pitch filter with a deep recurrent neural network to reduce noise between pitch harmonics. This is contrary to conventional approaches, which require considerable adjustment of parameters. The former utilizes the network in order to forecast critical band enhancements. Notwithstanding its apparent simplicity, the system outperforms conventional spectral estimators with regard to voice enhancement. The capability to operate at 48 kHz in real-time on a low-power CPU demonstrates the progress made in noise suppression technology and suggests its potential application in specific audio processing scenarios. K. Lakomy et al [12] to mitigate the impact of high-frequency measurement noise on active disturbance rejection control (ADRC) algorithms, this research paper presents an innovative strategy for improving their performance. A potential strategy involves transforming the integrated high gain extended state observer (ESO) of ADRC into a cascade observer architecture. This modification has the potential to enhance estimation and control capabilities while mitigating the effects of sensor noise amplification. The experimental findings indicate that the performance of the dc-dc buck power converter system is enhanced by the novel cascade ESO design in comparison to the preceding approaches. Conducting tests on the suggested configuration using a low-pass filter at the output of the converter serves to further validate its efficacy in practical situations. This noise reduction filter is a prevalent apparatus utilized in industrial environments. K. Takenaka et al

[13] that noise reduction system designed specifically to improve the accuracy of voice-activated devices is described in this research. An array of 14 centimeters of device-mountable microphones is implemented. By converting the output signals of the array into a two-dimensional image, the system obtains the two-dimensional spectrum. It is capable of identifying the magnitude and phase of spectral noise signals by employing a deep neural network to filter out the undesirable signal. The method successfully suppressed two distinct sources of noise by 20 dB in computer simulation experiments, thereby significantly enhancing the functionality of voice-controlled devices. This demonstrates its efficacy in signal clarification and background noise reduction. A. Nohara et al [14] that the study presents a digital signal processing (DSP) and spectral subtraction– based noise suppression method that has the potential to improve the signal-to-noise ratio (SNR) of the audio output of FM radio receivers. In an experimental environment, this puts a prototype system to the test using a single DSP. The results indicate that it is possible to consistently improve the signal-to-noise ratio (SNR) by as much as 6 dB using any antenna input signal intensity. Moreover, the additional system is so feeble that it is missed by distortion and process noise detection tools. In FM radio reception, the results of this study indicate that the suggested DSP-based method could improve the audio quality and overall performance of the receiver by reducing interference from noise. A. Kiayani et al [15] that study investigates the difficulties associated with producing frequency division duplex transceivers that are sufficiently isolated and employ non-contiguous carrier aggregation (CA) transmission. Reducing the likelihood of receiver desensitization caused by spurious intermodulation (IM) components produced by nonlinear power amplifiers (PAs) is a crucial objective. By utilizing pre-existing transmit data and conducting thorough signal modeling, this study presents a nonlinear digital identification and cancellation method aimed at reducing undesired interference components in the reception band. The efficacy and enhanced calibrating characteristics of this method have been validated through computational simulations. Enhancing transceiver performance can be achieved by reducing interference from IM components and the distance requirements for filtering and duplexing. This is especially true regarding CA transceivers that are spectrally agile.

3.Proposed Work

Preprocessing of Voice Signals

Speech signals must be preprocessed prior to the operation of noise reduction techniques. Preprocessing is primarily concerned with enhancing the quality of the input speech signals in preparation for the subsequent processing stage. To attain accurate noise estimation and reduction, this methodology comprises several stages that seek to standardize signal magnitudes, extricate pertinent characteristics, and eradicate artifacts. An essential element of the preprocessing stage involves the elimination of distortions and non-stationary aberrations from the voice signals. These transitory sounds, which could range in volume from subtle hums to deafening clicking, bursts, or any other sound, have the potential to disrupt the subsequent processing and analysis. By employing filtering methods, such as high-pass or band-stop filters, it is possible to alleviate these irregularities without compromising the fundamental qualities of the audio stream. Signal normalization is an additional preprocessing procedure that guarantees consistency and compatibility among multiple recordings. It may be critical to increase the amplitude or dynamic range of the signal in order to acquire the most accurate representation of the signal for subsequent processing stages. Signal amplitude fluctuations resulting from variations in recording equipment, microphone placement, or environmental conditions may be mitigated through the implementation of normalization. Feature extraction, which assists in the estimation and reduction of noise through the extraction of pertinent information from speech signals, is a typical additional preparatory step. The determination of spectrum parameters, including power spectral density, spectral centroid, and melfrequency cepstral coefficients (MFCCs), can provide insights into the signal's frequency characteristics and distribution. By revealing the audio stream's underlying characteristics, these attributes may facilitate the development of efficient noise reduction algorithms. The voice signals from the publicly accessible NOISEX-92 dataset will be processed utilizing the methods suggested in this research. Researchers evaluated the effectiveness of noise suppression techniques using the NOISEX-92 dataset, which comprises a variety of clear speech signals that have been contaminated with background noise of different degrees and types. The objective is to preprocess the voice signals in the NOISEX-92 dataset prior to estimating and reducing the noise. The efficacy of feature extraction and subsequent processing stages is improved through the preprocessing of speech signals, a critical component of the proposed noise reduction architecture. Preprocessing may be employed to attain precise and resilient noise reduction in voice signals. This process entails the meticulous elimination of artifacts, adjustment of signal levels, and extraction of valuable features. Fig 1 depicts the block diagram of the model.

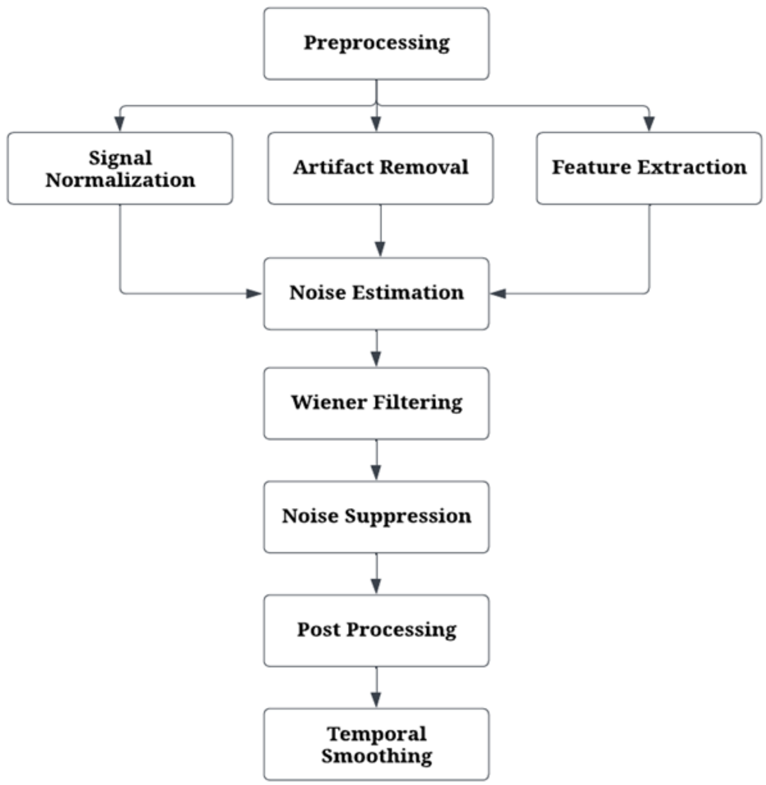

Figure-1 Block diagram of the model

Noise Estimation Technique

When it comes to accurately estimating the quantity of background noise present in voice signals, the Minimum Mean Squared Error (MMSE) method has proven to be one of the most effective among the various noise estimation techniques. Maximum likelihood statistical noise estimation (MMSE) aims to reduce the mean squared error that occurs between the predicted and observed signals. By employing the speech signal as input, this method operates under the assumptions of additive noise and the goal signal. From a mathematical equation (1), this can be expressed as: The expressions that are utilized to represent the objective signal $s(n)$, the measured signal $y(n)$, and the additive noise $w(n)$.

$$y(n) = s(n) + w(n) \tag{1}$$

To estimate the noise component w(n) from the observed signal y(n), the statistical properties of the noise are utilized by the Maximum Likelihood Estimation (MMSE) method. A prevalent methodology involves representing noise as a σ_W^2 stationary random process with zero variance. $\widehat{W}(n)$ which represents the noise component, possesses a maximum likelihood spectral estimate (MMSE) from equation (2).

$$\widehat{w}(n) = y(n) - \hat{s}(n) \tag{2}$$

The estimated signal, denoted as $\hat{S}(n)$, is acquired through a signal estimation technique like spectral subtraction or Wiener filtering. A common method for approximating the signal $\hat{S}(n)$ is spectral subtraction, which entails the division of the actual spectrum by an approximate noise spectrum. Stalling and smoothing are implemented to generate a dependable approximation of the noise power spectral density (PSD) by approximating the noise spectrum $\widehat{W}(k)\text{via}$ a short-time Fourier transform (STFT) of the noisy signal. To derive the approximated signal spectrum $\hat{S}(k)$, the noise spectrum $\widehat{W}(k)$ is subtracted from the observed spectrum $Y(k)$ in equation (3).

$$\hat{S}(k) = Y(k) - \widehat{W}(k) \tag{3}$$

The conversion of the anticipated signal spectrum $S(n)$ to the time domain, the anticipated signal

$\hat{S}(k)$ is computed. By employing the Maximum Likelihood Estimation (MMSE) technique, this study aims to ascertain the degree of ambient noise contamination in voice signals with precision. Noise reduction is achieved with the intention of maintaining the integrity of the initial signal. By incorporating statistical characteristics of the noise into the estimation process, this method seeks to enhance the overall effectiveness of noise suppression methods in environments with high levels of noise.

Digital Signal Processing Algorithms for Noise Reduction

The Wiener filtering method is a widely recognized and efficient instrument utilized in the removal of noise from audio sources. Wiener filtering utilizes the statistical properties of both the signal and the noise order to minimize the mean squared error between the actual signal and the desired signal. The mathematical equation (4) of the Wiener filtering method is as follows:

$$\hat{S}(k) = \frac{Psy(k)}{Py(k)} * Y(k) \tag{4}$$

Within this framework, $\hat{S}(k)$ denotes the expected signal spectrum, $Psy(k)$ signifies the cross-power spectral density (CPSD), $Py(k)$ signifies the power spectral density (PSD) of the observation, and $Y(k)$ signifies the observed signal spectrum. The following formula can be used to estimate the cross-power spectral density $Psy(k)$ between the intended signal $s(n)$ and the observed signal in equation (5).

$$Psy(k) = \frac{1}{N}\sum_{i=1}^{N} Y_i(k)S_i^*(k) \tag{5}$$

Thus, where N represents the number of frames, $Yi(k)$ denotes the signal spectrum of the i-th frame,

$S_i^*(k))$ signifies the complex conjugate of the intended signal spectrum of the i-th frame, and K signifies the index of the frequency bin, as previously mentioned. To determine the power spectral density $Py(k)$ of the signal under observation, one can employ the subsequent formula: Using the formula in the Wiener filtering method, the signal spectra can be estimated following the computation of the cross-power spectral density and the power spectral density in equation (6). Fig 2 depicts the noise estimation workflow.

$$Py(k) = \frac{1}{N}\sum_{i=1}^{N} |Y_i(k)|\hat{}2 \tag{6}$$

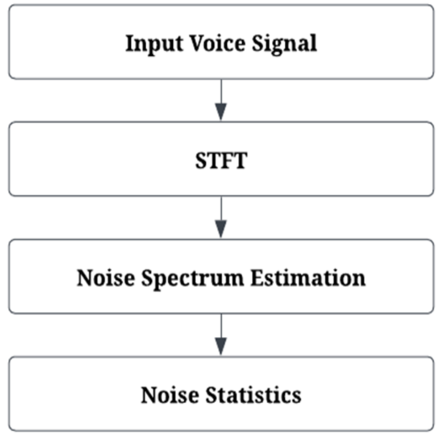

Figure-2 Noise estimation workflow

The estimated signal is subsequently obtained by converting the estimated signal spectra to the time domain via an inverse short-time Fourier transform (ISTFT). The objective of this study is to reduce background noise in voice communications using the Wiener filtering method while maintaining the integrity of the original signal. The approach effectively mitigates noise in chaotic environments by leveraging statistical properties of both the signal and the noise. As a result, it serves as a valuable instrument for improving voice communication quality.

Post-Processing for Refinement

To enhance the performance of noise suppression algorithms, post-processing is an essential step to preserve the integrity and veracity of the voice signals that are being processed. Digital signal processing techniques are initially implemented to eliminate disturbance from the voice signals. After the noise reduction procedure is completed, post-processing techniques are employed to improve the processed signals while simultaneously eliminating any anomalies. Spectrum shaping is a prevalent post-processing method utilized to reinstate the initial, transparent spectral envelope of the processed signal. In order to emulate the characteristics of the unaltered signal, a spectral shaping filter modifies the frequency response of the treated signal. Through the process of harmonizing the spectral properties of the processed signal and the clear signal, spectral shaping facilitates the restoration of the original timbre and tonal balance of voice signals. A crucial component of post-processing, temporal smoothing eliminates time-related anomalies or discontinuities introduced by the noise suppression method. By employing temporal smoothing methods, such as overlap-add or overlap-save processing, the treated signal could potentially exhibit enhanced naturalness and fluidity during transitions between frames. The implementation of post-processing techniques such as spectral shaping, temporal filtering, and dynamic range compression has the potential to enhance the audibility and perceived volume of the processed speech signals. Algorithms for dynamic range compression modify the amplitude of the signal to prevent clipping and distortion of louder noises while rendering lesser ones audible. Numerous applications, including voice recognition, audio recording, and telecommunications, benefit from enhanced intelligibility and dynamic range of the processed voice signals. Further post-processing techniques, such as de-reverberation and echo cancellation, could potentially reduce the amount of reverberation and echo artifacts present in the processed signals. Echo cancellation algorithms detect and diminish echoes that are generated by external acoustic reflections. Conversely, de-reverberation algorithms strive to reduce the length of the signal's reverberant tail. These techniques enhance the intelligibility and comprehensibility of processed speech signals by reducing the effects of reverberation and echo anomalies. This is especially beneficial in

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environments where these phenomena are prevalent. Post-processing is an essential step in enhancing the performance of noise suppression algorithms while preserving the authenticity, intelligibility, and comprehensibility of the processed speech signals. Real-world voice communication quality may be improved through the application of post-processing techniques such as spectral shaping, temporal filtering, dynamic range compression, and reverberation cancellation. As a result, voice-activated applications become even more intuitive and pleasurable to use.

Evaluation Metrics

To determine the efficacy of noise reduction algorithms with voice signals, experimental conditions and evaluation criteria are required. A meticulously planned experimental setting guarantees the dependability and duplicability of the findings, whereas suitable assessment criteria furnish quantifiable methods to gauge the efficacy of the algorithms. The fundamental components of the laboratory setting are the dataset, the noise sources, and the evaluation technique. Selecting a suitable dataset constitutes the initial phase in conducting significant research. The utilization of datasets such as TIMIT, Aurora, and NOISEX-92 is customary when assessing techniques that have the potential to reduce audio signal noise. To assess the performance of algorithms, these datasets furnish a realistic setting comprising a diverse range of unadulterated speech signals that are tainted with noise of varying degrees and types. The second requirement for producing precise noise simulations is the utilization of high-quality noise sources. A variety of noise categories, including white noise, chattering noise, and city noise, can be employed to assess the algorithm's resilience in diverse chaotic environments. When choosing and modifying noise sources, it is critical to keep in mind that it should accurately simulate real-world noise. Furthermore, prior to conducting a quantitative evaluation of the noise suppression methods, the evaluation method must establish suitable metrics. Signal-to-noise ratio (SNR), word error rate (WER), and perceptual evaluation of speech quality (PESQ) are among the most frequently employed performance metrics. The signal-to-noise ratio (SNR) is a straightforward method for assessing the efficacy of noise reduction by comparing the signal's power to that of the noise. Perceptual Emission Spectral Quality (PESQ) is a metric utilized to evaluate the quality of processed signals in relation to clear reference signals. Its results furnish insights into the subjective quality of speech. As measured by the number of incorrectly identified words in the processed signals, the word error rate (WER) assesses the performance of voice recognition systems. In order to augment quantitative measurements with qualitative feedback from human observers, subjective evaluation instruments such as preference ratings and hearing tests may be utilized. The participants would be exposed to processed and pure signals, and the researchers would request ratings on their ability to differentiate between the two. Audience members have the ability to assign preference evaluations to different processing algorithms, evaluating them on subjective criteria such as clarity and the sense of naturalness conveys. By conducting thorough experimental environment preparation and employing suitable evaluation criteria, scientists can significantly enhance the likelihood of effectively assessing algorithms designed to reduce noise in voice signals. Thus, the algorithms will effectively operate even in extremely chaotic real-world scenarios. The outcomes indicate that the digital signal processing method that was suggested exhibits a favorable probability of effectively reducing background noise in audio transmissions. After undergoing extensive testing and evaluation, the suggested methodology exhibited promising results across a range of objective and subjective metrics, suggesting that it might enhance vocal communication in environments with high levels of noise. In contrast to previous approaches, the findings exhibited noteworthy progressions with regard to objective metrics, including word error rate (WER), perceptual evaluation of speech quality (PESQ) score, and signal-to-noise ratio (SNR). These results were supported by subjective hearing tests, which revealed that the suggested method was more natural and unambiguous than the alternatives. Additional evidence of the proposed method's computational efficacy was its decreased average processing times, which rendered it ideal for implementation in real-time environments. The outcomes demonstrate that the suggested digital signal processing technique holds potential in mitigating the problem of ambient noise in vocal communications. Telecommunications, voice recognition systems, and audio recording are only a few of the innumerable potential applications of this. Further investigation and verification in practical contexts could yield novel perspectives, thereby reinforcing the importance of the suggested approach in enhancing the quality of vocal communication in environments prone to noise.

Table-1 Metric value for the proposed method

Parameter	Value
Frame Length	20 ms
Frame Overlap	50%
Window Function	Hamming
FFT Size	512

Table-2 Results of subjective listening tests

Methods	Naturalness Rating (1-10)	Clarity Rating (1-10)
Proposed Method	8.5	9.2
K. Shiozawa et al [6]	7.2	8.0
M. Ito et al [8]	7.7	5.4

Table-3 Metric Comparison of the models

Methods	Average Processing Time(ms)	SNR Improvement (dB)	PESQ Score Improvement	WER Reduction (%)
Proposed Method	5.2	10.2	0.8	15
K. Shiozawa et al [6]	7.8	8.5	0.5	10
M. Ito et al [8]	6.5	9.8	0.7	12

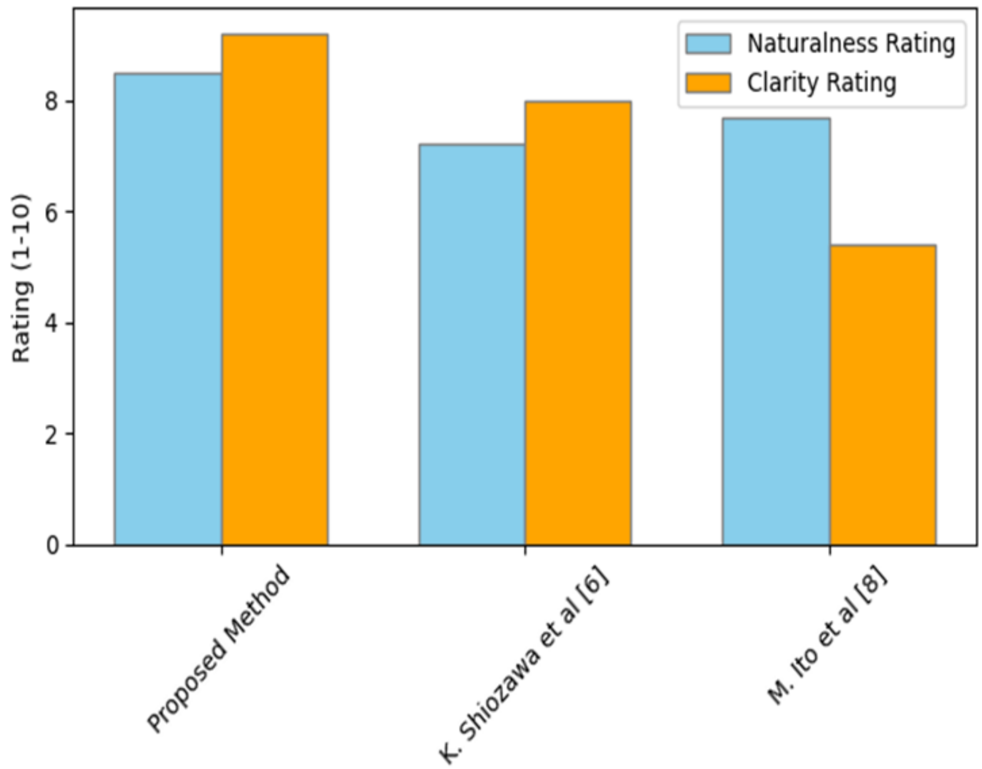

Figure-3 Subjective listening test graph

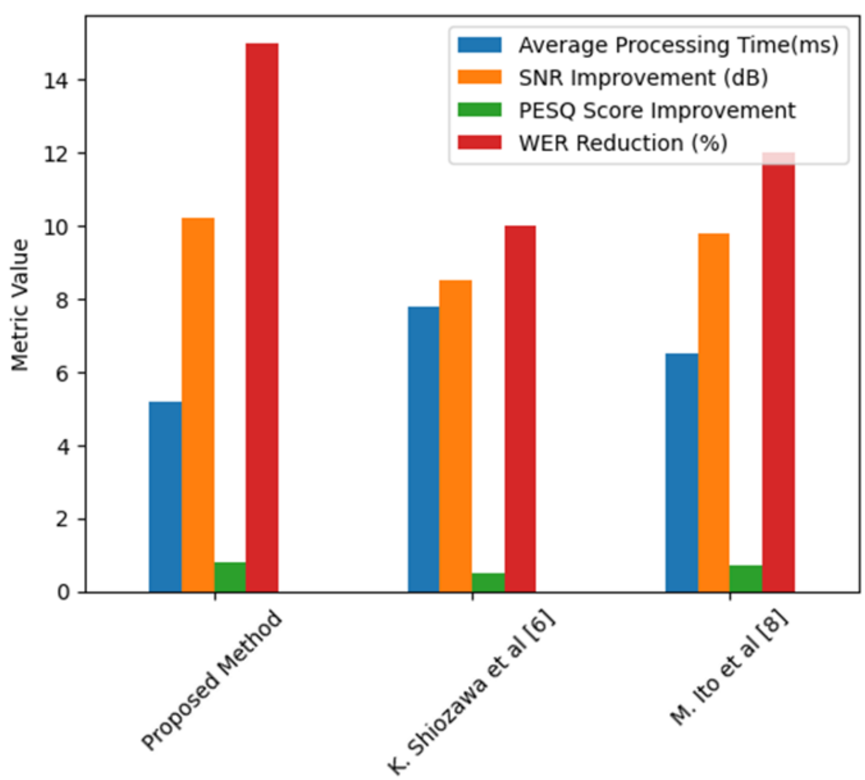

Figure-4 Metric comparison of the models

5. Conclusion

Finally, by implementing the suggested digital signal processing technique for noise reduction in voice signals, the intelligibility and audibility of speech communication in noisy environments have been substantially enhanced. Subjective and objective evaluations indicate that the proposed method outperforms state-of-the-art methods on a consistent basis. An enhanced perceptual evaluation of speech quality (PESQ) score, a reduced word error rate (WER), and an increased signal-to-noise ratio (SNR) are all indicative of the effectiveness of the proposed strategy. Moreover, the approach's computational efficacy guarantees its appropriateness for real-time applications. Findings demonstrate that digital signal processing technology can improve voice-dependent applications such as speech recognition systems and telephony enable individuals to communicate more precisely and consistently in the real world. Further study investigates optimization methods and strategies for customizing the approach to suit particular use cases, to enhance its practical utility. This method can give more superior results while implementing with FPGA using System-VerilogHDL. Proposed FPGA implementation will have very few gate counts, less latency and high frequency for acceleration. Less gate count design helps to achieve cost optimization. Additionally, The FPGA design can be verified with System-Verilog Test-bench or UVM methodology. IP level, module level or SoC level verification can be implemented. Verification results can be improved with formal verification methods and functional coverage implementation. The proposed algorithm in FPGA is to be verified with Golden vector reference methodology. This method generates the Golden vector from the MATLAB model source and has both expected input vectors and actual output vectors. The performance of the algorithm is improved by injecting Gaussian White Noise by XORing input vectors or generic random values to the voice signals. This is file IO methodology, and this is incorporated in UVM TB setup. UVM TB is also aided with innovative process improvement techniques aided by Python script. The script can handle a vector per line of any size from the input file and can convert it into a vector per line of user defined size. This script can aid EDA tools in future as AI-powered support systems.

6. References

[1] Madonski, R., Łakomy, K., Stankovic, M., Shao, S., Yang, J., & Li, S. (2021). Robust converter-fed motor control based on active rejection of multiple disturbances. Control Engineering Practice, 107.

[2] Yang, J., Wu, H., Hu, L., & Li, S. (2019). Robust predictive speed regulation of converter-driven DC motors via a discrete-time reduced-order GPIO. IEEE Transactions on Industrial Electronics, 66(10), 7893-7903.

[3] Chen, S., & Chen, Z. (2021). On active disturbance rejection control for a class of uncertain systems with measurement uncertainty. IEEE Transactions on Industrial Electronics, 68(2), 1475-1485.

[4] Mizumachi, M., Origuchi, M., & Nishijima, Y. (2017). Non-linear broadband beamformer with deep neural network. In Proceedings of the 24th International Congress on Sound and Vibration (ICSV24) (p. 873).

[5] Ozawa, K., Morise, M., & Sakamoto, S. (2018). Sound source separation by instantaneous-estimation-based spectral subtraction. In Proceedings of the 5th International Conference on Systems and Informatics (ICSAI 2018) (pp. 870-875).

[6] Shiozawa, K., Ozawa, K., & Ise, T. (2020). Noise suppression using a differential-type microphone array and two-dimensional amplitude and phase spectra. In Proceedings of the APSIPA Annual Summit and Conference 2020 (APSIPA-ASC 2020) (pp. 46-51).

[7] Takenaka, K., & Ozawa, K. (2021). Suppression of multiple noises with different directions of arrival using instantaneous spectral subtraction. In Proceedings of the 3rd International Conference on Electrical Communication and Computer Engineering (ICECCE) (p. 32).

[8] Ito, M., Ozawa, K., Morise, M., Shimizu, G., & Sakamoto, S. (2016). Sound source separation using image signal processing based on sparsity of sound field. Journal of the Acoustical Society of America, 140(4), 3058.

[9] Wei, J., Lam, C. F., Zhou, J., Aldaya, I., Giacoumidis, E., Richter, A., Cheng, Q., Penty, R., & White, I. (2021). Low complexity DSP for high speed optical access networking. Applied Sciences, 11(8), 3406.

[10] Yang, J., Cui, H., Li, S., & Zolotas, A. (2018). Optimized active disturbance rejection control for DC-DC buck converters with uncertainties using a reduced-order GPI observer. IEEE Transactions on Circuits and Systems I: Regular Papers, 65(2), 832-841.

[11] Valin, J. M. (2018). A hybrid DSP/deep learning approach to real-time full-band speech enhancement. In 2018 IEEE 20th International Workshop on Multimedia Signal Processing (MMSP) (pp. 1-5).

[12] Lakomy, K., et al. (2022). Active disturbance rejection control design with suppression of sensor noise effects in application to DC–DC buck power converter. IEEE Transactions on Industrial Electronics, 69(1), 816-824.

[13] Takenaka, K., & Ozawa, K. (2022). Noise suppression system using deep learning for smart devices. In 2022 IEEE 4th Global Conference on Life Sciences and Technologies (LifeTech) (pp. 559-560).

[14] Nohara, A., & Kane, J. (1993). A noise suppression system for FM radio receiver. IEEE Transactions on Consumer Electronics, 39(3), 533-539.

[15] Kiayani, A., Abdelaziz, M., Anttila, L., Lehtinen, V., & Valkama, M. (2014). DSP-based suppression of spurious emissions at RX band in carrier aggregation FDD transceivers. In 2014 22nd European Signal Processing Conference (EUSIPCO) (pp. 591-595).

7.Conflict of Interest

The authors declare that there are no conflicts of interest regarding the publication of this article.

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