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Enhancement of Speech Signal Using Improved FA-ANFIS Classifier for Hearing AIDS Application

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Abstract--- *Research is undergoing in hearing aids application in the sense of dealing with background noise in speech. It aims to understand the speech in the availability of background noise. It acts as challenging environment to enhance the speech understanding during presence of noise. Thus hearing aid applications had introduced to understand the speech even in presence of noise in various environments for end users. It is one type of methodology adopted for improving the user to understand the information in signal. This speech enhancement application helps to improve availability of data. Additionally, it increases the speech signal intelligibility and quality of application. This review introduced the mechanism for improving these above mentioned characteristics in applications such as hearing aids. This could be followed out using enhancement methods namely, Fast Independent Component Analysis algorithm shortly Fast ICA. This helps to reduce the noise presence in speech related signals. This process also involves IDWT techniques where IDWT stands for Improved Discrete Wavelet Transform to do feature selection. Features have been extracted from speech signal that are denoised. AANOVA stands for Advanced Analysis of Variance helps to identify the important feature from the features that are already selected in previous process. Finally, the features which are extracted as important have been given for optimization algorithm in order to get the features in optimized manner. Now lastly collected features are then classifies efficiently using ANFIS. ANFIS stands for Adaptive Neural Fuzzy Interference System is a classification approach. It also called as Improved FA-ANFIS Method. The result obtained in proposed approach proves that it would gives out promisingly better speech intelligibility and the good quality over speech signal in hearing aids commodities as compared to applications already exist.*

Keywords--- *Speech Signal Enhancement, Fast Independent Component Analysis (Fast ICA), Improved Discrete Wavelet Transform (IDWT), Firefly Algorithm, Improved FA-ANFIS.*

I. INTRODUCTION

Now days, Speech enhancement is one of the leading research area among many. Speech enhancement works on processing of speech signals especially for listening. The aim of enhancing [1] the speech signals is to increase the entire quality of signals and to separate the noise signals from speech occurs.

This process will enhance the clearness over speeches and it may decrease the ambiguity, exhaustion of hearing aid users based on application that they use. The users of this aid application found difficulty in mixture of background noise with speech that they receive.

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This Speech enhancement process focuses on separating the speech signals from mixture of signals or to extract the speech by reducing the signals that interferes and background ones [2].The signal obtained after above process could be uses directly or it may be brings out for further process based on the applications they uses. The applications may be hand free based telephony, surveillance, human computer interaction. Humans with hearing as normal could go for binaural clues in order to focus on specific sounds but it may insufficient in terms of ability sometimes. The methodologies and mechanisms used for speech enhancing process in applications of hearing aid would improve the quality and intelligibility of speech signals.

The Techniques [3] of DSP stands for Digital Signal Processing are using for the process of speech enhancement. Techniques may involves are suppression in non-harmonic frequencies, adaptive filtering and spectral subtraction. Mostly these mechanisms might require secondary microphone to give noisy reference, or may need characteristics in noise that is stationary. But the non-stationary signals are only takes place in hearing aid commodities. The main advantage of this proposing methodology is it has simple and high computational requirements. The intelligibility and quality of speech [4] has been improved by using classification algorithm that developed with the help of methods in signal processing.

The enhancement methods that includes are Fat ICA [5] stands for Fast Independent Component Analysis which is used to reduce noise from noisy signals. Then Improved Discrete Wavlet Transform shortly IDWT has been used especially for feature based extraction in noised signals of speech. AANOVA stands for Advance Analysis of Variance algorithm has been involved in order to select the important feature from the features that extracted. Finally the features obtained are bring out for optimization using algorithm named Firefly shortly FA. Then it has been classified using ANFIS [5] namely Adaptive Neural Fuzzy Interference System which is an optimized version. From the result obtained, the proposed system will gives out better intelligibility and good quality of signals especially in Hearing aid commodity as compared with existing methodologies [6].

The overview of the speech signal enhancement is provided in this part. The related work of this paper is provided in section II. The proposed technique of the speech signal enhancement is provided in section III. The performance analysis of the research technique is provided in section IV. The conclusion and future work of this paper is provided in section V.

II. LITERATURE SURVEY

The authors of [1] aims to enhance the signal of the speech while hearing thereby reducing the background noise. This effective method is implemented for effective hearing. The various techniques involving many algorithms namely, Wiener filtering, decomposition of values, beam former and spectral subtraction is employed for the enhancement of speech signal in hearing aid. PSEQ and SNR are the metrics utilized for measurement of performance.

In [2] the authors have discussed about the various methods for improving the comprehensibility and understandability of speech signal. It also involves various kinds of noise and its eradication techniques. The enhancement techniques like multi-channel and single channel enhancement has been briefed. The time domain

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methods like linear predictive coding, Winer filtering, Signal subspace method based DFT and Kalman filtering has been provided in this study.

The authors of [3] have been aimed to propose the multichannel enhancement depending on directionality. It involves the conservative strategies where the arrival in not known and the aggressive strategy for deriving fast acting post-filter for output. SNR is used for better construction of post filter. This enhances the hearing aid in a better way.

In [4] the authors defined the opportunities and challenges for the development of hearing- aid in audio-visual (AV) speech context. The speech quality and intelligibility has been enhanced by means of new modal AV algorithm. The audio visual settings have to be employed on the hearing aid evaluation for the better performance.

The authors of [5] proposed a modified classifier algorithm for sound classification and various methods has been applied for speech enhancement with hearing aid application. The Independent Component Analysis (ICA) provides the high quality hearing aid in comparison with others. The resistance to the noise has been increased by the usage of ICA algorithm along with Adaptive Neural Fuzzy Interference System (ANFIS) and is more feasible.

The authors of [6] have estimated an effective noise using algorithm in which the fusion of Gaussian model assumption and minimum statistics estimation has been performed. This model works in two steps. The pitfalls in the first stage are eliminated in the second stage by employing Bayes theorem. The performance is estimated by both subjective and objective tests in the various circumstances and the outcome is found to yields better results in comparison with conventional MS based estimate.

In [7] the authors introduced a reduced complexity integrated active noise cancellation approach in hand with the scheme for noise reduction. It offers the aid for digital hearing and enhances the speech intelligibility. DCD helps to reduce the complexity of computation. The better speech quality stands as an example for simulation.

The authors of [8] introduced speech enhancement based on Extreme Learning Machine ELM/H-ELM for the effective outcome. The ELM replaces the single layer model by the multi-layer model for the better performance. The H-ELM is in comparison with traditional speech enhancement algorithm and DDAE for the verification of consistency. The H-ELM when compared with other techniques provides better result on limited training data.

The authors of [9] proposed a deep learning methodology for the enhancement of speech signal. The deep leaning is a technique which is employed for the purpose of classification and detection. The deep learning involves the Deep Neural Network (DNN) and ADALINE. These methods are used for enhancement. The speech enhancement by ADALINE is the basic model of neural network. DNN is for improving the validity and it is the hidden network. DNN provides the efficient outcome and ADALINE is for measuring performance.

The authors of [10] propose a study on noise-robust voice activity detection (VAD) in which the various signals is utilized. The VAD algorithm is used for differentiating the voiced region of speech over unvoiced noise or signal. This is made possible by total signal ratio and energy ratio. The outcome is tested against the real speech signal.

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III. PROPOSED METHODOLOGY

A new improved FA-ANFIS Classifier with fast ICA is presented in this research of Speech Signal enhancement for Hearing AIDS Application. The proposed system flow diagram is presented in figure 1.

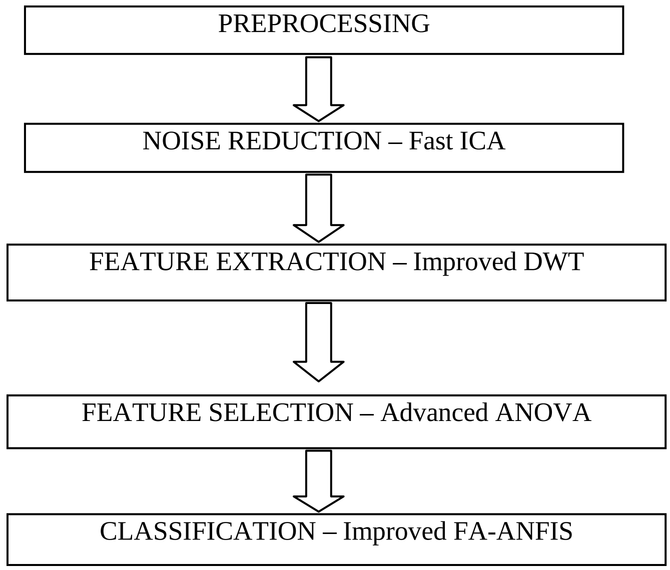

Figure 1: Proposed Methodology Flow Diagram

Pre-Processing

These models of speech signal enhancement indulge various process among which pre-processing is a technique. The pre-processing is performed with a view of eliminating the noises at the background to improve the intelligibility of speech. The following are the methods being proposed for speech signal enhancement.

Framing

In this process, in order to achieve stationary the speech signal is being split into frames as the signals are generally non-stationary. This method of framing greatly helps in quicker processing of speech signals.

Windowing

Each individual frame is brought under the process of windowing as it reduces the discontinuities among the frames at the beginning and at end of frames. At the beginning and end of the frame the signal is narrowed to zero as it is enhanced by the speech signal.

Noise Reduction

The noise reduction is the effective measure to be involve in speech signal enhancement as it improvise the intelligibility of speech thereby eliminating the background noise in the data. The various noise reduction methods exist where Fast ICA is one of the effective means of improvising the quality of speech signal.

Fast ICA

This algorithm is most effective algorithm for independent component analysis. It is employed for examining the hidden factors as it is a computational and statistical technique for speech signal enhancement.

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Feature Extraction

This process in where the acoustic and linguistic features has been extracted based on frequency and time domain as it is an remarkable process in gaining the important features. Improved Discrete Wavelet Transform is employed for extracting feature as it is used for wavelet analysis and also aids in decomposition of signals.

Feature Selection

The feature selection is helpful for selecting the relevant features in ceptral or spectral domain or in temporal domain by eliminating the noise occurring in speech. Various amount of information based on spatio-temporal is available in this feature selection process. Advanced Analysis of Variance (ANNOVA) is utilized for feature selection. This is used for ranking of attributes based on its importance and intend is to make comparison between means of variance. This process aids to provide a robust model for classification purpose.

Classification

ANFIS is the classifier used in this model of speech signal enhancement. The fitness among the feature has been calculated and the ANFIS classifier is used for classifying the features in this process. The Firefly algorithm, an ANFIS classifier is used based on the distant measure features.

3.1. Fast Independent Component Analysis

Independent Component Analysis shortly ICA is a statistical and computational technique for reveal the factors that are hidden namely, measurements, signals or random variables. In this system, the variable relates to data are assumed as linear mixtures for latent variables that are unknown, and mixing system also found to be unknown. These latent variables could be nongaussian or mutually independent, thus they were called as independent components or factors or source which is finding by ICA. Various algorithms are used in order to perform ICA which is more efficient called Fast ICA [11], also called as fixed-point algorithm. Quantitative measures of non-gaussianity have been used for estimation of ICA estimation are kurtosis and negentropy. A Negentropy is one that depends on informative theoretic quantity for entropy. Procedure for implementing Fast ICA with negentropy as follows,

1. Centering of data has been made to mean as zero.

2. Data is allowed to whiten in order to provide z.

3. Choosen of initial vector namely, w in the manner of unit norm.

4. Let $w \leftarrow E\{zg(w^T z)\} - E\{g'(w^T z)\}w$,

g is represented as,

$$g_1(y) = \tanh(y)$$
$$g_2(y) = y\exp(-y^2/2)$$

5. Thus, $w \leftarrow w / \|w\|$

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6. If convergence not occurs then go back to 4th step.

3.2. Improved Discrete Wavelet Transform

The Discrete Wavelet Transform shortly DWT is using in wavelet analysis and it helps to decomposing the signal in basic functions of mutually orthogonal wavelet. It aims to find the difference of sinusoidal basis functions which are spatially localized in manner. In case of Fourier analysis, the Discrete Wavelet Transformation has been invertible. Thus the original signal could be recovered completely from DWT representation. DWT signals are passing in to various filters. Samples are passed in to low pass filters in impulse response which results in to convolution as,

$$y[n] = (x * g)[n] = \sum_{k=-\infty}^{\infty} x[k] g[n-k] \quad 1$$

With high pass filter are used to decompose the signal. Additionally, half of frequencies in signal could be removed; then half of samples were discarded based on Nyquist's rule. The output has been sub sampled by 2 and passed to new filter of high and low pass type which having half frequency of existed one in the manner of cut-off.

$$y_{\text{low}}[n] = \sum_{k=-\infty}^{\infty} x[k] g[2n-k]$$
$$y_{\text{high}}[n] = \sum_{k=-\infty}^{\infty} x[k] h[2n-k] \quad 2$$

Thus decomposition has made to half time resolution and all output has half of frequency band. Finally the frequency resolution gets doubled using sub sampling operator,

$$(y \downarrow k)[n] = y[kn] \quad 3$$

Then it could be write as,

$$y_{\text{low}} = (x * g) \downarrow 2$$
$$y_{\text{high}} = (x * h) \downarrow 2$$

3.3. Advanced Analysis of Variance (AANOVA) Method

Analysis of Variance helps to perform the ranking over important attributes. It is methodology which evaluates the data were the measurement of classification variables has been carried out. This process would take over in various situations. In ANOVA [12] process, the variation in response has been separated as variation attribute in order to differentiate the classification variables and the variation attributable as random error. The importance of classification effects have been identified using test conduct by this approach. Overall, the intent of this AANOVA approach is to make comparison between means of variance. This model would act as linear model and since it is a two way analysis, it is extended to one way analysis. The term factor represents the two variables that are independent in two way analysis which would find out how the response affects these factors. AANOVA is an

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approach that involves advance concepts of analysis of variance which offers various advantages in sense of statistical analysis.

3.4. Firefly Algorithm (FA)

Firefly algorithm has been designed for performing optimization and it helps to evaluate the positive, accuracy, specificity, sensitivity and predictive value. The optimization towards problem could be in discrete or in continuous manner. Thus for this above said purpose, firefly algorithm [13] has been used. It aims to find out specifically global solution in or0der to deal and solve the NP hard problems. It also applicable for solving issues related to optimization in case of search space. It will interpret the characteristics among light intensity too. Each and every firefly is unisex and may be seems to an attractive and which is proportional to the intensity. One which possesses low intensity will move towards other fireflies. One with high intensity would moves over search space in the sense if not present. Issues are listed here,

Attraction function: The structure of attractive function is β

$$\beta = \beta_0 \times e^{-\gamma r_{ij}^2} \quad 4$$

Encoding method: In length m, possible solution has been defined.

Where, m=|C|

1 = denotes selection over corresponding attribute.

0 = it represents the attribute which are not selected.

Fitness function: By reduction of attribute, best solution is identified using fitness function which is defined as follows,

$$Fitness(X) = \frac{m - |X|}{m} + \frac{n|R|\gamma_X(D)}{m\Gamma} \quad 5$$

Where,

$\gamma_X(D)$ = represents classification over quality

R = denotes the redact of the condition attribute namely C

3.5. Improved Adaptive Neuro- Fuzzy Inference System

Adaptive Neuro-Fuzzy Inference System [14] is an algorithm applicable for performing classification on character or signals which is machine oriented. This methodology could be trained with the help of the back propagation gradient descent and the least squares based methods. The result occurred during training and by means of accuracy over classification, performance has been evaluated. ANFIS which stands for Adaptive Neuro- Fuzzy Inference system would purposefully focus on to deal the problems that are related to parameter identification. ANFIS has forward and backward pass. One having forward pass, signals of network would propagates forwardly and backward pass makes the signals to move backward by means of error. The parameters are identified and made as fixed. The result occurred would be a linear combination among parameters which is denoted by means of f.

Thus output f can be,

$$f = (\bar{w}_1 x)c_{11} + (\bar{w}_1 y)c_{12} + \bar{w}_1 c_{10} + (\bar{w}_2 x)c_{21} + (\bar{w}_2 y)c_{22} + \bar{w}_2 c_{20} \quad 6$$

Where,

$$c_{ij}\ (i = 1,2,\ j = 0,1,2).$$

Parameters which are consequent is represented as c_{ij} in forward pass

The premises parameters are represented as $\{a_i,\ b_i,\ c_i\}$ in backward pass.

IV. EXPERIMENT RESULTS

This model has been developed for enhancing the speech signal as it is helpful for hearing aid application. This involves various steps like pre-processing of the data, feature extraction, classification and so on. Thus this model has been developed for enhancing the signal with great accuracy and efficiency. The proposed algorithm increases the efficiency.

Speech Signal Enhancement Quality of the recommended method is evaluated by using different performance measures. This metrics of the proposed techniques are compared with existing algorithms like Spectral subtraction, wiener filter, genetic SVD, Hybrid PCA-ANFIS and Modified Bayesian-ANFIS.

SNR

Signal-to-Noise Ratio (SNR), computes the average of the SNR values.

$$SNR_{Seg} = \frac{10}{M} \sum_{m=0}^{M-1} \log_{10} \sum_{i=Nm}^{Nm+N-1} \left(\frac{\sum_{1=1}^{N} x^2(i)}{\sum_{1=1}^{N} \left(x(i) - y(i)\right)^2} \right) \quad 7$$

Where N and M are the segment length and the number of segments respectively.

PESQ

Perceptual Evaluation of Speech Quality measure is the computation used for speech quality assessment. The final PESQ score is obtained by a linear combination of the average disturbance value D and the average asymmetrical disturbance values as follows A:

$$PESQ = a_0 - a_1 .\, D - a_2 .\, A \quad 8$$

Where, $a_0 = 0.1$, $a_1 = 0.1$, and $a_2 = 0.0309$

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Figures 2, 3, 4, 5 demonstrates the results of various speech processing algorithms under three background noises at SNR and PESQ levels. The result indicates that proposed method outperforms existing methods.

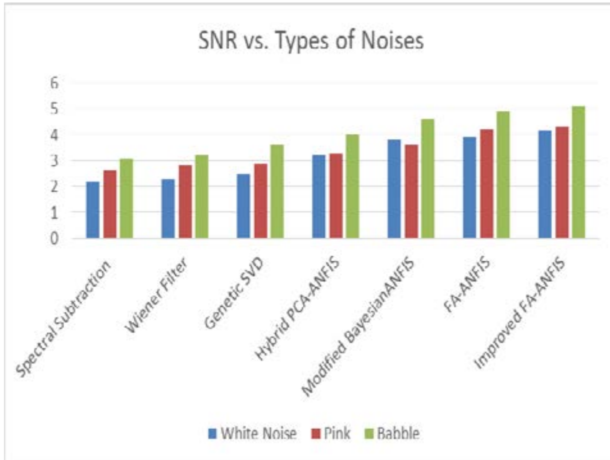

Figure 2: SNR vs Types of Noises (X-Axis: Methods, Y-Axis: SNR)

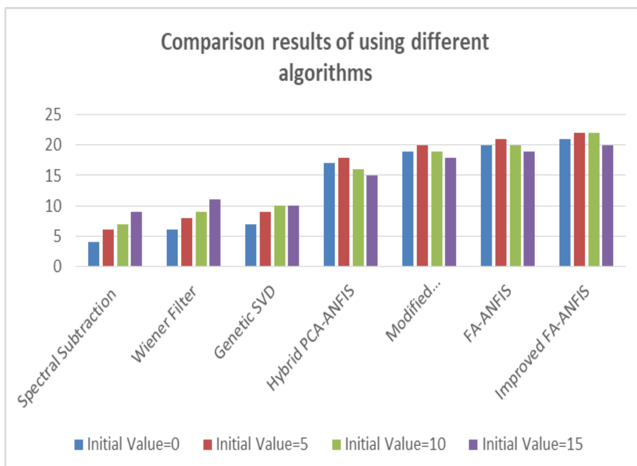

Figure 3: SNR - The Comparison results of using different algorithms (X-Axis: Methods, Y-Axis: Final Value SNR)

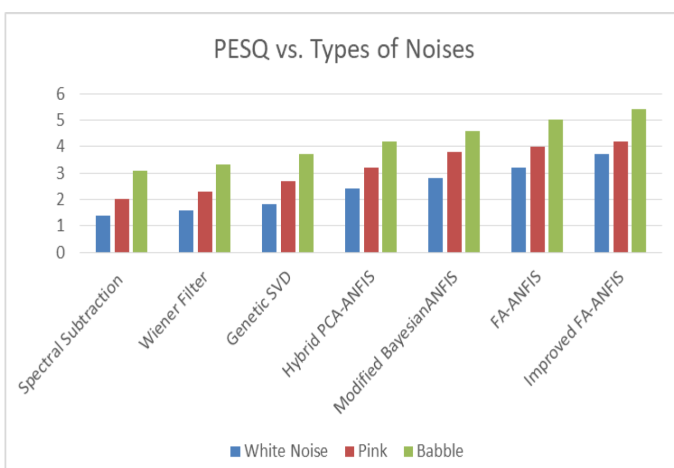

Figure 4: PESQ vs Types of Noises (X-Axis: Methods, Y-Axis: PESQ)

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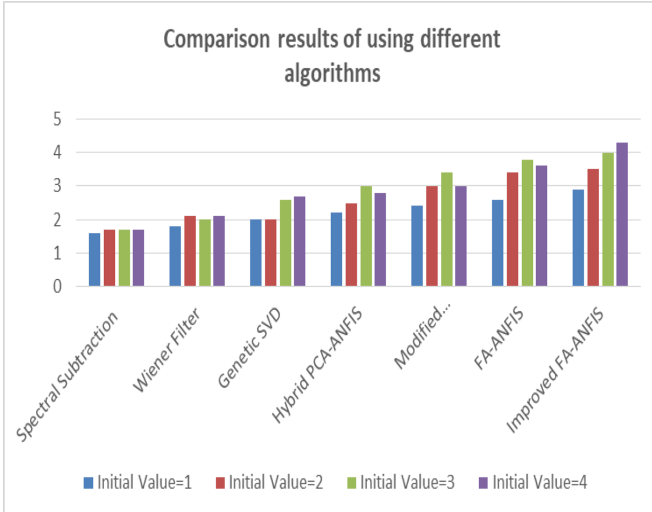

Figure 5: PESQ - The Comparison results of using different algorithms (X-Axis: Methods, Y-Axis: Final Value PESQ)

V. Conclusion and Feature Work

The signal generally contains noises which are eliminated by means of this model. The pre-processing eliminates the background noise thereby improving the speech signal. Windowing and framing are the pre-processing methods involved in this model. Then the noise reduction by means of fast ICA is performed and then the feature extraction and feature selection is being performed on the selection for extracting the important features and providing ranks. Finally an ANFIS classifier using Firefly algorithm has been used for efficient classification. The features are used for the classification which are inputted from the feature selection as it is arranged based on ranking. Experiment results shows that the new proposed method performs well compared to existing methods like Spectral subtraction, wiener filter, genetic SVD, Hybrid PCA-ANFIS and Modified Bayesian-ANFIS. The future work with the improvised models will be performed in order for increasing the performance of the signal and to ensure the absence of background noise.

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