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Speech Enhancement through Implementation of Adaptive Noise Canceller Using FHEDS Adaptive Algorithm

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Abstract: Speech analysis is the modelling and estimating of the different speech characteristics that would provide the importance on each set of criteria established on the real time applications. One such analytic section in enhancement process on speeches would improve the need of speech enhancement. This paper compares the performance analysis of our proposed Fast Hybrid Euclidean Direction Search (FHEDS) algorithm with other adaptive algorithms such as NHP and FEDS algorithm. These algorithms have been tested for their adaptive noise cancellation of speech signal corrupted by different noises such as Babble, Factory, Destroy Engine, Car, Fire Engine and Train Noises. Ensuring the design criteria with current design limits of the database and its analysis have been encapsulated with each phase of design with Noise model, improving the better performance aspects. The relative factors for comparisons have been tabulated with each set of the noise and clear speech data with proposed filter operation. The proposed model effectively reduces the noise for achieving better speech enhancement. The proposed model achieves high Signal-to-Noise Ratio (SNR) when compared to traditional models.

Index Terms: Speech Enhancement, Normalized Hybrid Projection (NHP), Fast Euclidean Direction Search (FEDS)**,** Fast Hybrid Euclidean Direction Search Algorithm (FHEDS), Signal to Noise Ratio (SNR).

1. Introduction

In the current real time environments adaptive filtering would improve the novel process for implementing the speech enhancement which is corrupted with different random noises such as machines, other speakers, animal voices etc. The distortions observed with SNR criteria would reduce the signal capabilities. Depending upon the listener's types NH and HI would impart challenging scenario's to improve the quality and intelligibility of speeches. The performance characteristics of speech enhancement would improve the different situations to create the different optimization models and its usage application [1]. Learning scenarios approach would suggest the importance on how the data would classify the noise and original speech and its performance characteristics [2].

Many algorithms for adaptive filters are popularly used for acoustic signal, acoustic noise cancelation radar, sonar and biomedical analysis, connectivity, yet they either have a high electrical and control network [3]. In the adaptive algorithm collection considered to solve a variety of adaptive filtering problems related to structures based on subfine projections [4] of normal version of the adaptive algorithm. In all algorithms, convergence can change and the selected fixed step size is in stable state means square defect is observed at a stable state [5].

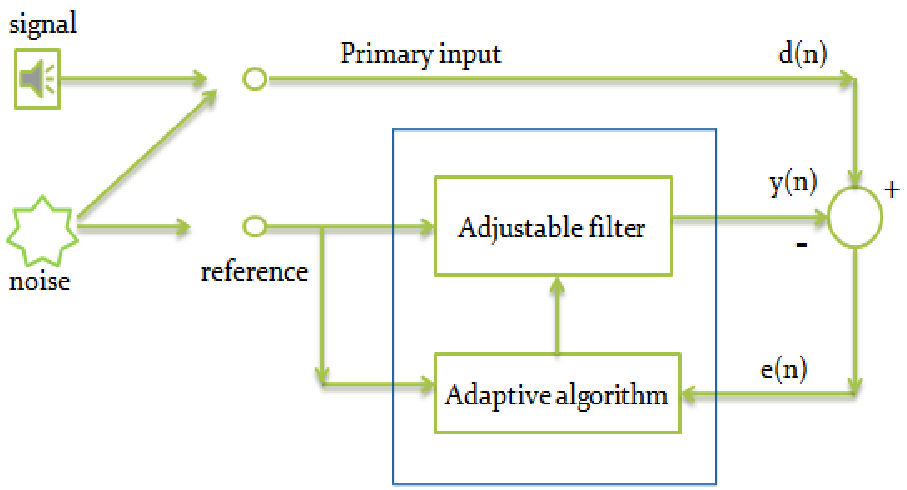


Fig.1. Adaptive Noise Canceller Block Diagram

Considering the design block diagram where the initial data i.e. primary one is $d(n)$ and the noise signal is $n(n)$. Assuming that noise is an uncorrelated values observed from the desired signal [6]. The reference input $x(n)$ is mean to consider a measuring for a distorted signal which have to correlated with the noise input $n(n)$. Now, the current reference input $x(n)$ for the above model is operated and processed by the adaptive filter designed to obtain the estimated output $y(n)$. Hence to analyze specific errors occurred and observed while filtering the error value from the filtered output, the output equations is given by:

$$y(n) = w^T(n) * x(n) \tag{1}$$

$$e(n) = d(n) - y(n) \tag{2}$$

Adaptive filter is a digital filter that adjusts itself and automatically adjusts if the input is not stable, adjust the input signals. Adaptive noise filter canceller contains two different elements, one of which is an adjustable filter and another one is adaptive algorithm [7]. The desired signal (d) is considered to be the unwanted portion that is used as adaptive filter that is corrupted and our goal is to delete it.

Our design aims to improve the de noising scenario on each set of data observed after the adaptive filtering process [8]. The parametric considerations would emphasize on SNR, MSE, RMSE and Distortion. [9].

1.1 Motivation

Speech Signals are Corrupted in real time environment by several forms of noise such as Competing Speakers, Background noise, Room reverberation, Low quality microphones, Babble noise etc [10]-[14]. The Clean Speech Signal is necessary for applications such as Speech or Speaker recognition, Hearing aids, Mobile communication etc. The objective of Speech Enhancement is to improve Speech quality from degraded Speech signal. The performance of these Adaptive Filtering algorithms are dependent on their filter length (N) and the convergence parameter (μ) commonly known as step size.

1.2 Contributions of this paper

The key contributions of the proposed system are as follows:

- To design proposed FHEDA adaptive algorithm to overcome the problems of existing adaptive algorithms, this can improve the performance of the conventional adaptive noise cancellation scheme.
- To select the proper filter for implementation of the ANC system.
- To collect Speech signal as they apply to ANC from the NOIZEUS AND TIMIT data bases.
- To examine the adaptive filtering techniques as they apply to Speech Enhancement.
- To analyze filtering operation of noise corrupted Speech signal using proposed FHEDA adaptive algorithm.
- To compare the simulated results with existing techniques.

2. Description of Adaptive Filtering Algorithms

The primary signal is the sum of speech signal s(n) and noise signal n(n)

$$d(n)= s(n)+n(n) \tag{3}$$

The referred input x(n) are mean to consider a measuring for a distorted signal which have to match with the noise input n(n).

The filter output, y(n) is given by equation (1).

The output of adaptive filter subtracts from primary input to generate the adaptive filter error which is presented in equation (2)

Where x(n) is the time delayed input vector $x(n) = [x(n), x(n-1), x(n-2), \ldots\ldots x(n-M+1)]^T$, the vector $w(n) = [w_0(n), w_1(n), w_2(n), \ldots w_{M-1}(n)]^T$ represents the coefficients of the filter at time index n [15]-[17].

The variable filter has a Finite Impulse Response (FIR) structure. For such structures the impulse response is equal to the filter coefficients.

2.1 Normalized Hybrid Project Algorithm (NHP)

The weight update calculation for the NHP design is as follow

$$w_e = \frac{\epsilon.*mean(b).*d(n)}{sqrt(d(n)).*d^T(n)*\varepsilon} \tag{4}$$

Overall update equation for NHP Algorithm is

$$w(n+1) = w(n) + \mu * x(n) * e(n) + w_e \tag{5}$$

Here ***d*** is desired signal, ***b*** is FIR filter output for input response, x***(n)*** is input signal, μ is step size and ***e***(n) is error signal [18].

2.2 Fast Euclidian Direction Search Algorithm (FEDS)

The overall update equation for FEDS Algorithm is

$$w(n) = w(n-1) + \mu * x(n) * C^{-1}(n) * e(n) \tag{6}$$

Where

$$C^{-1}(n) = \frac{1}{\left\|x_j(n)\right\|^2}\ i_j(n)$$

3. Proposed Fast Hybrid Euclidian Direction Search Algorithm(FHEDS)

The proposed optimization technique with the FIR filter and its structure have been utilized with order and depth of the filter. The current proposed design approach where the weight of the design is combined with NHP and Euclidian Direction Search. The predictive analysis of the current noise model is characterized with Bayes method where the Euclidian Direction of each signal and noise are estimated. EDS is a modern class of algorithms focused on least squares. Originally, this class of algorithms was developed to combine the advantages of fast RLS convergence with the low computational complexity of LMS. The EDS and RLS algorithms have O(N2) completed and a similar rate of convergence to overcome the computation complexity problem. The O(N) complexity of FEDS is lower than the EDS but much higher than LMS, and a convergence rate is much greater. In this text, we create two new algorithms, based on the intrinsic characteristics of EDS and FEDS algorithms, with faster rates for converging than the original algorithms.

3.1 Design Procedure:

With current design optimization algorithms and its analysis would estimate the importance of the machine learning approaches in our current design model [18]. The model is provisioned with Bayes approach for the noise estimation on the current optimization technique to improve the design parametric criteria as per the requirements. Our model with FIR filter approach with Hamming window, initiates the design features of improvements such as SNR, MSE, RMSE, Distortion for each set of practical and database noises are considered. The weight's characterized with each adaptive filtering on the optimization algorithms have to be featured with Euclidian Direction and Bayes noise model. In accordance with design model and its criteria we have initiated a procedure to ensure the design for the current FIR filter with adaptive filtering as follows:

3.2 Proposed Design Framework:

i. Initiate the model with the input speech with acoustic model from database.
ii. The Noise characteristic with Bayes model and with Euclidian approach for Noise reducing after the desired signal obtained.
iii. The obtained signal (d) is applied to the respective optimization algorithms to ensure the correct output obtained as per the input speech signal.
iv. Calculation of each model with respective to the parametric criteria (SNR, MSE, RMSE and Distortion) values are depicted in the results and discussions table.

3.3 FHEDS Modeling Using Bayesian Noise Reduction:

The NHP with Euclidian Direction Search model and its algorithm have utilized the proposed approach of noise model. We initiate the filter with required number of coefficients and create a noise related to input as audio signal. The weight update calculation for the proposed design is as follow:

Overall update equation for FHEDS Algorithm is

$$w(n+1) = w(n) + \mu * x(n) * e(n) * p_e \tag{7}$$

Where

$$p_e = c_1 exp\left[-\frac{\|D-S\|_F^2}{2{\sigma_n}^2}\right] + \frac{\epsilon .*mean(b).*d(n)}{sqrt\left(d(n)\right).*d^T(n)*\varepsilon} \tag{8}$$

Here ***d*** is desired signal, ***b*** is FIR filter output for input response, x***(n)*** is input signal and ***e***(n) is error signal, ***s*** is speech signal, α is step size, the parameter ϵ is small positive value. Considering the Noise reduction design model using Bayes method utilizing the probability scenario.

The probability of signal strength is calculated as

$$p(D|S) = C_1 exp\left[-\frac{||D-S||_F^2}{2\sigma_n^2}\right] \tag{9}$$

The magnitude of the $p(D|S)$would represent the weights associated with each noise and signal utilized for each algorithm.

4. Results and Discussion

In this section, we apply FEDS, NHP and proposed FHEDS adaptive algorithms to noise reduction in speech applications, and examine their performance. The speech data which sampled at 8 kHz was used to evaluate the proposed method and recorded time is 2 sec. The performances of these algorithms are investigated for speech enhancement in different noise samples. Six types of noises were implemented namely Babble, Factory, Destroy Engine, Car, Fire Engine and Train Noise at 0, 5, 10, and 15dB SNR. These noises are taken from the NOIZEUS and TIMIT databases. The filter order was set to M=10. The objective measures that are considered for the evaluation of the proposed Methods are SNR, MSE, and Root MSE (RMSE) and Distortion values for the given samples.

Table 1. Quality Assessment Parameters

PARAMETERS	FORMULA
Mean Square Error	$\frac{1}{N}\sum_{n=0}^{N-1}[s(n)-e(n)]^2$
Root MSE	$\sqrt{\frac{1}{N}\sum_{n=0}^{N-1}[s(n)-e(n)]^2}$
SNR	$10\log_{10}\left(\frac{\sum_{n=0}^{N-1}[s(n)]^2}{\sum_{n=0}^{N-1}[s(n)-e(n)]^2}\right)$
Distortion	$10log_{10}\sum_{n=0}^{N-1}[s(n)-e(n)]^2$

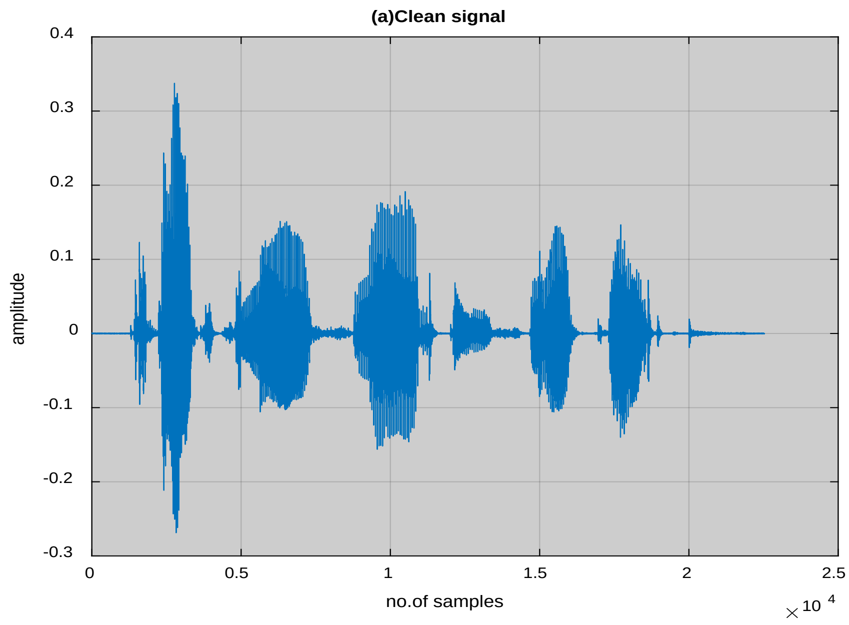
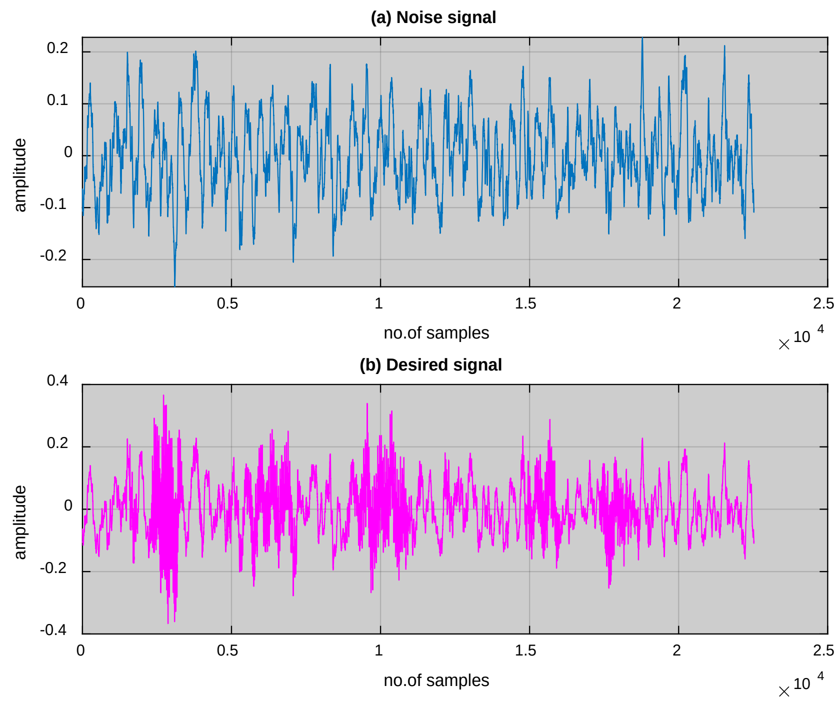


Fig.2. Clean Speech Signal

In the waveform of a clean speech signal ‘sp01’ is taken from NOIZEUS database. The original signal has 22529 samples. In the waveform of a clean speech signal the X-axis is calibrated to number of samples and Y-axis is calibrated to amplitude measured in decibels (dB). The amplitude of a speech signal is measured on a decibel scale as it is best correlated with perceived sound loudness. The Clean speech signal commonly used for proposed adaptive algorithms.

4.1. Speech Enhancement using FEDS algorithm



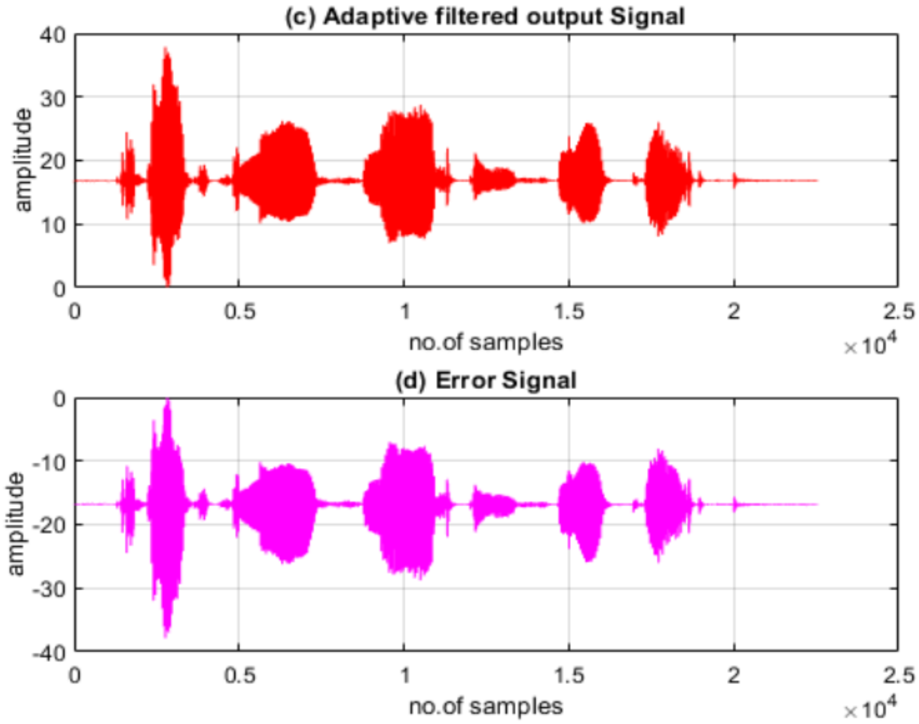
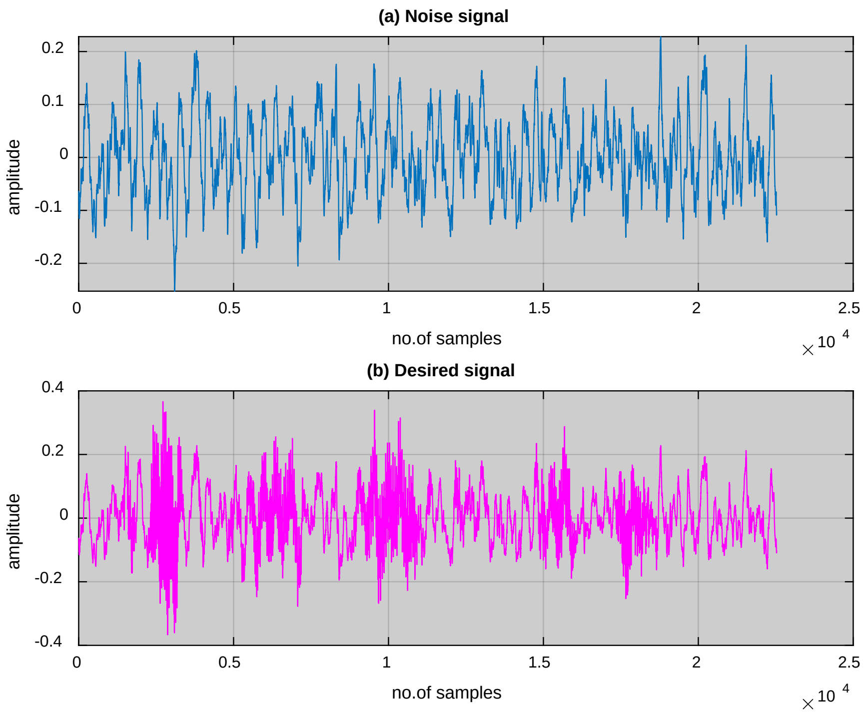


Fig. 3. Speech Enhancement Waveform representations using FEDS algorithm

4.2 Speech Enhancement using NHP algorithm



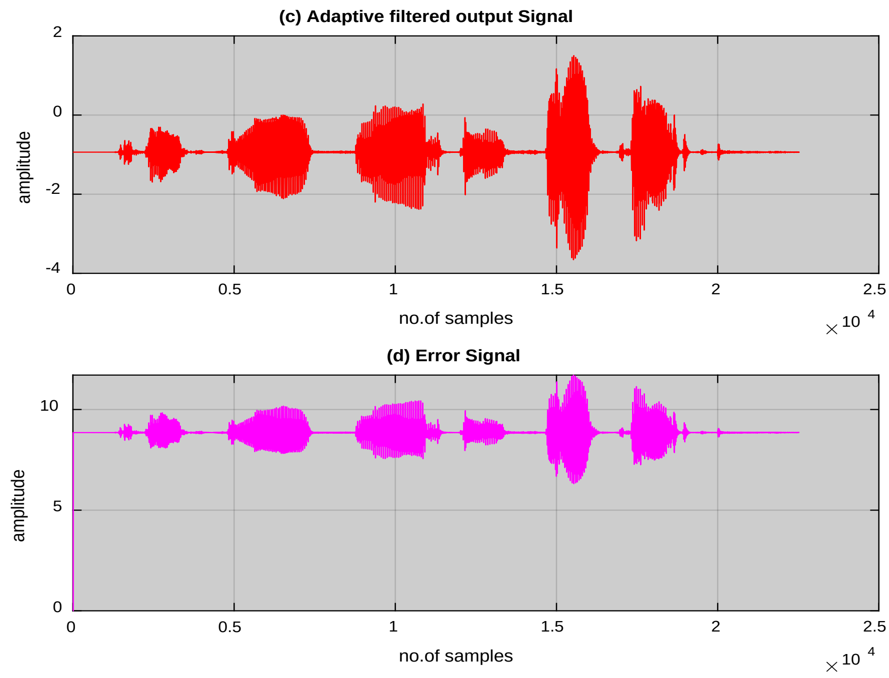
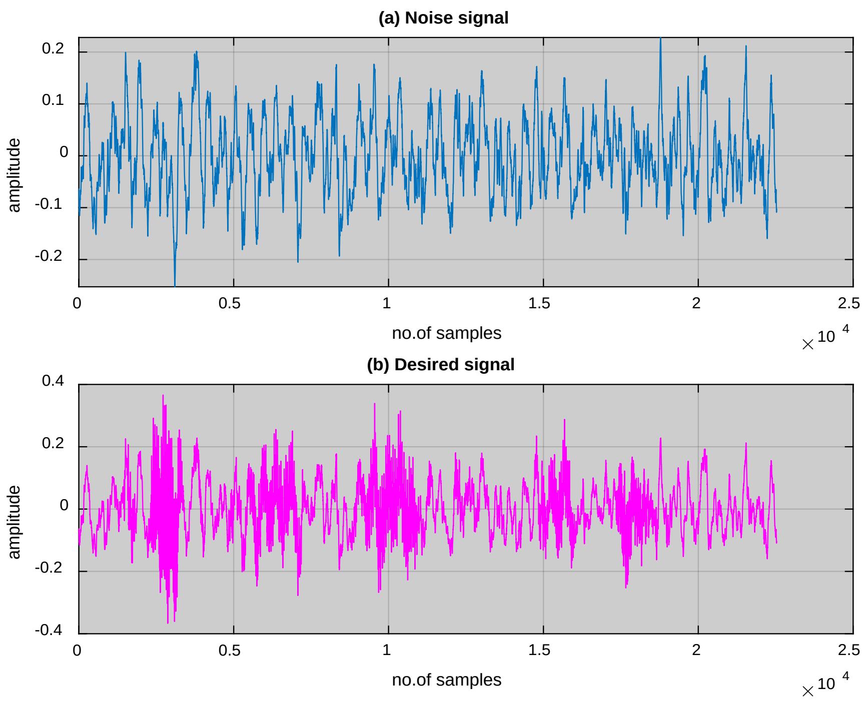


Fig. 4. Speech Enhancement Waveform representations using FEDS algorithm

4.3 Speech Enhancement using FHEDS algorithm



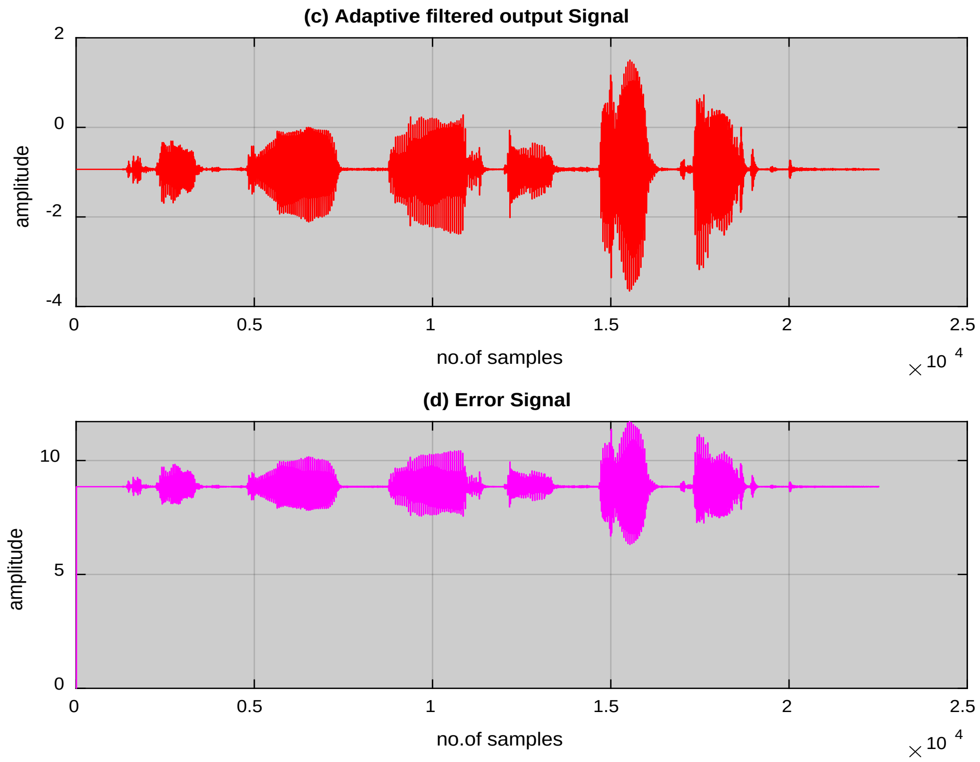


Fig. 5. Speech Enhancement Waveform representations using FHEDS algorithm

The Figs. 3, 4, 5 depict the plots of each stage of the input with respect to the adaptive filtering according to the test data sets from database. Plots for Clean Signal, Noise Signal, Desired Signal, Adaptive Filtered Output Signal and Error Output Signals using Car 10dB Noise have been plotted with respect to the total length of the input for NHP, FEDS, and FHEDS algorithms.

Table 2. Representing the SNR, MSE, RMSE and Distortion values for each noise with respect to the input considered for FEDS algorithm.

FEDS						
S.No	NOISE	Noise in dB	SNR	MSE	RMSE	DISTORTION
1	BABBLE	0	6.3809	9.86E-09	9.93E-05	-4.4083
		5	14.1198	9.46E-09	9.73E-05	-3.8698
		10	19.9548	8.82E-09	9.39E-05	-3.5366
		15	34.1885	4.32E-09	6.57E-05	-2.7928
2	FACTORY	0	6.7783	9.83E-09	9.92E-05	-4.3749
		5	13.2243	9.51E-09	9.75E-05	-3.9232
		10	20.9103	8.66E-09	9.31E-05	-3.4845
		15	32.3066	5.25E-09	7.25E-05	-2.8887
3	DESTROY	0	6.0007	1.02E-08	1.01E-04	-4.4413
		5	14.8511	9.69E-09	9.85E-05	-3.8261
		10	19.9532	9.11E-09	9.54E-05	-3.5367
		15	35.6115	3.72E-09	6.10E-05	-2.7203
4	CAR	0	5.8684	1.07E-08	1.04E-04	-4.4513
		5	15.0849	1.02E-08	1.01E-04	-3.8123
		10	19.9527	9.63E-09	9.81E-05	-3.5367
		15	36.0488	3.79E-09	6.16E-05	-2.6983
5	FIRE ENGINE	0	7.0259	1.24E-08	1.12E-04	-4.3546
		5	12.6248	1.21E-08	1.10E-04	-3.9616
		10	19.9587	1.13E-08	1.06E-04	-3.5363
		15	30.8701	7.95E-09	8.91E-05	-2.9622
6	TRAIN	0	7.1175	1.23E-08	1.11E-04	-4.3472
		5	12.3797	1.20E-08	1.10E-04	-3.9717
		10	20.9103	1.10E-08	1.05E-04	-3.4845
		15	30.2699	8.12E-09	9.01E-05	-2.9923

Performance of FEDS algorithm in terms of SNR, MSE, RMSE and Distortion values are shown in Table 2. The maximum SNR value in dB obtained is 36.0488 at Car noise. The minimum MSE value obtained is 3.72e-09 at Destroy noise. The minimum RMSE value obtained is 6.10e-05 at Destroy noise. The minimum Distortion value in dB obtained is -2.6983 at Car noise.

Table 3. Representing the SNR, MSE, RMSE and Distortion values for each noise with respect to the input considered for NHP algorithm.

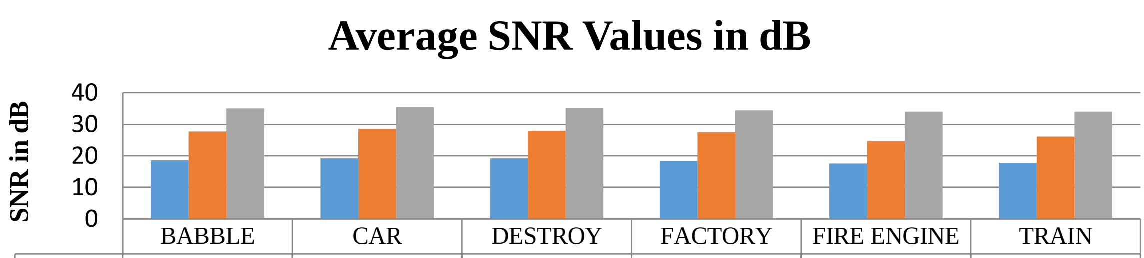
NHP						
S.No	**NOISE**	**Noise in dB**	**SNR**	**MSE**	**RMSE**	**DISTORTION**
1	BABBLE	0	11.9056	1.59E-07	3.99E-04	-3.3855
		5	22.8669	1.57E-07	3.96E-04	-2.8133
		10	30.1951	1.54E-07	3.92E-04	-2.4730
		15	46.0357	1.28E-07	3.57E-04	-1.7284
2	FACTORY	0	12.8555	1.59E-07	3.99E-04	-3.3484
		5	21.6137	1.57E-07	3.97E-04	-2.8684
		10	31.3039	1.53E-07	3.91E-04	-2.4203
		15	43.9468	1.34E-07	3.66E-04	-1.8205
3	DESTROY	0	15.6633	1.59E-07	3.99E-04	-3.4223
		5	23.8665	1.57E-07	3.96E-04	-2.7683
		10	30.1822	1.54E-07	3.92E-04	-2.4731
		15	47.6387	1.22E-07	3.49E-04	-1.6539
4	CAR	0	11.3717	1.59E-07	3.99E-04	-3.4356
		5	24.1784	1.57E-07	3.96E-04	-2.7581
		10	30.1929	1.54E-07	3.92E-04	-2.4752
		15	48.1301	1.20E-07	3.47E-04	-1.6297
5	FIRE ENGINE	0	15.2907	1.59E-07	3.99E-04	-3.3266
		5	20.6722	1.58E-07	3.97E-04	-2.9082
		10	30.1997	1.54E-07	3.92E-04	-2.4728
		15	42.3641	1.38E-07	3.71E-04	-1.8963
6	TRAIN	0	16.4941	1.59E-07	3.99E-04	-3.3179
		5	21.1718	1.58E-07	3.97E-04	-2.9242
		10	31.3039	1.53E-07	3.91E-04	-2.4203
		15	43.2095	1.39E-07	3.73E-04	-1.9285

Performance of NHP algorithm in terms of SNR, MSE, RMSE and Distortion values are shown in Table 3. The maximum SNR value in dB obtained is 48.1301 using at Car noise. The minimum MSE value obtained is 1.20e-07 at Car noise. The minimum RMSE value obtained is 3.47e-04 at Car noise. The minimum Distortion value in dB obtained is -1.6297 at Car noise.

Table 4. Representing the SNR, MSE, RMSE and Distortion values for each noise with respect to the input considered for FHEDS algorithm.

FHEDS						
S.No	**NOISE**	**Noise in dB**	**SNR**	**MSE**	**RMSE**	**DISTORTION**
1	BABBLE	0	28.008	6.77E-06	0.0026	-3.3822
		5	30.8605	6.56E-06	0.0026	-2.8149
		10	34.0225	6.50E-06	0.0025	-2.4740
		15	47.1015	6.28E-06	0.0025	-1.7246
2	FACTORY	0	28.0176	6.75E-06	0.0026	-3.3464
		5	30.4495	6.57E-06	0.0026	-2.8701
		10	34.6028	6.49E-06	0.0025	-2.4212
		15	44.4808	6.33E-06	0.0025	-1.8208
3	DESTROY	0	27.8088	6.80E-06	0.0026	-3.4176
		5	31.217	6.55E-06	0.0026	-2.7699
		10	34.0215	6.50E-06	0.0025	-2.4741
		15	47.9918	6.23E-06	0.0025	-1.6519
4	CAR	0	27.9618	6.81E-06	0.0026	-3.4302
		5	31.3336	6.55E-06	0.0026	-2.7557
		10	34.0212	6.50E-06	0.0025	-2.4741
		15	48.3253	6.39E-06	0.0025	-1.6225
5	FIRE ENGINE	0	28.03	6.74E-06	0.0026	-3.3246
		5	30.1709	6.58E-06	0.0026	-2.9100
		10	34.0249	6.50E-06	0.0025	-2.4738
		15	43.4129	6.36E-06	0.0025	-1.8946
6	TRAIN	0	28.0978	6.73E-06	0.0026	-3.3167
		5	30.063	6.59E-06	0.0026	-2.9260
		10	34.6028	6.49E-06	0.0025	-2.4214
		15	42.9715	6.37E-06	0.0025	-1.9255

Performance of FHEDS algorithm in terms of SNR, MSE, RMSE and Distortion values are shown in Table 4. The maximum SNR value in dB obtained is 48.3253 using at Car noise. The minimum MSE value obtained is 6.22e-06 using at Car noise. The minimum RMSE value obtained is 0.0025 using at Car noise. The minimum Distortion value in dB obtained is -1.6225 using FHEDS algorithm at Car noise.



	BABBLE	CAR	DESTROY	FACTORY	FIRE ENGINE	TRAIN
FEDS	18.661	19.2387	19.103875	18.308875	17.619875	17.66935
INHP	27.750825	28.4507	27.838125	27.429975	24.632925	26.089575
FHEDS	34.996575	35.410475	35.25595	34.387675	33.95225	33.93335

Fig. 6. Performance comparison of adaptive filtering average SNR values in dB

The performance comparisons of the average SNR value bar graphs for adaptive filtering are shown in Fig. 6. As can be seen from the Fig. 6 the maximum average SNR value in dB obtained is 35.4104 for FHEDS algorithm at car noise.

The signal to noise ratio of the proposed model is high that indicates the strong signal strength. The signals to noise ratio levels are depicted in the Fig. 7.

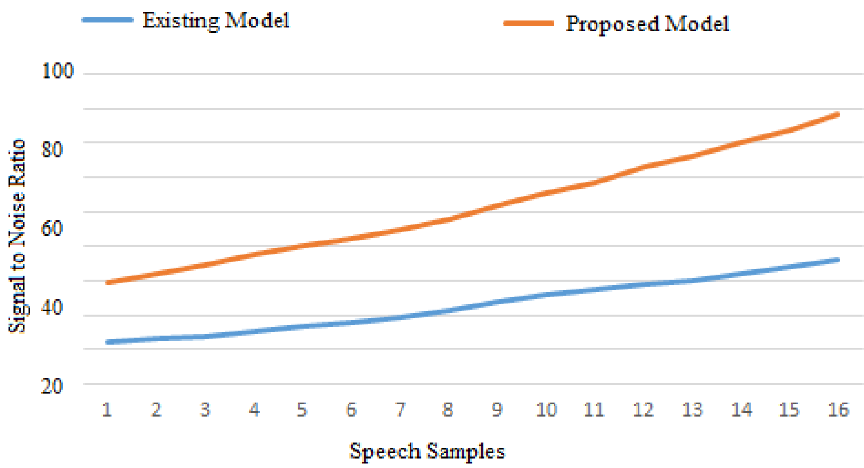


Fig. 7. Signal to Noise Ratio

5. Conclusion

In this paper, we have introduced a noise reduced approach using to process speech input signal by applying novel filtering method known as Fast Hybrid Euclidian Direction Search Algorithm. Under the testing approach we have introduced an optimization algorithm for each set of scenarios to ensure the filtering technique with FEDS and NHP technique to impart a novel model. Now these values would determine the stability of the design and its performance characteristics. The simulation results show that the SNR rate of these algorithms is comparable with the proposed FHEDS design technique. The performance of proposed algorithm is compared quantitatively by parameters SNR, MSE, RMSE and Distortion. Better filtering performance results are obtained by FHEDS adaptive algorithm and also two algorithms guarantee the better estimation of noise. Computer simulation demonstrates that the proposed algorithm gives improved performance and achieves good adaptation. The proposed FHEDS algorithm performs better SNR. The future work is as follows:

- To implement sub band adaptive filtering for speech enhancement,
- To overcome high computational complexity and low convergence rate.

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