IJCSI International Journal of Computer Science Issues, Vol. 9, Issue 2, No 2, March 2012
ISSN (Online): 1694-0814
www.IJCSI.org

Environmental Noise Classification and Cancellation using Fuzzy Classifier and Fuzzy Adaptive Filters

T.Meera Devi [1], Dr.N.Kasthuri [2] and Dr.A.M.Natarajan [3]

[1] Department of ECE, Kongu Engineering College
Perundurai, Erode -638 052, TamilNadu, India

[2] Department of ECE, Kongu Engineering College
Perundurai, Erode -638 052, TamilNadu, India

[3] Department of CSE, Bannari Amman Institute of Technology
Sathyamangalam, TamilNadu, India

Abstract

The background noise is one of the major factors, which adversely affects the perceived grade of service in audio communication systems. The main problem in most of the environmental noise reduction system is source of noise signal which is to be used as a reference signal. Once the noise source is known then the noise elimination process will become easier. In any environmental conditions, predominant noise signal which is corrupting the original speech signal can be identified using neural network classification. In conventional gradient-based learning algorithms, tuning methods need to be differentiable and it leads to slow in convergence and if the noise is nonlinear, it will not provide a good generalization performance. To overcome this, an automatic noise reduction system is proposed which is an integration of fuzzy classifier and fuzzy adaptive filter.

Keywords: *Fuzzy Inference System, Adaptive Filter, Fuzzy Adaptive Wiener filter, MELfrequency coefficients, Independent Component Analysis*

1. Introduction

The quality and intelligibility of the speech in the presence of background noise is especially an important issue for devices used in mobile environments, such as mobile phones and hands-free telephones in cars and also in hearing aids. Speech Intelligibility in high noisy environments makes speech communication difficult.

The most common problem in speech processing is the effect of interference noise in speech signals. Interference noise masks the speech signal and reduces its intelligibility. Interference noise comes from acoustical sources such as ventilation equipment, traffic, crowds and commonly, reverberation and echoes. It also arises electronically from thermal noise, tape hiss or distortion products. Several speech enhancement algorithms have been proposed over the years.

The background noise is one of the major factors, which adversely affects the perceived grade of service in audio communication systems. When the background noise level is high, the person on the receiving end is asked to speak louder or the caller may increase his volume, while having a conversation on mobile phones. Further when the background noise levels are high, users tend to bring their mobiles very close to their ears. Several techniques have been proposed to minimize the background noise in mobile environment.

2. Existing Work

Microphone based spectral subtraction has been proposed by Zhipeng Zhang, Minoru Etoli for suppressing background noise. The technique differentiates between intervals with speech and intervals without speech. The input signal for the interval without speech is recognized as background noise and uses it to create a noise spectrum. The noise spectrum is then subtracted from the input signal for the intervals with speech and noises are mixed. Accordingly, good generalization performance is achieved in the noise cancellation when the noise signal is stationary and the technique is easily implemented. When the background noise is a non-stationary signal, however such as in a noisy environment or in a busy traffic good noise cancellation performance is not achieved.

Philipos C. Loizou,a_ Arthur Lobo, and Yi Hu (2005) proposed an algorithm based on subspace principles and projects the noisy speech vector onto “signal” and “noise” subspaces. An estimate of the clean signal is made by retaining only the components in the signal subspace. But if the content of noise is increased, it is very difficult to understand the clear signal, thus noise in speech recognition remains a paramount problem.

Zhang Yan Zhao Ji-yin Ma Yu-kuan Sun Ling-ming (2004) preferred the adaptive noise cancellation based on back propagation algorithm and genetic algorithm. The back propagation network adopts gradient searching. But the convergence rate was slow and the genetic algorithm used only optimized initial weight value in the network.

Juarez-Hernandez M.A.Sanchez Garcia J.C (2005) discussed the concept of converting linear adaptive noise cancellation into a nonlinear technique using ANFIS. ANFIS minimize the error component attributable to desired component. The estimation error is more pronounced at certain values. Hence improvement is needed for training data from a longer interval.

Ying He, Hong He, Li Li, Yi Wu, Hongyan Pan, (2008) proposed the Applications and Simulation of Adaptive Filter in Noise Cancelling. This paper describes the theory of the adaptive filter and adaptive noise cancellation. According to the RLS algorithms realize the design and simulation of adaptive algorithms in noise canceling, and compare and analyze the result then prove the advantage and disadvantage of algorithms.

M. A. Abd El-Fattah, M. I. Dessouky , S. M. Diab and F. E. Abd El-samie (2006) proposed the application of the Wwener filter in an adaptive manner in speech enhancement. The proposed adaptive Wiener filter depends on the adaptation of the filter transfer function from sample to sample based on the speech signal statistics

Lilatul Ferdouse, Nasrin Akhter, Tamanna Haque Nipa and Fariha Tasmin Jaigirdar (2011) simulated different adaptive algorithms for noise cancellation.

3.Proposed Work

The main problem in most of the environmental noise reduction system is source of noise signal which is to be used as a reference signal. Once the noise source is known then the noise elimination process will become easier. In any environmental conditions, predominant noise signal which is corrupting the original speech signal can be identified using neural network classification. Conventional gradient-based learning algorithms, such as back-propagation (BP), Bayes classifier and Levenberg–Marquardt (LM) method have been extensively used in the training of multilayer feed-forward neural networks.

Although reasonable performances are obtained with the existing algorithms, gradient-based learning algorithms are still relatively slow in learning and also easily get stuck in a local minimum. Moreover, the activation functions used in these gradient-based tuning methods need to be differentiable it leads to slow convergence. And for environmental noises or if the noises are nonlinear, it is very difficult to achieve a good performance of the system and it makes it difficult to understand the clear signal, thus noise in speech recognition remains a paramount problem.

To overcome this, an automatic noise reduction system which is an integration of fuzzy noise classifier fuzzy adaptive filters is proposed. Though the environmental noises are fuzzy in nature, fuzzy inference systems are introduced in noise classifier and in noise canceller which will adapt to any environmental noise conditions

4. Fuzzy Noise Classifier

In any communication application, noise source classification has become a major part during the last decades. As the environmental noise is nonlinear and uncertain, fuzzy inference systems are proposed to adapt with any environmental situations for classification of noises. Fuzzy logic approach has a different conceptual model to differentiate objects than the ones used by the other common methods.

The fuzzy pattern classification of environmental noise presents a criterion to group a large range of environmental noise into a reduced set of classes of noise with similar acoustic characteristics and a larger set of background noise together with new multilevel classification architecture.

When the recorded noise data is too large to be processed and it is suspected to be notoriously redundant then the data will be transformed into a reduced representation set of features. The feature extraction process calculates numerical features that characterize the sample and it is needed to reduce the dimensionality of the data passed to the neural network.

Five commonly encountered non-stationary noise sources such as airport, babble, restaurant, horn and exhibition noises are chosen from the University of Texas-Dallas database (NOIZEOUS). From these noise signals feature sets are extracted using Line Spectral Coefficients (LPC), and MEL Frequency Cepstral coefficients (MELFC) and Independent Component Analysis (ICA) for dimensionality reduction.

For different dB levels of airport noise signal, 3 different feature vector sets each with the size of 1000 features are extracted using 3 different feature extraction processes. For the feature sets mean and standard deviation (SD) values are calculated. Similarly for all the noise signals with different dB levels (0dB, 5dB, 10dB and 15dB) mean and standard deviation values are calculated.

By analysing the range of mean and standard deviation values of all the noise signals, six input parameters (LPC_Mean, MELFC_Mean, ICA_Mean, LPC_SD, MELFC_SD, ICA_SD) are considered as inputs.

Table 4.1 shows the ranges of mean and standard deviation values for all the five different noises. With these inputs a fuzzy inference system is created with six inputs and five outputs and membership functions are also defined with the ranges of mean and standard deviation.

For the input signal to be classified, 3 sets of feature vectors are extracted and its mean and SD values are calculated. These six input values are given as input for this proposed fuzzy inference system to classify the predominant noise present in the noisy speech signal.

Figure 4.1 shows the fuzzy inference system for noise classification.

Table: 4.1 Ranges of Mean and Standard Deviation

Noise Type	AN	BN	RN	CN	EN
Range of Mean					
LPC	0.082-0.067	0.050-0.118	0.053-0.144	0.051-0.128	0.0582-0.255
MELFC	0.023-0.060	0.024-0.0938	0.035-0.070	0.023-0.058	0.042-0.074
ICA	0.88-2.337	1.823-3.512	1.35-4.575	0.585-2.115	1.984-4.235
Range of Standard Deviation					
LPC	1.7-13.7	2.2926-18.1	1.35 - 8.84	1.718-13.29	0.7477-10.4
MELFC	0.229-0.35	0.232-0.333	0.22-0.35	0.199-0.328	0.199-0.325
ICA	3.02-12.3	5.61-15.65	4.12-14.98	2.3-10.56	3.45-13.587

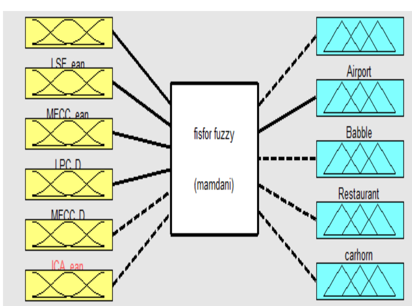


Figure: 4.1 Fuzzy inference system for noise classifier

Parameters of this fuzzy inference system are given below

No of inputs – 6 (LPC_Mean, LPC_SD, MFCC_Mean, MFCC-SD, ICA_Mean, ICA_SD)

No of membership functions – 5 (Very low, Low, Normal, High, Very High)

No of output - 5 (Airport, Babble, Restaurant, Car horn, Exhibition)

Membership function (MF) - Triangular member function

Membership functions of the first three inputs are shown in figure 4.2. This system is evaluated for noise input signal and also for noisy speech input signal. The performance matrix for noise only input and noisy speech input are compared.

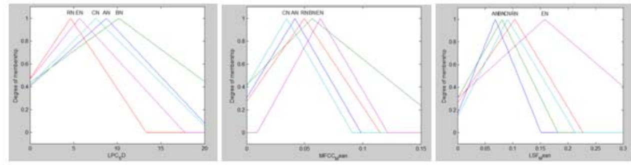


Figure 4.2 Membership function for LPC_mean, MELFC_mean and LPC_SD

Classification matrix for this fuzzy classifier is show in table 4.2 for noise only input and table 4.3 for noisy speech input.

Table 4.2 Confusion matrix for noise only input

For Noise signal as input					
Actual Noise	AN	BN	RN	CN	EN
AN	85	2	0	10	3
BN	0	88	8	0	4
RN	0	8	89	0	3
CN	7	1	1	91	0
EN	0	12	6	0	82

It gives the percentage of classification as the maximum of 91% by giving different dB levels of noise signal of the same class and 70% as the maximum for noisy speech signal input.

Table 4.2 Confusion matrix for noise only input

For Noisy speech signal as input					
Actual Noise	AN	BN	RN	CN	EN
AN	64	8	4	18	6
BN	3	61	26	2	8
RN	0	25	60	1	14
CN	21	4	3	70	2
EN	0	22	8	2	68

5 .Fuzzy Adaptive Filter

The principal of adaptive noise cancellation technique is to acquire an estimation of the unwanted interfering signal and eliminate it from the corrupted signal. Noise cancellation operation is controlled adaptively with the target of achieving improved signal to noise ratio.

5.1 Fuzzy Least Square filters

An adaptive filter is a filter that self-adjusts its transfer function according to an optimization algorithm driven by an error signal. Because of the complexity of the optimization algorithms, most adaptive filters are digital filters. The adaptive filter uses feedback in the form of an error signal to refine its transfer function to match the changing parameters. The adaptive process involves the use of a cost function, which is a criterion for optimum performance of the filter, to feed an algorithm, which determines how to modify filter transfer function to minimize the cost on the next iteration.

Classified pre-dominant noise source from the fuzzy classifier is used as a reference signal for the least square adaptive algorithms like Least Mean Square (LMS), Normalized Least Mean Square (NLMS) , Recursive Least Square (RLS). This proposed fuzzy adaptive system of noise cancellation uses an adaptive neuro fuzzy inference system (ANFIS) for updating its weight. For this adaptive neuro fuzzy system, the classified noise signal and its delayed versions are given as training data set. ANFIS parameters used for this adaptive system are given in table 5.1. This ANFIS based adaptive filter provides and output noise signal which will almost resembles the original noise signal present in the noisy speech signal. So instead of eliminating this classified noise signal from the corrupted speech signal, this generated noise signal from an adaptive system will be eliminated from the corrupted speech signal to obtain the clear signal signal.

Table 5.1 ANFIS parameters

ANFIS Parameter	Input noisy speech signal	Input noisy music signal_1	Input noisy music signal_2
No. of nodes	55	60	60
No of Linear parameters	80	91	83
No. of non-linear parameters	24	35	28
Total no. of parameters	104	126	111
No. of training data pair	1000	1000	1000
No. of fuzzy rules	16	18	16

Figure 5.1 shows the block diagram of fuzzy adaptive system for noise cancellation.

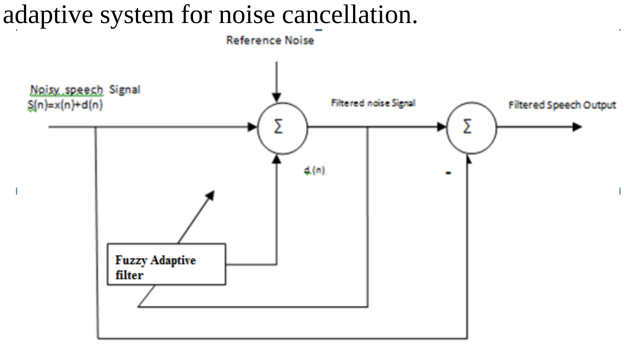


Figure 5.1 Fuzzy Adaptive Noise canceller

Table 5.2 SNR & Mean values of least square algorithms before and after using ANFIS

Algorithm	SNR (0dB)		MMSE (0dB)	
	Without Fuzzy	With Fuzzy	Without Fuzzy	With Fuzzy
LMS	21	27	3.256	1.87
NLMS	24	27	2.102	1.86
RLS	27	30	1.97	1.49

Table 5.2 shows tha performnce of LMS, NLMS and RLS algorithms are tested without and with using ANFIS system for weight updation.

5.2 Fuzzy Wiener filter

The goal of the wiener filter is to filter out the noise that has corrupted a signal. It is based on a statistical approach. A Wiener filter approach assumes that it should have the knowledge of the spectral properties of the original signal and the noise, and it seeks the linear time invariant filter whose output would come as close to the original signal as possible. The Wiener filter has solutions for two possible cases, the case where a causal filter is desired, and another where a non-causal filter is acceptable. The non-causal filter is simpler but is not suited for real-time applications. The Wiener filter is designed so as to minimize the mean square error.

Wiener filters are optimum in the sense of minimizing an appropriate function of the error, known as the cost function. The cost function that is commonly used in filter design optimization is the mean-square error (MSE). Minimizing MSE involves only second-order statistics and leads to a theory of linear filtering that is useful in many gradient applications.

In adaptive wiener filter, the optimum filter coefficients which are found out by the autocorrelation method are updated based on the error signal. Output of an adaptive wiener filter is further improved by updating the wiener coefficients. The noise signal and it delayed versions of noise signals are given as training data set for the fuzzy system used by the wiener filter. So it itself adjusts it parameters towards the noise signal. This generated noise signal from the adaptive system is subtracted from the noisy speech input to obtain the clear speech signal.

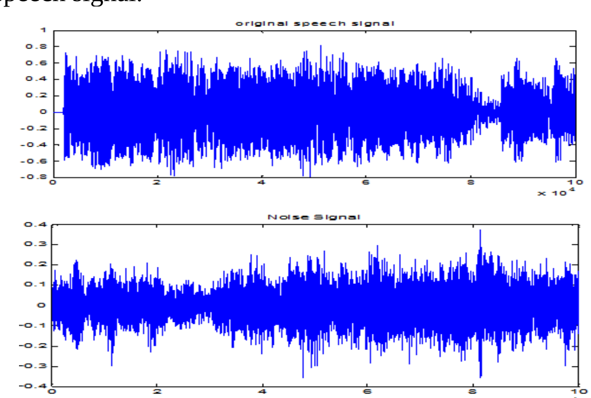
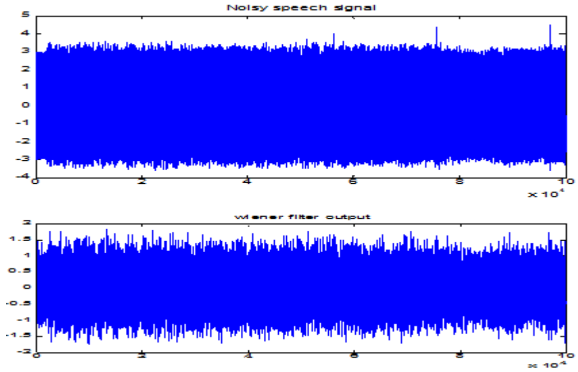
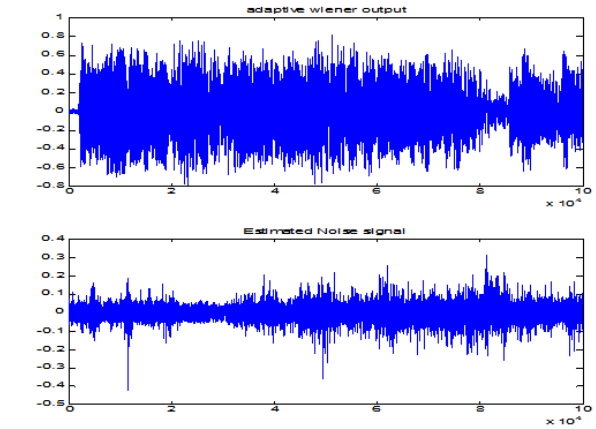


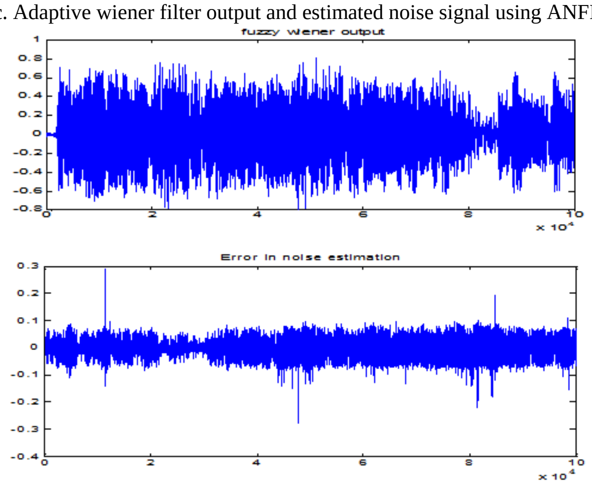
Figure 5.2 a. Oriiginal speech signal and reference noise sigal



b. Noisy speech signal and wiener filter output



c. Adaptive wiener filter output and estimated noise signal using ANFIS



d. Fuzzy wiener output and error in estimated noise signal

Figures 5.2 a,b,c,d shows the output of wiener , adaptive wiener and fuzzy wiener filter. Table 5.3 shows the performance of all the wiener algorithms.

Table 5.3 MMSE & SNR of wiener algorithms

Algorithm	MMSE		SNR	
	With Fuzzy	Without Fuzzy	With Fuzzy	Without Fuzzy
Wiener	0.61	0.72	27.5	25.7
Adaptive wiener	0.48	0.8	29.2	24.0
Fuzzy adaptive wiener	0.3	0.7	33.0	26.0

In this wiener filter, minimum mean square error reduces by 30% than the wiener filter and SNR increases by around 5dB.

6.Conclusion

It is very tedious to predict the class of noise signal in real time environments. Hence an automatic fuzzy based noise classification system along with fuzzy filters is proposed and its performances are compared. Proposed automatic environmental noise source classifier using fuzzy inference system shows classification accuracy rate between 60 to 70% for any kind of noisy speech input and 82 to 91% for noise input. In the adaptive filters, fuzzy wiener filter shows an increased performance in terms of SNR by around 5dB more than the least square adaptive algorithms by reducing the minimum mean square error by 30%.

References

[1] M.A.Abd El-Fattah, M.I.Dessouky , S.M.Diab And F.E.Abd El-Samie (2010), "Adaptive wiener filtering approach for speech enhancement", Journal of Ubiquitous Computing and Communication ,Vol 2. No.3. pp 23-31

[2]V.R.Vijaykumar, P.T.Vanathi (2007), "Modified adaptive filtering algorithm for noise cancellation in speech signals",Journal of Electronics and Electrical Engineering. No. 2(74). Pp 17-25

[3] Burak Uzkent, Buket D. Barkana' , Jidong Yang (2011), "Automatic environmental noise source classification model using fuzzy logic" Journal of Expert Systems, Vol.38. No.7, Pp 8751-8755

[4] Jingdong Chen, Jacob Benesty, Yiteng Huang and Simon Doclo (2006), "New insights into the noise reduction wiener filter" IEEE Transactions on Audio, Speech and Language Processing, Vol. 14, No. 4, pp 1218-1234

[5] A Sayed. A. Hadei, Student Member IEEE and M. lotfizad(2011) , " Family of adaptive filter algorithms in noise cancellation for speech enhancement" International Journal of Computer and Electrical Engineering, Vol.2.No.2 pp 307-315

[6]Fei Chen and Philipos C. Loizou(2010), "Speech enhancement using a frequency-specific composite wiener function" International Conference on Speech and Signal Processing, pp 4726-4729