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RESEARCH ARTICLE

REDUCTION OF NOISE TO IMPROVE QUALITY OF VOICE SIGNAL DATA USING NEURAL NETWORKS

Shashi[1], Soniya[2]

[1]M.Tech Scholar & DPGITM College (MDU Rohtak), India

[2]Soniya & DPGITM College (MDU Rohtak), India

[1] shashiahlawat9@yahoo.in; [2] sonya_leo2007@yahoo.co.in

Abstract— This report focuses on the neural network based solutions to noise cancellation problems. Firstly the Adaline network and its associated LMS learning rule are used to adaptively filter out noise interference. Secondly the use of a MLP and the back-propagation learning algorithm is used to improve noise problem. For each problem conventional solutions were studied and new solutions will be achieved through using neural networks. Implementation and testing was carried out using the Matlab programming environment and the Matlab toolboxes for neural networks and signal processing.

Keywords— ANN, ADALINE, BPNN, MATLAB, MLP

I. INTRODUCTION

Speech recognition is fundamentally a pattern recognition problem. Speech recognition involves extracting features from the input signal and classifying them to classes using pattern matching model. Performance of ASR system is measured on the basis of recognition accuracy, complexity and robustness. The deviation of operating conditions from those assumed during training phase may result in degradation of performance.

II. LITERATURE SURVEY

- Farhadrahdari et.al explores the capability of some intelligence methods, emphasizing the **Bayesian classifiers** to assess the quality of voice [3]. Five parameters namely CLP and ULP, codec mode, gender and language are inputted to the model to calculate MOS score. The performance of Bayesian classifier is compared with other models and the result is obtained in **F-measure**.For comparing the performance of different classifier the WEKA java based open source tool is used. Model is compared in terms of accuracy through F- Measure [1].

F measure = 2*precision *recall/precision +recall

- U.R.ALO et.al proposed **simulation technique** which measures the performance of VoIP over WLAN. Simulation technique works over various parameters of VoIP. This method specifically investigate the performance of VoIP over WLAN for increased in no of VoIP call, use of different coding scheme and increased no of workstation in video conferencing[5].

  Three scenarios are concluded:

  i. Increasing no of workstation leads to increase in jitter delay. Packet will arrive at different time and this jitter will make the voice difficult to understand.
  ii. Different codec scheme will lead in different jitter delay, end to end delay, WLAN delay and throughput delay
  iii. With this a determination of the actual number of VoIP calls that each wireless access point can adequately support with enhanced voice quality was made alongside with the encoding scheme that yields the best quality of service in WLAN[5].

- Yoanesbandung et.al aimed to maximize number of calls while maintaining a minimum level of quality of service in low speed IP networks. Various parameters in VoIP network including delay, packet loss and jitter are used to determine the level of quality of service by using extended e model. Objective is to find the maximum no calls in some given bandwidth capacities while maintaining a certain level of the quality of service. Using numeric estimation it is concluded that the optimum solution for some given bandwidth capacities can be achieved by applying G.723.1, 5.3 kbps voice coder, packet loss level less than 1%, jitter less than 80ms, and utilization network less than 80% [6] .

- Joarderkamruzzaman et.al developed an analytical model to estimate VoIP call capacity over Wi-Fi networks. This model is ITU T E model and is developed for single hop and multiple hop networks. The model insures the voice call quality by implying R-score defined in the ITU-T E model. VoIP capacity is investigated on different codec with the help of this model. The numerical result shown by this model shows that the capacity increases up to a certain level and then start decreasing as the aggregation level of codec increases [7].

- ITU-T E model is used for quality improvement in VoIP in combination with quality estimation. Kevin M.McNeill et.al proposed an adaptive jitter buffer play-out scheme to improve VoIP quality in wireless networks. The main reason for jitter delay is that voice packets are delay in the network randomly and jitter buffer is required to maintain consistently spaced play out of voice samples. Time series model is introduced here which enables the VoIP application to manage the jitter buffer to maintain a minimum play out buffer, while keeping the packet loss rate above a minimal threshold to maintain consistent voice quality. Simulation results show an improvement of 11% to 15% using metrics based on the subjective ITU E model (R-factor) when compared with other methods [8].

- Leandro carvalho et.al proposed an E model implementation for speech quality evaluation in VoIP. A voice quality measurement tool based on the ITU-T E model is discovered. Firstly ITU-T and ETSI specification of model are reviewed and errors are pointed and corrected. VoIP calls through the Brazilian national education and research network (RNP) backbone are used to verify the tool operation. Three mistakes in the E model calculation were found as defined in the ITU-T specifications and corrected them in the tool. These errors do not affect the E model but represent the inconsistencies between some E model statements. [10]

- KapilanRadhakrishan, HadiLarijani and Tom buggy proposed a method to predictthe perceived voice quality using a RNN model. Both IP network impairments (delay, jitter and packet loss) and non IP impairment codec (G711and iLBC) as voice quality affecting parameters to test their impact on perceived voice quality are used. An Intrusive objective speech quality evaluation algorithm PESQ is used to validate the results obtained from our RNN model. The outputs from RNN model and ITU-T PESQ MOS correlate well. This lends confidence to the conclusion that RNN model is accurate and more consistent. [14]

III. PROJECT OBJECTIVES

This report is concerned with exploring how artificial neural networks (ANN) can be applied beneficially in the field of communications. ANN has been widely used in the application of communication systems. This project will investigate the viability of using neural networks for two communication applications:

1. Using an Adaline network to eliminate background noise whilst speaking into a microphone.
2. Using a MLP to perform beam forming. This involves obtaining a desired signal whilst removing any noise or interference signals which may have come from different sources.

In both cases the objective is to obtain a desired signal which has had the background noise and any interference eliminated.

IV. OVERVIEW OF ANN

Neural Networks are an information processing technique based on the way biological nervous systems, such as the brain, process information. They resemble the human brain in the following two ways:

- A neural network acquires knowledge through learning.
- A neural network's knowledge is stored within inter-neuron connection strengths known as synaptic weights.

Neural networks are being applied to an increasing large number of real world problems. Their primary advantage is that they can solve problems that are too complex for conventional technologies; problems that do not have an algorithmic solution or for which an algorithmic solution is too complex to be defined. In general, neural networks are well suited to problems that people are good at solving, but for which computers generally are not. These problems include pattern recognition and forecasting, which requires the recognition of trends in data.

V. WHY USE NEURAL NETWORKS?

In communications the key features of neural networks are their asynchronous parallel and distributed processing, non-linear dynamics, global interconnection of network elements, self-organization, and high speed computational capability [1].

In 2007 Gardener et al published a paper [2] stating that ANNs are convenient for their adaptability, noise tolerance and potential for hardware implementation and fault tolerance. These characteristics are paramount when designing communication applications.

VI. THE ADALINE

Adaline stands for ‘ADAptive LINEar element’ and it has been one of the most widely used neural networks in practical applications. It was first published by Bernard Widrow and

Marcian Hoff and has a tight relationship to psychological learning theories. The Adaline has certain improvements over the original perceptron and belongs to the group of supervised learning; feed forward-only neural networks.

The transfer function for the Adaline is linear and therefore the output can have any value rather than just a binary output. However it is only possible for the Adaline to solve linearly separable problems. A diagram of the Adaline with its linear transfer function 'purelin' is shown in figure 1.

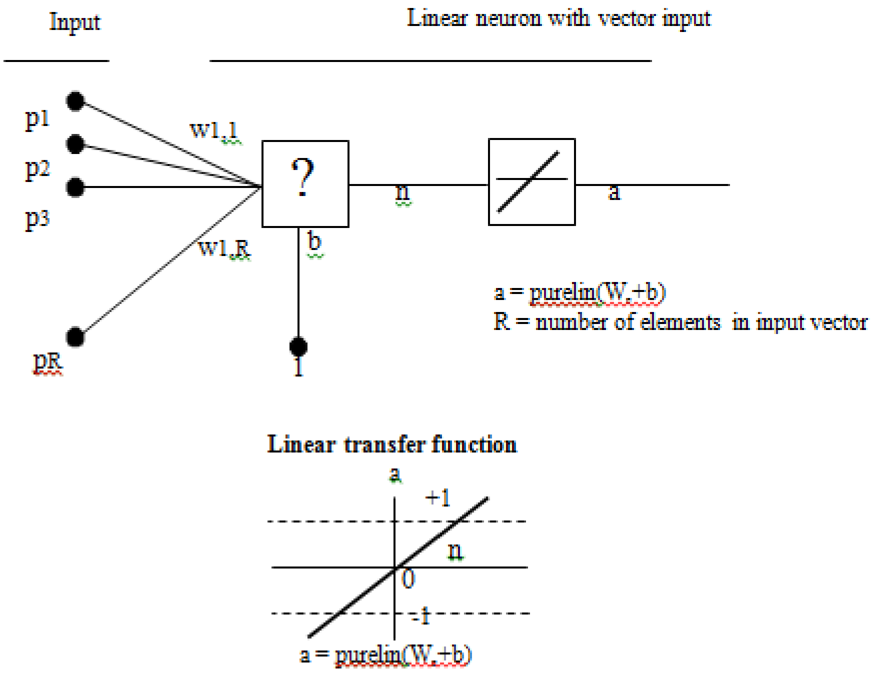

Figure 1: The Adaline (Adaptive Linear element)

VI. **Back propagation algorithm**

The Error Back-Propagation (BP) algorithm uses supervised learning to update the weights in the MLP. Initially the input vector is propagated through the MLP to produce an output. The generated output will differ from the desired output and subsequently produce a global error, e. This error can be minimized by passing the input through the MLP numerous times and altering the neuron connection weights, w k jeach time to obtain outputs closer to the desired outputs. For each training pass (epoch) the weights of the MLP are adjusted until the error gradient reaches zero.

VII. RESULTS AND ANALYSIS

The figure 2 is a interface which has been programmed in matlab with simple scripting. GUI is divided into two part: an offline GUI and online GUI.

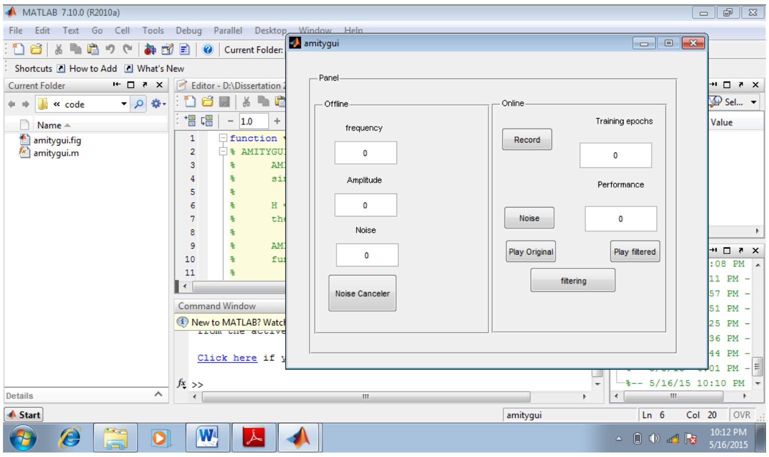

FIGURE 2

The figure 3 shows the system developed some related amplitude values corresponding to some delay which is shown in a sequence. The figure 1 plot 1 shows original signal followed by delayed signal and lastly the amount of delay at each instance.

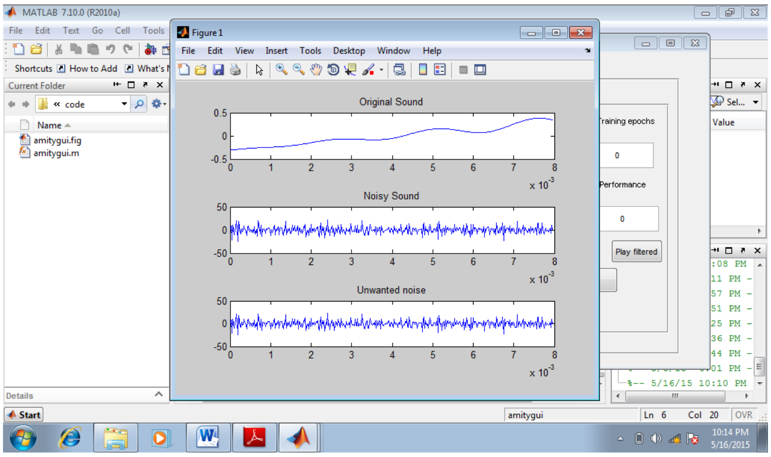

FIGURE 3

Figure 4 is an illustration of the error out of network 1 which is only 2.6 units in error validation, the system shows quite high value of error for the static signal generated by neural network.

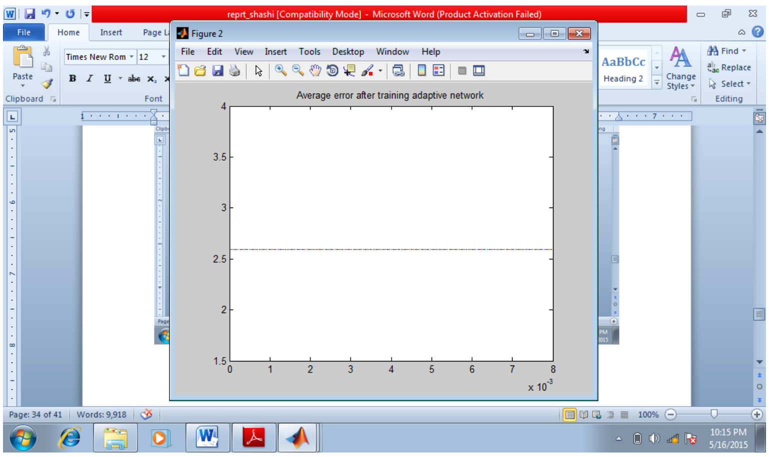

FIGURE 4

Figure 5 shows the error generated because of adaline network because of its property to adapt the network shows low blight of error in the presence of static signal.

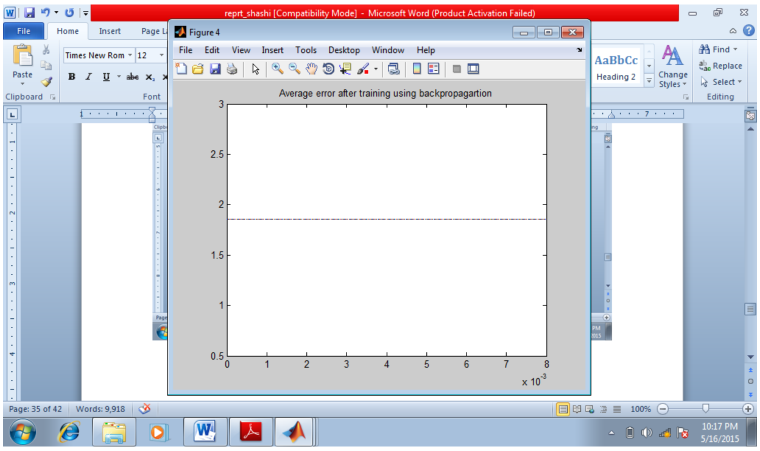

FIGURE 5

The figure 6 shows the signal taken in the terms of audio captured live from the audio acquisition device installed in the device readily in the laptop.

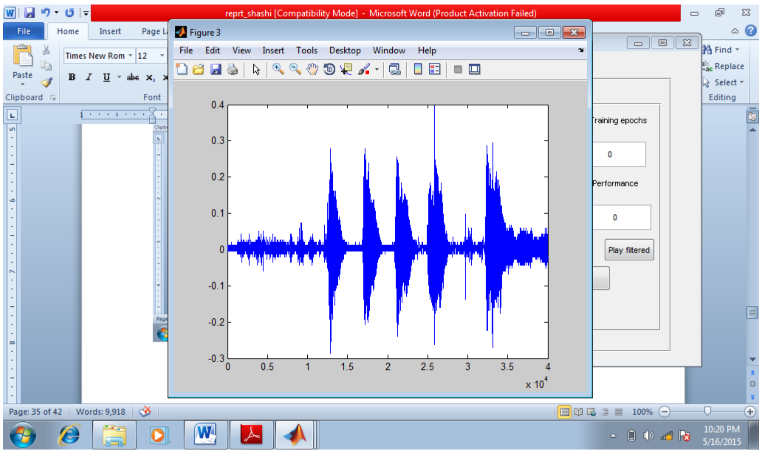

FIGURE 6

The figure 7 shows that the system has a error value close to 0 meaning that system developed using neural network with such values is reproducing the input as output with negligible error.

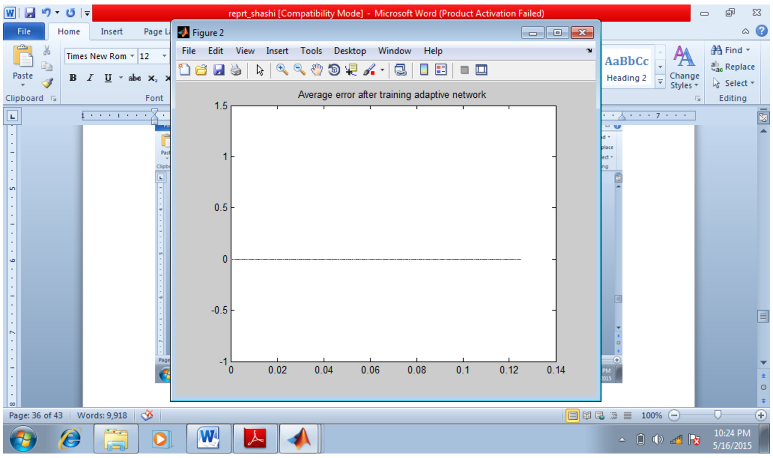

FIGURE 7

The figure 8 shows that the system has a error value close to 0.1 to 0.2 meaning that system developed using neural network with feedforwrad and back propagation training and such values is reproducing the input as output with error value visible at the graph.

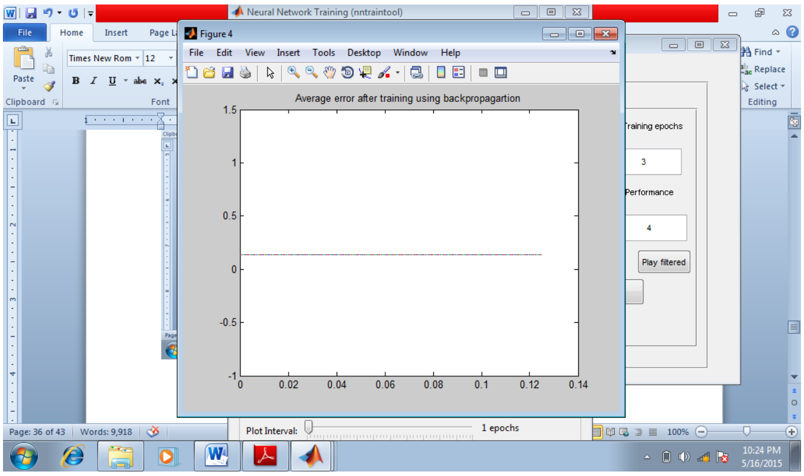

FIGURE 8

VIII. CONCLUSIONS

From the results it is clear to see that the Adaline is an effective system to eliminate noise in near real time. In a very short amount of time the desired signal can be approximated with very little error. Therefore it is a suitable alternative to the more conventional method of using filters. Conventional methods which use filters are not adaptive and therefore the Adaline is a better alternative. However it may be more expensive to implement and if there is an unusual disturbance from background noise the error may increase again for a short period of time.

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