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Neural Networks and Intelligibility Test for Sensory Neural Hearing Impairment

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ABSTRACT
Digital technology has made an important contribution in the field of audio logy. Digital signal processing methods offer great potential for designing a hearing aid but, today's Digital hearing aids are not up to the expectation for Sensory Neural Hearing Loss (SNHL) patients. Background noise is particularly damaging the speech intelligibility for SNHL persons. Transform domain adaptive methods can be used for noise reduction. But are having high computational complexity and the decorrelation efficiency is also less in most if the transformation methods. Artificial neural networks provide an analytical alternative to conventional techniques, which are often limited by strict assumptions of normality, linearity, variable independence etc. Hence this paper uses the neural network noise canceller to enhance the speech signal in digital hearing aid for SNHL person. Off-line implementations shows that the SNR improvement in direct time-domain neural network filtering approach with backpropagation training algorithm is almost equal to non-neural methods like transform domain adaptive filters.

KEYWORDS: Hearing Impairment, Neural-networks, Sensorineural hearing loss, SNR improvement, Digital Hearing Aid.

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I. INTRODUCTION

Hearing Impairment is the number one chronic disability affecting people in the world. In SNHL, the problem is not only with the transmission, but also with the processing of sound [3]. The SNHL patient's experiences difficulty in making fine distinction between speech sounds, particularly those having a predominance of high frequency Energy [6], [10]. He may hear the speaker's voice easily, but be unable to distinguish. For example between the words 'fat' and 'sat'[7], [9]. Amplification of the high tones may improve intelligibility, but in these circumstances dynamic range of the ear is a handicap [8], [4]. Since, the dynamic range of the impaired ear may not be sufficient to accommodate the range of intensities in speech signals. So, the stronger components of speech are perceived at a level, which is uncomfortably loud, while the weaker components are not heard at all [2], [5]. Hearing-impaired patients applying for hearing aid reveal that more than 50% of hearing problems are due to SNHL. So for only direct Adaptive filtering methods are suggested in the literature for the minimization of noise from the speech signal for SNHL patients [1], [11].

II. NEURAL NETWORK NOISE CANCELLER

The earliest and most straightforward use of neural networks for speech enhancement is a direct nonlinear time-domain filter. This is a multi-layer network, which is used to map a windowed segment of the noisy speech to estimate the pure speech. The number of inputs depends on the sampling rate of the speech signal. The number of outputs is usually equal to the number of inputs. Standard backpropagation training methods is being employed to minimize the mean-squared error between the target and the output of the network. This method forms a direct mapping of noisy speech to pure speech. Backpropagation networks also tend to be slower to train than other types of networks and sometimes require thousands of epochs. Fortunately, there are several fast versions, which increase the speed of backpropagation algorithms. If the learning curve is too steep, it represents that the network can learn quickly and vice-versa. A number of researchers have reported superior results over linear filtering by using neural network methods.

This paper presents speech enhancement in time-domain filtering, using neural networks. The major advantage of neural network approach is the flexibility. The direct time-domain neural network filtering approach is most applicable for reducing noises like babble, low frequency etc. Off-line implementation shows that the SNR improvement in neural network trained withbackpropagation algorithm is almost equal to non-neural methods.

The network is trained on 3 different sentences from different speakers. The network was tested using a speech signal, which not in the training set. The algorithm is evaluated for corrupted speech signals with different types of noises like cafeteria, low frequency and babble noise with different SNR. The various parameters like µ, number of layers in the network and number of neurons in each layer were changed and the performance of the algorithm was evaluated. The input signal is a speech sentence in English and is recorded with sampling frequency 22050 Hz in different noisy conditions to evaluate the effectiveness in removing the noise from the speech signal.

The performance of the algorithm was studied, for different values of μ, number of layers and numbers of neurons in each layers. From the studies we noticed that for $\mu = 0.01$, number of layers are three (one input layer, one hidden layer and one output layer) has best suitable for this particular application. The number of neurons used in the network are 40 in each layers (40:40:40) and the sigmoid function is used as the activation function.

For different input SNR, the output SNR and convergence ratios are calculated. Off-line implementations show that the SNR improvement in direct time-domain neural network filtering approach with backpropagation training algorithm is 11.02 dB for 1dB input SNR. The learning curve (steep curve) shows that, the network can converge to the optimal solution. The network outcome is tabulated in table and the resulted speech signal is as shown in figure 1.

SNR of the input signal in dB	SNR of the output signal in dB
-5	9.97
1	11.02
+5	12.29

Table 6.1 *SNR of the output signal for different input SNR to NN filtering*

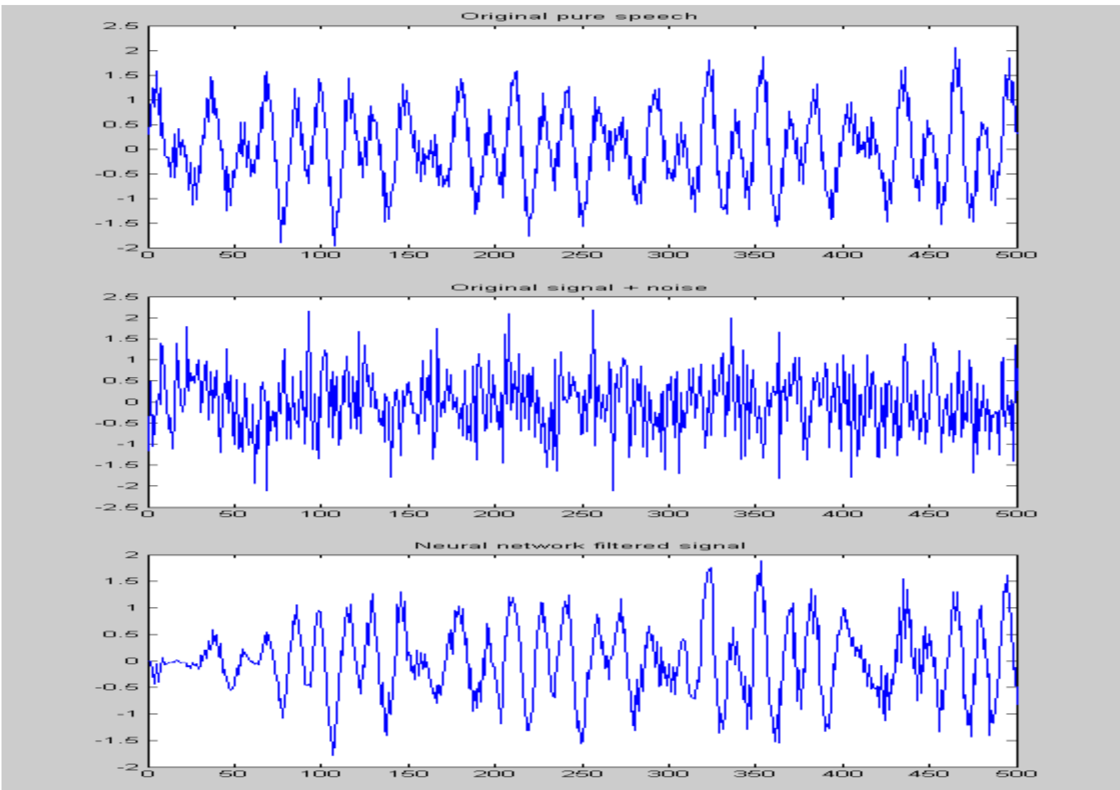

Figure 1 *Pure signal, contaminated signal and NN filtered signal*

III. INTELLIGIBILITY TEST

In order to measure the performance of clinical intelligibility of the algorithms listening tests were carried out. The tests were conducted on both hearing impaired and normal hearing persons. The experiment was carried out in a room whose size was about 4 m by 5 m. The main speaker and the noise source were placed 2.5 feet away from the microphones. For speech intelligibility test, we processed 10 sentences with different noise. These tests were performed on 15 subjects, 5 with normal hearing (Group 1), 5 with a mild to moderate SNHL (Group 2) and 5 with moderate to severe SNHL loss (Group 3). In the experimental evaluation, the target source was a male speaker reading sentences and interference consisted of 3 different types of noise (1) cocktail party noise (2) five speaker babble (3 male and 2 female) (3) low frequency noise. The noise level is varied to get different SNR. The subjects were listened the original, the noisy and the filtered signals. The percentage of correct responses was recorded. The results are displayed in Tables 2, 3 and 4 for –5dB input SNR. The result indicates that a considerable improvement is obtained, particularly for moderate to severe SNHL subjects.

Group1	Group 2	Group 3
96 %	78 %	63 %

Table 2 *Average intelligibility score for the noiseless signal*

Types of noise	Cocktail party noise	Babble noise	Low frequency noise
Group1	73 %	78 %	83 %
Group 2	31 %	34 %	38 %
Group 3	15 %	13 %	16 %

Table 3 *Average intelligibility score for the signal plus noise*

Types of noise	Cocktail party noise	Babble noise	Low frequency noise
Group1	95 %	92 %	93.5 %
Group 2	76 %	74 %	73 %
Group 3	68 %	65 %	63 %

Table 4 *Intelligibility improvements by neural network noise canceller for three groups of subjects.*

The result of recognition test for filtered signals is displayed in Table 3. It is seen that after adaptive neural network processing the intelligibility improvement is achieved. Neural network filter showed an average intelligibility improvement of 1 % with normal subjects, 1 % with mild to moderate SNHL subjects and 5 % with moderate to severe SNHL subjects as compared to adaptive least means square noise reduction methods with cocktail party noise.

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