Bird Species Recognizer using LMS Algorithm

K. Shivaani, Munukutla Vandana, Malaya Kumar Hota

Abstract: Birds play a vital role in many ecosystems, acting as both predators and preys for other living organisms. Therefore its important to monitor the population of various bird species in the environment in order to maintain balance in the ecosystem. This process will become tedious if it is done manually as it involves handling large sets of data at the same instant. We can do this by developing an automatic bird species recognizer which identifies the bird species based on bird songs and voice signals. In this research, we have used a tenth-order LMS adaptive filter to remove noise from bird voice signals which are recorded in different environmental conditions where different noise frequencies are present. The design of a tenth-order LMS adaptive filter using MATLAB has been implemented. The performance and characteristics of the filter for five different methods of LMS has been shown. After removal of noise from the noisy bird voice signal using LMS algorithm, we have made use of cross correlation to identify the bird species that it corresponds to. Signal to Noise Ratio (SNR) and Mean Square Error (MSE) of the filtered bird signals obtained using the variants of LMS like Normalized LMS, Sign-Data LMS, Sign-Error LMS and Sign-Sign LMS have been estimated and compared. We have made use of signal processing tool kits and various noise parameter schemes have been computed to show the effectiveness of the designed filter in the field of bird recognition.

Keywords: Adaptive filter, cross-correlation, LMS algorithm, Normalized LMS

I. INTRODUCTION

There are more than 9,700 species of birds in the world. People sometimes see birds and hear their sounds, but they don't know which kind of bird species they see. Domain experts can conduct bird identification manually; but, with growing amounts of data, it quickly becomes a repetitive and time-consuming process. We use speech recognition techniques to create an automated bird sound identification system to help people learn how to recognize bird species from their sounds.

When the user records a bird sound, environmental noises get added to it. Since the signal needs to be filtered before processing, we are using adaptive filter which uses Normalized Least Mean Square algorithm (NLMS) in our paper. For noise removal, filters such as low pass, Band pass, High pass, Band stop are available [1].

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But since the random noise contains multiple frequencies which start and end at different times, we choose adaptive filter which compares the input signal and desired signal at every instant and gives output. This makes the identification system dynamic i.e., each time the user gives an input, manual identification of different noise frequencies is not required.

II. LITERATURE REVIEW

Bird species recognition is done using various methods. Most of the existing methods are based on the VAD and HMM models [2]. Firstly, feature extraction is done. In this, Mel - Frequency Cepstral Coefficients (MFCC) representing the given signal were calculated from particular recordings. Secondly in the VAD Module, frames are classified into two classes, voices and non-voices (silence, engine sound, human voice, cough, microphone cracking, etc.). This way, sequences of frames of various lengths evaluated as voices are formed. Frames classified as voices enter further processing; the rest of the frames are not processed anymore. Thirdly, the results were obtained through a method of cross-validation across all data training models and extracting testing data from the training data. After calculating the testing data likelihood value for each model, the highest likelihood value model determines the correct bird species.

In our approach, we are trying to do similar processes using different algorithms which are also highly efficient. We have used LMS adaptive filter which helps in removing the bird voice signal from the noisy bird signal recorded in different environments where the noises are unknown. After that, we have used the cross-correlation method to compare the filtered signal with every original bird signal in the database and detect the corresponding bird species.

III. SYSTEM MODEL

A. Noise Cancellation using Adaptive Filters

When we record bird voices from the environment, there can be millions of noise signals along with the bird voice which we desired to record. Removing each of these noises one by one by finding the parameters of each signal like frequency will be a herculean task. Therefore clearly we can understand that noise is a random signal and in order to cancel a random signal, the basic requirement is that cancellation system has to be adaptive in nature. Numerous adaptive algorithms have been developed for design of noise cancellation systems based on prediction of filter coefficients to cancel the gradually changing noise signal[3].

Obtaining an ideal estimate of the noise from the contaminated noisy signals and hence obtaining an ideal estimate of the desired signal is the main purpose of noise cancellation using adaptive algorithms. It is used when there is spectral overlap between the desired signal and noise or when the band occupied by the undesirable noise signals is unknown or varies with time[4].

Adaptive algorithm begins its calculation from desired initial condition and uses information contained in the input data in order to estimate the weights of the filter. Since we keep updating the parameters of adaptive filters from one iteration to the next, the filter parameters become information dependent and hence this type of filter acts as a non-linear system[5].

An adaptive filter is characterized as a digital filter that has the ability of self modifying its transfer function based on some advanced algorithms and calculations. Generally used adaptive algorithms are Least Mean Square (LMS) and Recursive Least Square (RLS). Though RLS performs better than LMS, it is not useful in most of the practical applications because of its high computational complexity[6].

B. LMS Algorithm

LMS filter follows the approach of updating the filter weights so as to converge to the optimum filter weight. Initially, weights are assumed to be zeros and at each step, by finding the gradient of the mean square error, the weights are updated. Positive gradient indicates increasing error. So, weights are reduced. If the gradient is negative, weights are increased. LMS algorithm is sensitive to the scaling factor of input. Stability cannot be guaranteed using LMS [7]. The Normalized least mean square filter (NLMS) normalizes with the power of the input.

LMS Algorithm is one of the Stochastic Gradient Approaches.RLS Algorithm has less mean square error compared to LMS and NLMS Algorithms. MSE of RLS Algorithm is ten times less than that of the other two. But the complexity is much higher for RLS when compared with the other two methods[8].

Variable step size (μ) is essential for the adaptive algorithm to converge. The speed at which algorithm converges depends on μ. Proved studies showed that the values of μ can be decided as follows

1. μ will take the maximum value that ensures convergence, which can be found from the equations 1 and 2.

$$0 \le \mu \le \frac{2}{E_x} \qquad (1) \quad [9]$$

Where $E_x = \frac{1}{N}\sum_{n=1}^{N} |x[n]|^2$ (2) [9]

2. A constant value of μ is attained after analyzing the data received by the filter

Results showed that if μ becomes constant after $M^{1.5}$ consecutive iterations, best results are obtained.

Here M denotes filter length

The initial value of μ helps in accelerating the initial weight adjustment and the stable value is useful in fine-tuning the weights after adjustment. This helps in reducing computational complexity.

C. Observing and comparing the filtered signals using Fast Fourier Transform (FFT)

The bird voice signal enhancement technique enlightens upon the major use of bird voice degradation technique i.e. removal of environmental noise from the original bird voice signal. In this technique firstly the degraded signal i.e. original bird voice signal mixed with environmental noise is first converted to the frequency domain with the help of Fast Fourier Transform(FFT) tool in MATLAB Programming. The higher frequency noise components are then removed with the help of 10^{th} order adaptive LMS filter[9]. The resulting bird voice signal after filtration was then scaled and plotted with the original noisy signal to compare the extent of filtration. The same procedure is repeated for filtering noise using the other variants of LMS, namely, Normalized LMS, Sign-Data LMS, Sign-Error LMS and Sign-Sign LMS.

D. Database creation

In order to recognize the bird species corresponding to the recorded voice signal, a database containing the voices of all the known bird species is to be created. In this paper, for demonstration purpose, three sample birds from the database are taken, namely, jungle babbler, crow and sparrow.

E. Cross correlation

Cross connection is a standard technique for estimating the similarities between two signals. It helps us to find the similarities between a reference signal and one or more other signals by computing the relative displacement between each other[10]. Cross-correlation is applied between the recorded signal and each of the signal in the database. The cross-correlation plots corresponding to the recorded bird signal which is to be recognized and each of the signal from the database are also plotted. The bird signal from the database which has the least displacement from the bird signal to be recognized and hence the one which has the highest cross-correlation coefficient corresponds to the bird species which we have to identify. Hence, the bird species is recognized correctly.

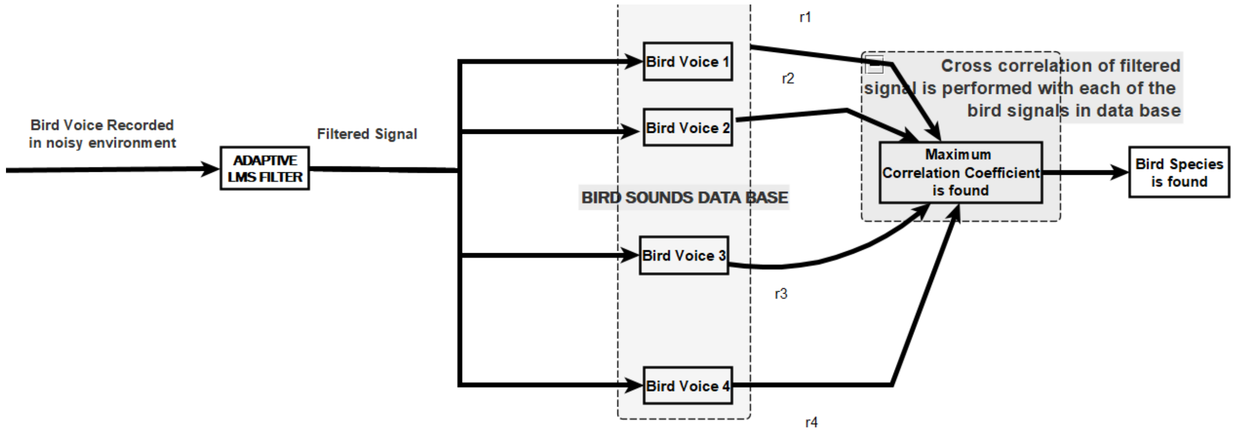

Fig.1. Block diagram of Bird Species Recognizer

Equations for LMS Algorithm [8]

$$y(n) = w^T(n-1)u(n) \quad (3)$$

$$e(n) = d(n) - y(n) \quad (4)$$

Equations to update weights [11]

$$LMS: e(n) = d(n) - y(n) \quad (5)$$

$$NLMS: w(n) = \alpha w(n-1) + \frac{\mu e(n) u^*(n)}{\varepsilon + u^H(n) u(n)} \quad (6)$$

$$Sign-Error: w(n) = \alpha w(n-1) + \mu sign(e(n)) u^*(n) \quad (7)$$

$$Sign-data: w(n) = \alpha w(n-1) + \mu e(n) sign(u(n)) \quad (8)$$

$$Sign-sign: w(n) = \alpha w(n-1) + \mu sign(e(n)) sign(u(n)) \quad (9)$$

where ,
y(n) is the filtered output at step n
u(n) is the vector of buffered input samples
w(n) denotes the vector of filter weight estimates at step n
e(n) is the estimation error at step n
d(n) is the Desired response at step n
μdenotes the adaptation step size
α denotes leakage factor

IV. SIMULATION RESULTS

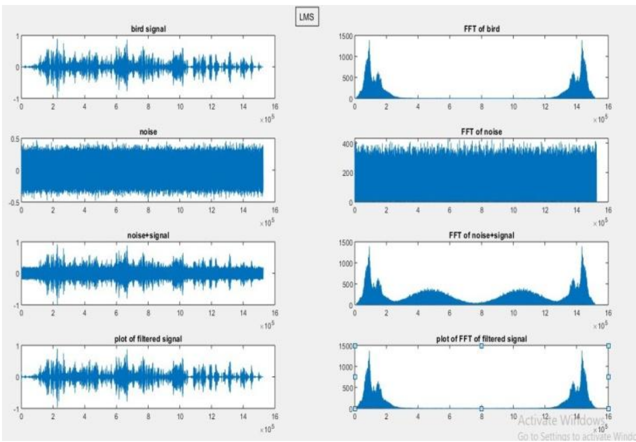

Fig. 2. Filtering using LMS method

Fig.2. represents the FFT of the original bird voice signal taken from the bird voices database, FFT of noise, FFT of noisy bird signal recorded from the environment and FFT of the filtered signal obtained by applying the method of LMS.

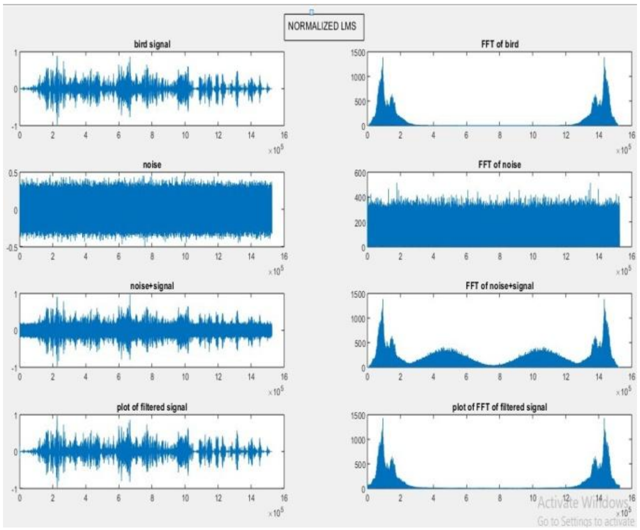

Fig. 3. Filtering using normalized LMS method

Fig.3. represents the FFT of the original bird voice signal taken from the bird voices database, FFT of noise, FFT of noisy bird signal recorded from the environment and FFT of the filtered signal obtained by applying the method of NLMS.

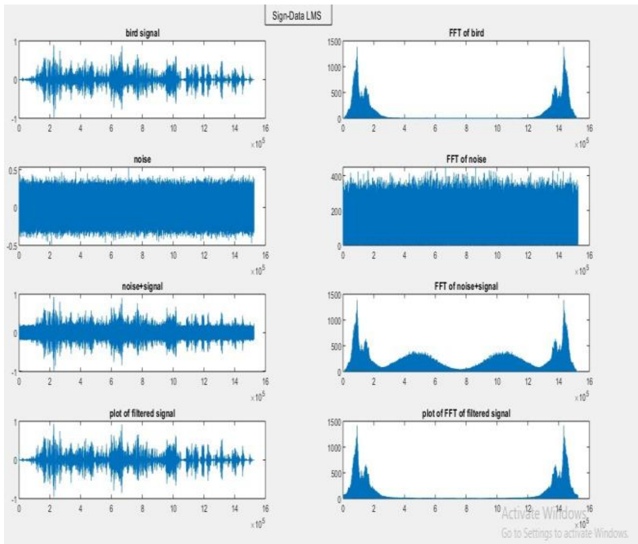

Fig. 4. Filtering using Sign-Data LMS method

Fig.4. represents the FFT of the original bird voice signal taken from the bird voices database, FFT of noise, FFT of noisy bird signal recorded from the environment and FFT of the filtered signal obtained by applying the method of Sign-Data LMS.

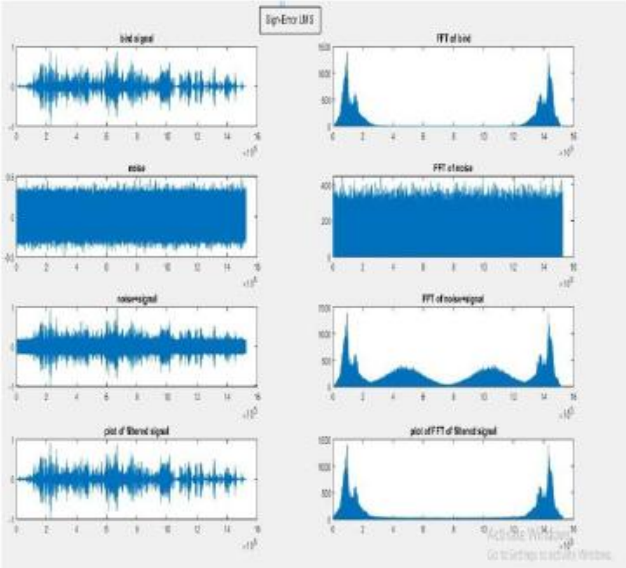

Fig. 5. Filtering using Sign-Error LMS method

Fig.5. represents the FFT of the original bird voice signal taken from the bird voices database, FFT of noise, FFT of noisy bird signal recorded from the environment and FFT of the filtered signal obtained by applying the method of Sign-Error LMS.

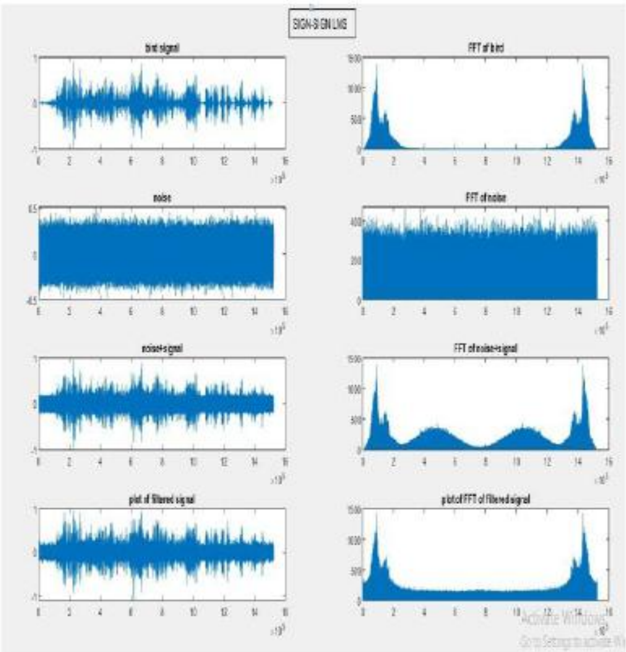

Fig. 6. Filtering using Sign-Sign LMS method

Fig.6. represents the FFT of the original bird voice signal taken from the bird voices database, FFT of noise, FFT of noisy bird signal recorded from the environment and FFT of the filtered signal obtained by applying the method of Sign-Sign LMS.

V. ANALYSIS

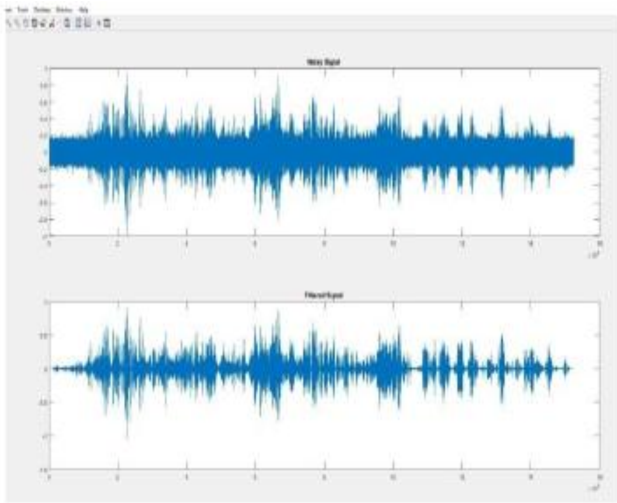

Fig. 7. Filtering noise from sample bird 1

Fig.7. shows the plots of the noisy signal of bird1 which is for example, here, jungle babbler and the filtered signal obtained after passing through NLMS filter.

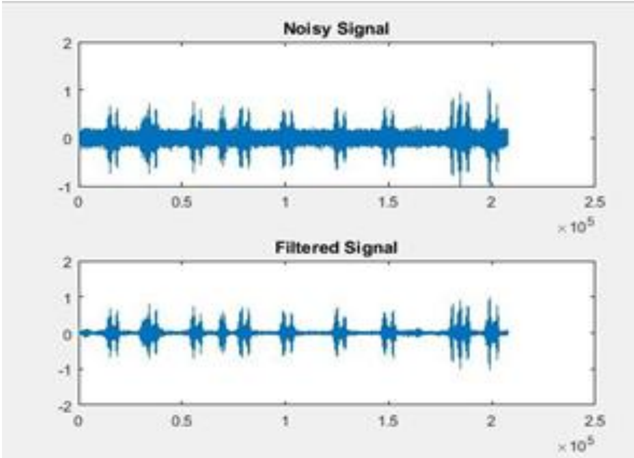

Fig. 8. Filtering noise from sample bird 2

Fig.8. shows the plots of the noisy signal of bird2 which is for example, here, crow and the filtered signal obtained after passing through NLMS filter.

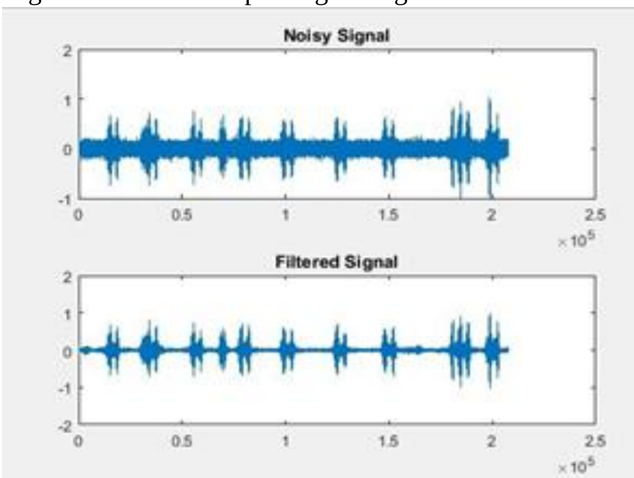

Fig. 9. Filtering noise from sample bird 3

Fig.9. shows the plots of the noisy signal of bird 3 which is for example, here, sparrow and the filtered signal obtained after passing through NLMS filter.

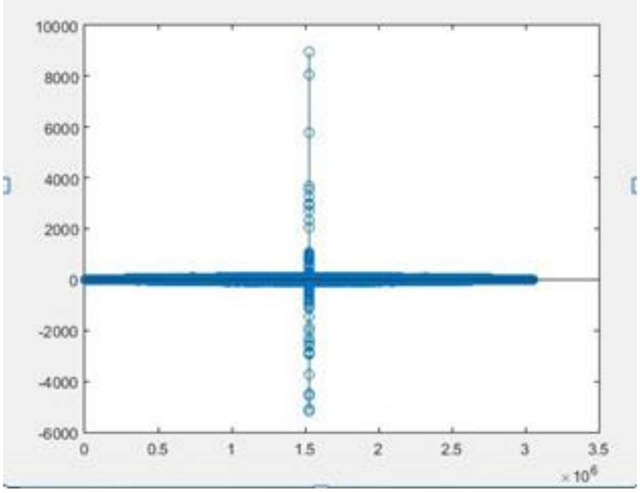

Fig. 10. Cross correlation plot of sample bird1

Fig.10. shows the cross correlation plot of the input signal with the first sample bird that is jungle babbler.

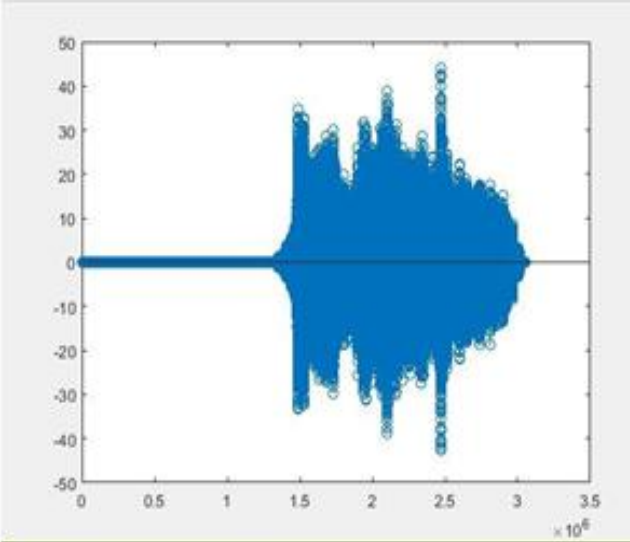

Fig. 11. Cross correlation plot of sample bird 2

Fig.11. shows the cross correlation plot of the input signal with the second sample bird that is crow.

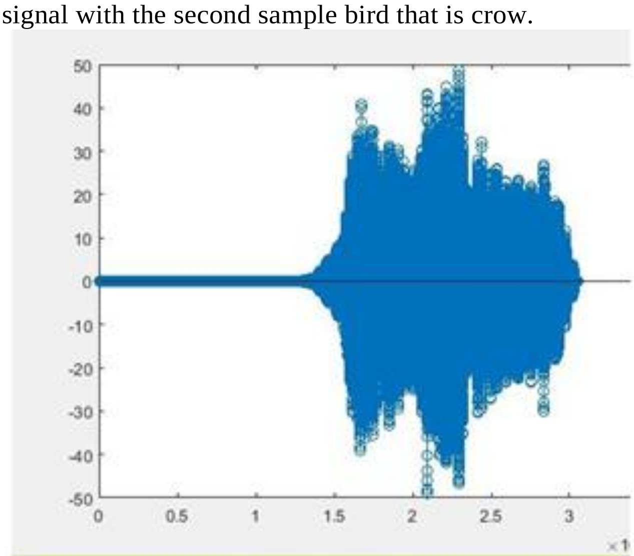

Fig.12.Cross correlation plot of sample bird 3

Fig.12. shows the cross correlation plot of the input signal with the third sample bird that is sparrow.

From the cross correlation plots, we can easily find that the given input signal is that of a jungle babbler. Here we have taken three sample birds for demonstration purpose. The same procedure can be extended to any number of sample birds in the database. Correlation won't work effectively if we don't remove the noise in the input signal. Therefore we use LMS algorithm here. The SNR and MSE of the different types of LMS have also been calculated and tabulated in Table-I to analyze and understand which is better. Then, we are correlating the noiseless input signal with each of the original bird sounds and hence recognizing the name of the bird.

VI. RESULTS AND DISCUSSION

$$SNR = 10\log\frac{(Signalpower)}{(Noisepower)} \qquad (10)\ [12]$$

$$MSE = \sum_{i=1}^{L}\frac{(Filteredsignal - originalsignal)^2}{(SignalLength)} \qquad (11)\ [12]$$

Table-I : SNR versus MSE using different methods of LMS Algorithm

METHOD	SNR	MSE
LMS	17.6688	0.00000208
Normalized LMS	17.7986	0.00017954
Sign-Data LMS	17.6775	0.00017467
Sign-Error LMS	17.666	0.00000149
Sign-Sign LMS	17.8645	0.00027853

For the applications which require only a very high value of SNR, Sign-Sign LMS gives the best results. For the applications which require only a very low value of MSE, Sign-Error LMS gives the best results. From the obtained results we can infer that, the optimum level of both high SNR and low MSE for bird species recognition is provided by Normalized LMS method.

On comparing the different methods of LMS algorithm we infer that the efficiency of filtering and bird recognition is in the following order.

1. Normalized LMS
2. LMS
3. Sign-Data LMS
4. Sign-Error LMS
5. Sign-Sign LMS

VII. CONCLUSION

The general method used for bird recognition is using HMM and VAD models. But, in this research, we have come up with a new approach for bird species recognition using LMS algorithm and cross-correlation. The major problem that we face during the process of bird recognition is noise cancellation because lot of unknown frequencies of noise signals are mixed with the desirable bird voice.

The adaptive filter algorithm used in this model helps us to eliminate all types of noises and extract only the bird voice signal from the noisy audio recorded for bird recognition. Thus this method is simpler and is applicable for any environmental condition. The accuracy of this method is also high which can be observed from the results and graphs obtained. From our analysis, we have found that NLMS is the most efficient method over the other LMS methods.

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