Analysis of BSC and AWGN Channel Distortion Effect on Sound Signal in Active Noise Cancellation Application

Janak Kapoor, G.R Mishra, Ajita Pathak, Manish Rai

Abstract: **Adaptive filtering algorithms holds wide application in the active noise cancellation. On applying the adaptive algorithm, the main factor of concern is to generate such filter coefficients that minimize the error between actual signal and predicted signal so that noise is cancelled in an effective manner. Various adaptive filtering algorithms applied on various filtering models are available in literature and research is still on to make this active noise cancellation process more and more practicalfor real world applications. This paper presents one such modelin which active noise cancellation is achieved through the use of biquad filter followed by band pass filter considering the effect of binary symmetric channel(BSC)and additive white Gaussian noise(AWGN) channel on signal transmission. The output is analyzed byapplying the discrete cosine transform(DCT), fast Fourier transform (FFT) and analytic signal transform to the signal recovered after noise cancellation. The simulation results prove the effectiveness of the proposed model in reducing the channel distortion effect on the transmitted signal in the presence of added noise.The signal is recovered using normalized least mean square (NLMS) algorithm.**

Keywords: Active noise cancellation, Biquad filter, Discrete Cosine Transform, Binary Symmetric Channel.

I. INTRODUCTION

Adaptive algorithms are the main driving force behind the adaptive filtering i.e., adaptive filters are known to be adaptive due to the variable filter coefficients generated by the application of adaptive algorithms. As noise is a random signal it can only be predicted. To predict and cancel noise the filter should be adaptive in nature, the application of cancellation of noise by use of adaptive filtering is known as active noise cancellation (ANC).

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It is a well-known fact that when things are based on prediction there are more chances of error and efforts has to be made to reduce this prediction error to minimum. Likewise, is the case with the active noise cancellation many algorithms have been developed and many types of models has been implemented by researchers and published in literature with the aim of reducing the difference between the desired signal and the predicted signal. Still research is going on in this field and this paper presents one such approach to make the active noise cancellation more effective by use of biquad filter and the paper also analyses the effect of the BSC and AWGN channel on the performance of active noise cancellation using normalized least mean square algorithm. The basic block diagram of an active noise cancellation model is shown in the fig 1 given below:

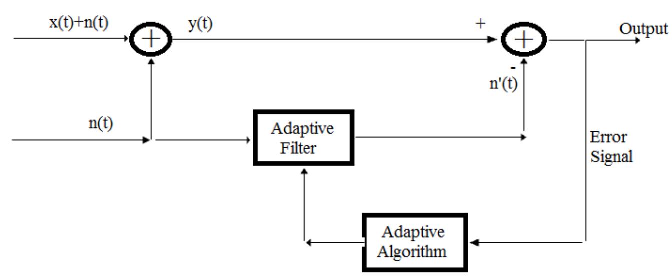

Fig 1 Basic block diagram of active noise cancellation (ANC) system

The basic blocks as shown in fig 1 consists of an input signal x(t) which is corrupted by the noise signal n(t) the work of the adaptive filter block is to predict the noise signal n(t) so that when it is subtracted from the corrupted signal and the original signal x(t) can be obtained without noise distortion. The prediction cannot be accurate in one go so a feedback path is required in which the difference signal which is the estimate of the input signal is used as feedback signal to adjust the filter coefficients using adaptive algorithm. The research area includes the adaptive algorithm block combined with the filter block. Various adaptive algorithms used are least mean square (LMS), recursive least mean square(RLS), normalized least mean square(NLMS), variable step size LMS(VSSLMS), block LMS(BLMS), filtered-x-LMS(FxLMS) and many more having various advantages and disadvantages has been used by researchers from time to time and are published in literature[1].Similarly the adaptive filter used can be an infinite impulse response (IIR) or an finite impulse response (FIR) but FIR filter being an all zero and no poles filter is more stable as compared to IIR therefore used more often in adaptive filtering applications. Theproposed model is based on the normalized least mean square algorithm (NLMS). The input signal is a pre- recorded bird chirp sound in wav format, noise signal is also a prerecorded wav format media file. The input signal is passed through a biquad filter before applying it to the desired input port of the NLMS filter block, the corrupted signal is passed through a binary symmetric channel block and AWGN channel. The output of the NLMS block is analyzed using analytic signal, FFT and DCT transforms. The simulation results show performance of NLMS algorithm in consideration with different channel distortions. The paper is divided into four sections the section 1 consists of the introduction, the section 2 gives a small literature review and the proposed idea, section3 presents the proposed model and section 4 gives the simulation and result analysis [8]-[12] followed by conclusion and references.

II. LITERATURE REVIEW AND PROPOSED IDEA

The small insight of the literature reviewed before coming to the proposed model is described as under. There are many techniques published in literature for implementing the active noise cancellation all with the aim of addressing one of the constrains of high computational complexity, high power consumption, huge processing time for prediction and cancellation of noise, high cost of hardware and accuracy in cancellation[2].An active transmitter cancellation scheme with antenna interface integrated on a CMOS chip instead of high-cost DSP processor is used for active cancellation and has resulted in the efficient cancellation of the transmitter phase noise in the receiver band [3].Wide band low noise amplifiers without inductor front ends consume more power as compared to front ends with inductors so a compensation between power and efficiency is an important factor to deal with[4].Use of wave domain adaptive algorithm can reduce noise to a significant level in spatial region and has a faster convergence rate as compared to the conventional multi point algorithm[5].Feed forward based noise cancellation algorithm using sub sampling phase detector can be used to improve the drawbacks of very high jitter and noise sensitivity present in ring oscillator, phase locked loop [6].A de-correlation-based algorithm can be used to estimate the adaptive filter coefficients[7].Likewise the idea proposed in this paper is based on the NLMS algorithm block with the use of the biquad filter for filtering the input signal. The corrupted signal is passed through a binary symmetric channel and results are analyzed and compared with the conventional model without the biquad filter [13]-[20].

III. PROPOSED MODEL

The block diagram of the active noise cancellation model proposed is shown in fig 2 as under:

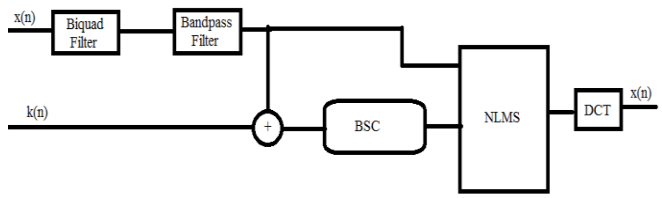

Fig 2 Proposed active noise cancellation model

Fig 2 given above shows the proposed model which consists of the input signal x(n) which is an audio signal of prerecorded bird chirp sound. The input signal is passed through a biquad filter having phase and magnitude response shown in fig 3 below:

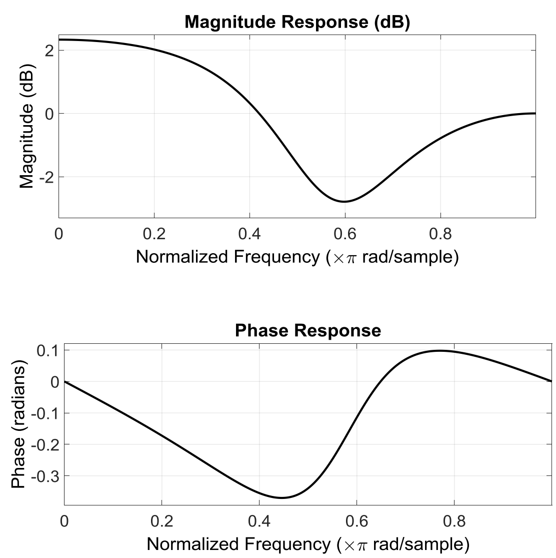

Fig 3 Magnitude and Phase response of a Biquad filter

The biquad filter is an IIR filter used as a basic building block for complex filters, it reduces the coefficient sensitivity problem [8]. The output signal of the biquad filter is passed through the bandpass filter with the magnitude and phase response shown in the fig 4 below, the magnitude response is the plot of filter characteristics in terms of magnitude in dB plotted in response of normalized frequency in radian per samples and the phase response of the filter is the plot of phase response characteristics of the filter showing phase response of the filter in radians with respect to the normalized frequency in radian per second. Both amplitude and phase characteristics are important for analyzing any filter:

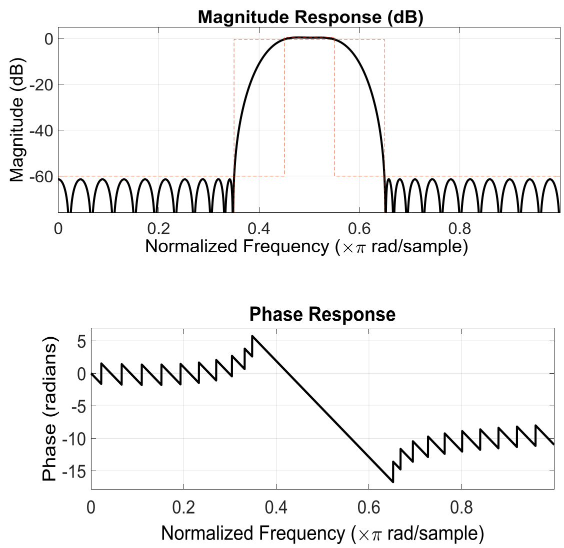

Fig 4 Magnitude and Phase response of bandpass filter

The output from the bandpass filter is applied to the adder which adds the input signal x(n) with the noise signal k(n), the combined signal x(n)+k(n) is the noise corrupted signal is passed through a binary symmetric channel (BSC). The binary symmetric channel adds binary error to the input signal, the proposed model is also analyzed using additive white Gaussian noise channel (AWGN) channel as shown in fig 5 given below:

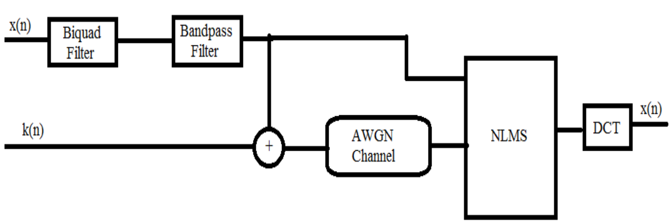

Fig 5 Proposed model with AWGN channel

The additive white Gaussian noise channel introduces the white gaussian noise to the input signal. The motive behind adding different channels is to analyze their effect on input noise corrupted signal. The signal from the channel is now applied to the normalized least mean square (NLMS) filter block as the input signal. Noise and channel errors have to be removed and desired signal x(n) is to be obtained from the input signal. Normalized least mean square algorithm is an upgraded version of the basic least mean square (LMS) algorithm. It has an advantage in adaptive filtering of large dynamic range signals, the basic equation of the normalized least mean square algorithm is given as under:

$$h(n+1) = h(n) + \frac{\Delta}{\|x(n)\|^2}\, e(n)x^*(n)$$

dividing the step size by the norm of data vector results in variable step size defined by equation below[8].

$\Delta(n) = \frac{\Delta}{\|x(n)\|^2}$ At each iteration the step size is inversely proportional to energy in the data vector x(n). In the case of slowly fading communication channels in the communication process numerical instabilities may arise if the norm of data vector is small. To avoid such numerical instabilities a small positive constant is added to the denominator of the above equation as shown below:

$$\Delta(n) = \frac{\Delta}{\delta + \|x(n)\|^2}$$

δ represents any small positive constant added to reduce instability. The normalized least mean square adaptive algorithm block gives the input signal x(n) after removing the noise k(n) and the induced channel error. The output from the normalized least mean square filter block is processed using discrete cosine transform, fast Fourier transform and analytic signal. Results for both binary symmetric channel and additive white gaussian noise are obtained and given in next section.

IV. SIMULATION AND RESULTS

SIMULINK tool has been used to test and simulate the proposed model and the simulation results are shown below.

Fig 6 shown above shows the signal x(n), the noise signal k(n), noise corrupted signal after passing through the additive white gaussian noise channel (AWGN) and recovered signal after applying the transform (DCT), thefast fourier transform (FFT) and analytic signal transform. It also shows the frequency spectrum of the recovered signal. Theeffect of the additive white gaussian noise channel(AWGN) can be clearly seen on the noise corrupted signal. The recovered signal plot and the frequency spectrum shows that how efficiently the signal has been recovered in the proposed model and it is seen that the effect of the channel, the noise has been nullified.NLMS active noise cancellation algorithm has been effectively implemented. Fig 7 shows the simulation results of proposed model using binary symmetric channel (BSC) channel, signal analysis by applying the discrete cosine transform (DCT), the fast Fourier transform (FFT) and the analytic signal at the output and the output spectrum in all the three cases is also shown in the figure 7. The plots shown in figure 7 are simulation results of Simulink model showing the signal plots in the time domain. Signal spectrum is plotted using the spectrum analyzer is a plot in dBm with respect to frequency in kilohertz. The time is in seconds and varies in the scale from 0 to 10. Plots shows different results for the variation output as channel varies.

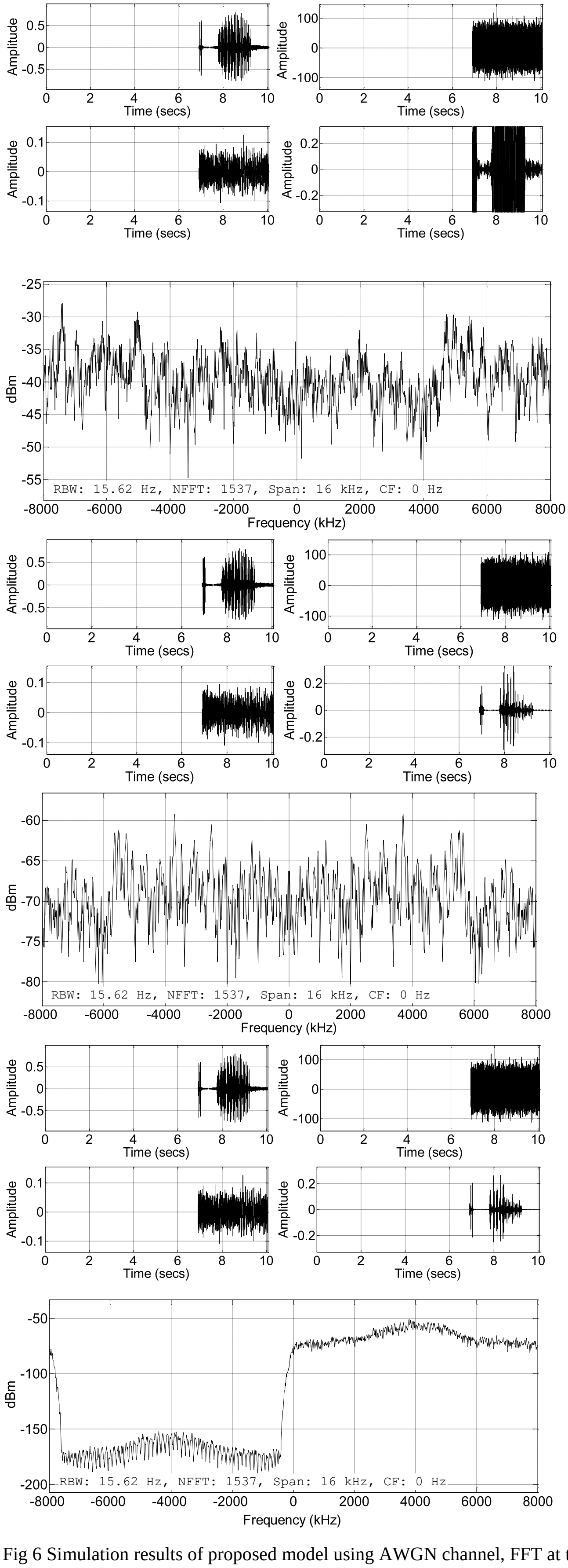

Fig 6 Simulation results of proposed model using AWGN channel, FFT at the output and the output spectrum

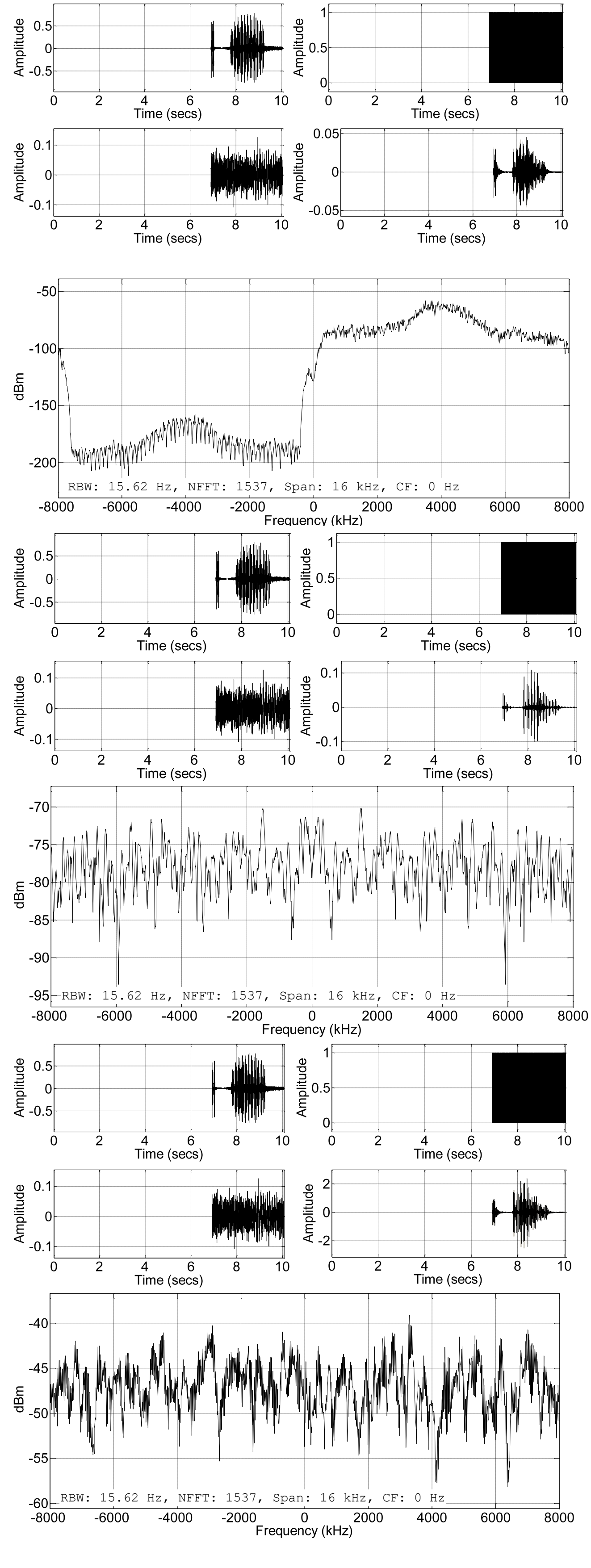

Fig 7 Simulation results of proposed model using BSC channel, FFT at the output and the output spectrum.

Fig 7 shown above shows the signal x(n), the noise signal k(n), the noise corrupted signal passed through binary symmetric channel (BSC) and the recovered signal after applying the fast fourier transform. It also shows the frequency spectrum of the recovered signal. The effect of the binary symmetric channel (BSC) can be clearly seen on the noise corrupted signal. The recovered signal plot and the frequency spectrum show that how efficiently the signal has been recovered in the proposed model and the effect of the channel, the noise has been nullified and the active noise cancellation algorithm has been effectively implemented.

V. CONCLUSION

From the simulation results shown in the fig 6 and fig 7 given above it is seen that in addition to the noise signal, channel also has a considerable distortion effect on sound signal. This channel distortion effect is analyzed in the proposed model by implementing additive white gaussian noise Channel (AWGN) and binary symmetric channel (BSC)in the signal path. In order to recover original signal through active noise cancellation the challenge before adaptive filter is not only to adaptively predict the added noise but also to adapt with the distortion caused by channel characteristics. This fact that channel characteristics is also a major challenge before active noise cancellation techniques has been justified and proved in this paper and the proposed model depicts that normalized least mean square algorithm (NLMS) has proved to be an effective tool in noise cancellation through adaptive filtering.

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