Audio Improvement using Symlet Wavelet Transform and Remez Exchange Algorithm

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Abstract - Filters became important when noise in signals causes loss of actual information. Many filters have been developed and still improving. Paper presents an audio enhancement system that aims to recover common audio signal from received noisy audio signal after long communication. We do so in the context where each received is uniquely corrupted. To this end, we propose a method of DWT remez exchange to analyses on synchronized inputs. In the proposed model, some of the parameters are fixed. Our model also allows for prior knowledge about the parameters of the model, A highly optimized core discrete Remez part of the Parks-McClellan (PM) filter along with wavelet type sym6 filter been proposed in the paper. As available PM and Wavelet filters are already an achievement and works quite good but some time we requires to have filter highly resolution filters which have high SNR, low BER and very low MSE for any input signal.

Keywords: Bit Error Rate (BER), Finite Impulse Response (FIR), Parks-McClellan (PM), Remez exchange, Signal to Noise ratio (SNR).

I. INTRODUCTION

Many applications where audio enhancement requires because of Noise or distortion or fading, paper work presents a solution for this problem by using Remez Exchange algorithm followed by DWT. In order to enhance the audio quality many researchers proposed their unique methods. The Remez exchange algorithm can be a good solution for optimization audio enhancement that is commonly used in the design of FIR filters. It is popular because of its flexibility and computational efficiency. Also known as the Parks-McClellan algorithm, it works by converting the filter design problem into a problem of polynomial approximation. The optimal Chebyshev FIR filter can often be found effectively using the Remez multiple exchange algorithms (typically called the ParksMcClellan algorithm when applied to FIR filter design). The Parks-McClellan/Remez algorithm also appears to be the most efficient known method for designing optimal Chebyshev FIR filters.

The discrete wavelet transform (DWT) is a linear transformation that operates on a data vector whose length is an integer power of two, transforming it into a numerically different vector of the same length. Wavelets have an important application in signal denoising. Properties of Discrete wavelet transforms were employed to recover a signal from the signal with noise. The process of filtering can be broken into further steps which are: Analysis, Applying wavelet transform, Analysis Step Selecting an appropriate wavelet was a very important task in this step. The wavelet chosen should be similar to the signal that has to be filtered to give the best possible results, applying wavlets to filter out the signal, Parks-McClellan or remez exchange based filtering approach gives us a signal filtered signal with very small amount of noise present that small amount can be negligible in some application but there are application like secure data communication, multiplexed broadcasting, real time digital data communication where noise cannot be ignore. So it is requires to have a filter which can remove maximum noise and low BER and low MSE than available technique and it is the necessary requirement of any communication system. So paper proposed a combination of Remez exchange FIR optimal chebyshev followed by symlet filter.

Problem Statement: The only problem with available work is that its performance depends on the signal type some filters are good with audio signal some are good with high frequency signals. Proposed works aims to develop a general filter which can filter any type of input signal. A combination of PM filter of order 40 and 6 frequency edges in preceding of Wavelet 'sym6' type of order 40 have been develop and tested with two different test scenario and it is been observed that this specific combination gives very low MSE as compare to available filters.

II. PERFORMANCE PARAMETERS

In statistics, the mean squared error (MSE) of an estimator measures the average of the squares of the "errors", that is, the difference between the estimator and what is estimated

$$\text{MSE}=\sum_{j=1}^{R}\sum_{i=0}^{C}({x^2}_{ji} - {x'_{ji}}^{2}) / \text{R*C}$$

Where j are the number of rows and i are the number of column x is the original signal and x' is received signal. Peak Signal-to-noise ratio (PSNR) is a measure that compares the level of a desired signal to the level of background noise

$$\text{PSNR}= 20 \log_{10}(256^2 / \text{MSE})$$

III. PROPOSED METHOD

Figure 1 presents the proposed modules it proposed, here first wavelet filtering module filter the noisy signal with wavelet which is type 6 symlet (sym6) which performs an interval dependent denoising of the noisy signal, using a wavelet decomposition at the level '5' with a wavelet which name is 'sym6'and perform soft thresholding. Next type 6 REA is a Parks-McClellan optimal equiripple FIR filter design, FIR filter which has the best approximation to the desired frequency response described by F and A in the minimax sense. F is a vector of frequency band edges in pairs, in ascending order between 0 and 1. 1 corresponds to the Nyquist frequency or half the sampling frequency. At least one frequency band must have a non-zero width. At least one frequency band must have a non-zero width. A is a real vector the same size as F which specifies the desired amplitude of the frequency response. In present work because denoising is our aim F is chosen [0 0.14 0.15 0.16 0.17 1] and A is [1 1 1 1 1 1] the value of F can be vary as per the input signal. The order of FIR filter is '40', As the order 40 in FIR-PM filter gives delay of 40/2 samples hence it is requires to have 20 sample advance the output signal.

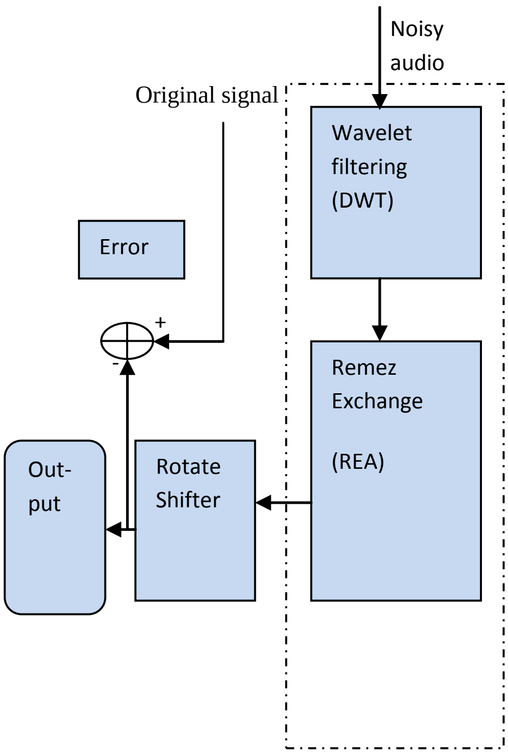

Figure 1: The Proposed Design

IV. EXPERIMENTAL RESULTS

The results is been observed for two different input signal a chirp signal of 25Hz to 125Hz and audio file of 44200Hz sampling frequency with different AWGN noise value in 'db'. Different Noise needed for testing all type of possible real life situations. Proposed design DEA-RSA the order of FIR is 40, type 6 of REA and symlet order 6 with 5 level decomposition wavelet fitter is been selected.

Case study-1: for chirp signal figure 2 shown below shows actual input chirp signal and its 30db awgn noisy signal

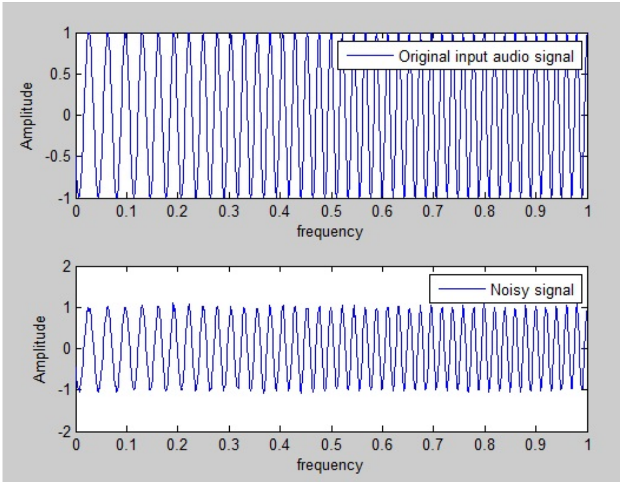

Figure 2: the original and chirp signal

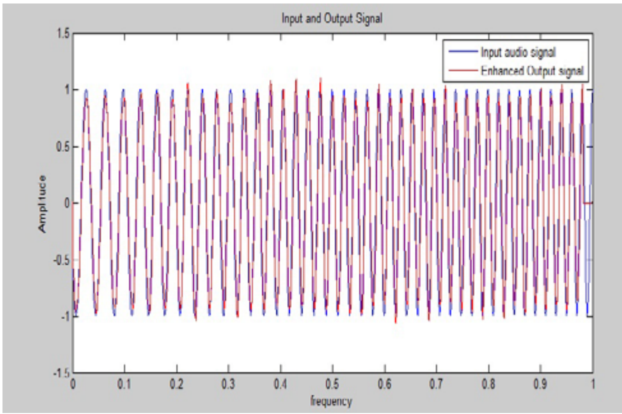

Figure 3: Original chirp shown by blue filtered output chirp signal by red

Figure 3 shown above shows analytical comparison of given input chirp signal and filtered output signal.

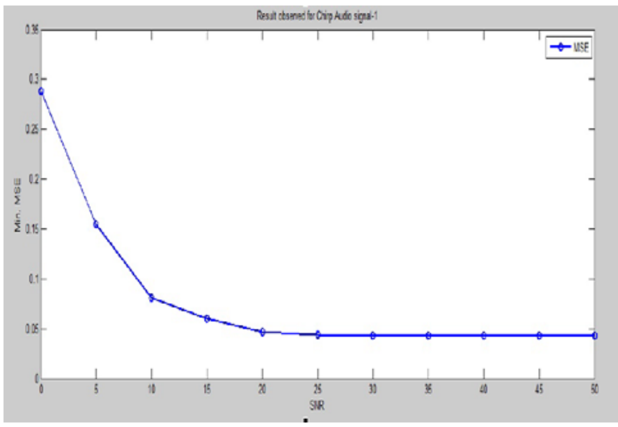

Figure 4: MSE for different noise value in Chirp signal

Figure 4 shown below shows a comparative analytical plot which is been develop for different noise quantity in input and output observed, the comparison is been done as mean square Error between chirp input and chirp output.

Table 1: MSE observed in chirp for different Noise level

S.No.	SNR	Min.MSE
1	0	0.0105
2	5	0.0078
3	10	0.0055
4	15	0.0038
5	20	0.0029
6	25	0.0024
7	30	0.0022
8	35	0.0021
9	40	0.0021
10	45	0.0021
11	50	0.0021

The MSE for chirp signal in worst case obtain is 0.0105.

Case study-2: for chirp signal figure 5 shown below shows actual input audio signal and its 30db awgn noisy signal.

Figure 6 shown below shows analytical comparison of given input audio signal and filtered output signal. Comparative analytical plot which is been develop for different noise quantity in input and output observed, the comparison is been done as mean square Error between audio input and audio output.

Table 2: MSE observed in Audio for different Noise level

S. No.	SNR(dB)	Min. MSE
1	0	0.2879
2	5	0.1546
3	10	0.0813
4	15	0.0597
5	20	0.047
6	25	0.0441
7	30	0.0427
8	35	0.0429
9	40	0.0431
10	45	0.0432
11	50	0.0433

The MSE for Audio signal in worst case obtain is 0.2879.

Figure 8 shown below shows plot which is been develop for different noise quality in input and output MSE observed, the comparision is been done as MSE between audio input and audio output.

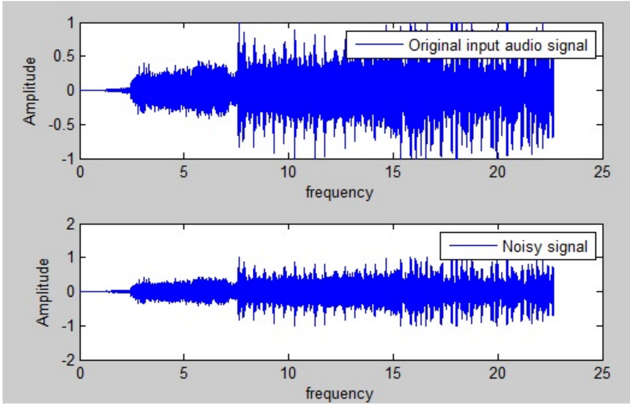

Figure 5: the Audio & noisy signal

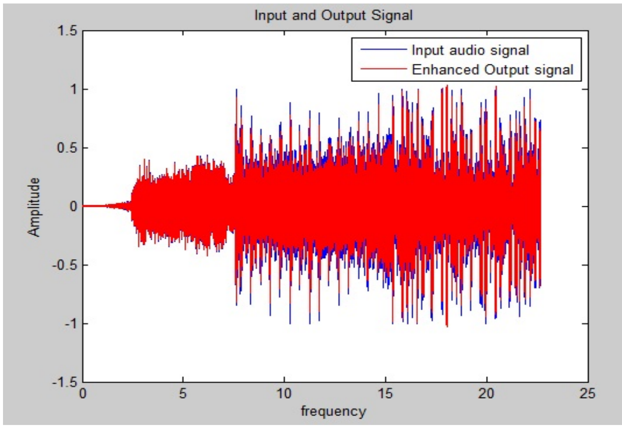

Figure 6: the blue is audio input signal and red filtered output

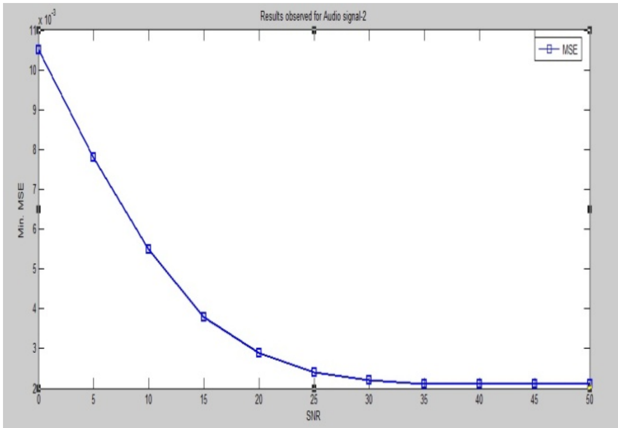

Figure 7: MSE for different noise value in Audio signal

Comparison of base work and DWT-REA: the comparison is been done as MSE between REA and proposed DWT-REA algorithm as shown in table 3 for an audio signal.

Table 3: MSE observed in REA and DWT-REA

SN	*Algorithm*	*SNR(dB)*	*MSE*
1	Alexander Schasse	*35*	*0.0442*
2	*DWT-REA*	*35*	*0.0021*

V. CONCLUSION

De-nosing Filters are very important in signal processing the aim of the proposed work was to develop a filter which have high accuracy in filtering the receive signal through any medium, this is been achieved by using the DWT filtering along with Parks-McClellan filter. The observed results for two different type of signals first was a chirp signal and second was audio signal shows that signal which are been transmitted and after propagating through various awgn noisy channels if filtered by proposed filter at receiving end, it have very less MSE which can be avoidable. In future the work can be done for various other than just chirp/voice or audio signals, the work can be implemented on DSP processor for emulation and in future the higher type Parks-McClellan new Wavelet can be use for more optimized filtering.

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