Highly Reconfigurable Trimmed Mean Filter for Multiband Noise Cancellation

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Abstract - *The active noise cancellation headphone use sound to cancel sound. The existing ANC system use Multi Band Filter (MBF) which provide variable and multi band filtering by tuning cutoff frequency. The existing system has an adaptive structure with single order and multi order coefficient generation. The Existing system use 41 logic elements and has the total power dissipation of .154W. This paper proposed the designing of trimmed mean filter for the cancellation of noise in the in-ear headphones. The trimmed mean filter is designed to cancel the multiband noise without affecting the actual input audio. The proposed design is expected to reduce the size and power consumption than the existing system.*

Key Words: Active noise cancellation, multi band filter, trimmed mean filter, in-ear headphone, trimming.

1.INTRODUCTION

Active noise cancellation is done by introducing anti noise wave through secondary sources which cancel out the noise wave. Destructive interference notion is applied. In [8] the basic adoptive algorithm for ANC is developed and analyzed based on single- channel broad band feed forward control. This algorithm is then modified for the narrow band feed forward and adaptive feedback control. These single channel ANC algorithms were then expanded to multiple channel cases for controlling the noise field in an enclosure or a large dimension duct. Various adaptive algorithms such as the lattice, frequency domain, sub band and RLS algorithms were also modified for ANC applications. [6] and [7] focuses on an active noise cancellation system for a home window using a transparent acoustic transducer. In a traditional active noise cancellation system, direct microphone measurements are used for reference and error signals. In the case of window application, both external and internal sound would be picked up by such microphones. This leads to adverse effects on the performance of the active noise cancellation system and also to distortion of the internal sound.

To address this problem, [6] and [7] proposed a wave separation technique to separate the internal and external component of sound. The wave separation algorithm is based on the use of two microphones and an algorithm that separates components based on their direction of travel. The theory for active cancelling of noise is simple, but the realization of an efficient ANC system is challenging due to several physical constraints. The existing ANC systems often use high-speed digital signal processors to cancel out disturbing noise, which results in high power consumption for a commercial ANC headphone. The contribution of the paper [3] can be classified into:

1) Proper filter length selection;

2) Low-power storage mechanism for convolution operation; and

3) High-throughput pipelining architecture.

H-S Vu and K-H Chen [3] have developed an area-/power-efficient ANC circuit by using the TSMC 90-nm CMOS technology for in-ear headphone applications. The performance of ANC systems is determined by the magnitude response of the secondary path, which can be affected by the placement of the error microphone. In [9], the optimum location of the error microphone is subjected to the flat magnitude response of the secondary path, is found by experiments, and explained by the spectral filtering of pinna. The emphasis of S.M Kuo, S Mitra and W-S Gan [9] is on the design and experiment of the AFANC headphone in real time, with the goal to achieve higher noise cancellation as compared to high-end commercial ANC headphones. Multi-band filters are used to separate the frequency components of a signal and pass certain selected frequency ranges while filtering out other frequency ranges. It uses Z transform and inverse functions to achieve multiband filtering.

In this paper a trimmed mean filter is designed to overcome all the shortcomings of the previous designs and to have better sound quality. Trimmed Median Filter (TMF) [5] is a decision based asymmetric filter. TMF is a two stage filter. First it detects the noisy pixels and then restores them. TMF considers all saturated pixels (0 or 255) as noisy pixels. If a pixel value lies within the dynamic range then it is considered a noise free pixel. Noise free pixels are left unchanged in the restoration stage. For each noisy pixel, the neighboring pixels within the 3X3 window are analyzed in the restoration stage. If all the pixels of the selected 3X3 window are deemed to be noisy, then the center pixel is replaced by the mean of the 3X3 window in the restored image. If the selected 3X3 window contains both the noisy pixels and noise free pixels, then the center pixel is replaced by the median of the noise free pixels in the 3X3 window. In this paper the trimmed mean filter is designed to cancel the noise in audio signal.

The rest of this paper is organized as follow. Section 2 presents the working of the multiband filter in cancelling the noise. The proposed trimmed mean filter architecture for cancelling noise in audio signal is presented in section 3 and the section 4 presents the conclusion.

2. MULTI BAND FILTER

A multi band filter for noise cancellation in VLSI circuit for in-ear headphones using frequency tuning filter architecture is designed. It combines all digital filters in single interpolation using coefficient transform technique. The first order coefficients are required for variable low pass and high pass responses and the second order coefficients are required for variable band pass and band stop frequencies. A look up table is used in the circuit to store the coefficients of the low pass filter. To obtain the other filters like high pass, band pass and band stop the coefficients in the look up table is altered. Multiplier-Accumulator (MAC) unit has been used in the circuit which is used to multiply the input data and the coefficients from the look up table and then the value is added to the accumulator. Z transform has been used to convert the low pass values in the look up table to band pass coefficients.

The block diagram of the multi band filter is shown in the Fig - 1. The block consists of a look up table. It has four MAC units, two inverse functions, an output selection unit and a filter. The look up table has the coefficients of the low pass filter. The coefficients of the high pass filter, band pass filter and band stop filter can be obtained by tuning the coefficients in the look up table.

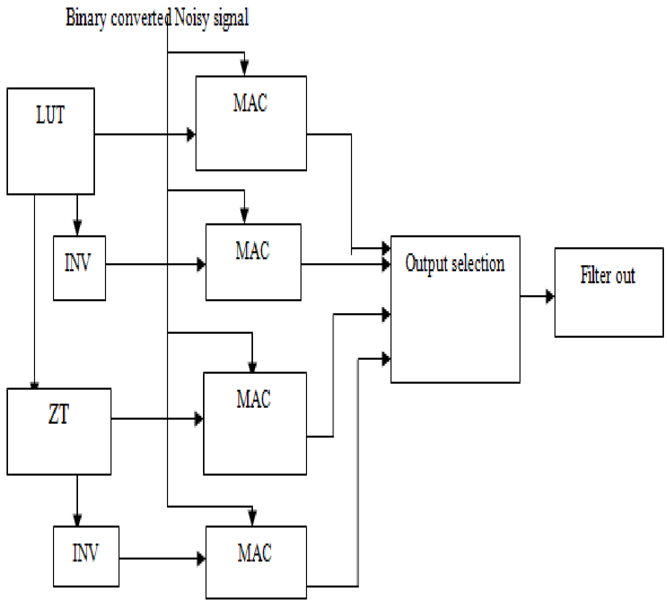

Fig - 1: Block Diagram of the Multi Band Filter

The input is the noise added sound signal. This signal is converted into binary text before giving it to the circuit. The binary converted noisy sound signal is then given to the MAC unit as one of its input. The outputs of the four MAC units are given to an output selection unit. The desired band response can be obtained from that unit. For the low pass filter, the coefficients are already in the look up table. The coefficients of the low pass filter and the input noisy signal is processed in the MAC to get the filtered low pass response. For the high pass filter, the coefficients of the look up table are given to inverse function which will give the inverse of the low pass coefficients. As the inverse of low pass is high pass this process will give the high pass filter coefficients. This is then given to the MAC unit along with the input noisy signal to get the filtered high pass response. For the band pass filter, the z transform is used. The low pass coefficient from the look up table is given to the z transform. The z transform of the low pass filter will give the band pass filter. The obtained band pass coefficient is then given to the MAC unit along with the input noisy signal. The result is the filtered band pass response. For band stop filter, the coefficients of the band pass have to be inversed. The coefficients of the band pass are given to the inverse function which will give the band stop coefficients. Then they are given to the MAC unit along with the input noisy signal to get the filtered band stop response. The MAC unit is given two inputs. One is the binary converted noisy sound signal and other is the coefficients of the variable filter. If the MAC filter is given the coefficients directly from the look up table then it will pass the low frequencies and filter the other frequencies. In such cases it acts as a low pass filter. If the MAC filter is given the inverse of the coefficient from the look up table it then act as high pass filter. It will only pass the high frequencies and filter all other frequencies. If the MAC filter is given the z transform of the coefficients from the look up table then it acts as band pass filter. It will pass only the band pass frequencies and filter all other frequencies. If the MAC filter is given the inverse of the z transform then it will pass only the band stop frequencies and filter all other frequencies. In this situation it will act as a band stop filter. The MAC unit contains a finite impulse filter. This filter is responsible to filter out the noise signal from the input signal. A good property of FIR filter for active noise cancellation in headphones is that they are less sensitive to the accuracy of constants. The outputs of the four MAC units are given to the output selection unit. From that unit desired response for the filters can be obtained. The output of the circuit is the noiseless signal. The given input signal without noise is obtained as the output signal.

3. TRIMMED MEAN FILTER

The proposed model use the trimmed mean filter instead of the multiband filter for the noise cancellation in the headphones. The objective of the proposed system is to cancel noise in the headphone by using trimmed mean filter and to reduce area and power consumption. Trimming means removing some portion of the largest and smallest values before taking the mean. By this the extreme local values does not affect the output. The trimming portion varies between 0% and 100% equivalent to the mean. Digital signals are quantized to a finite number of levels having an upper and a lower bound. If the signal is contaminated by noise values then the signals exceed these bounds. Then the signal is trimmed, reducing the range of values of noise may take in regions where the signal takes extreme values.

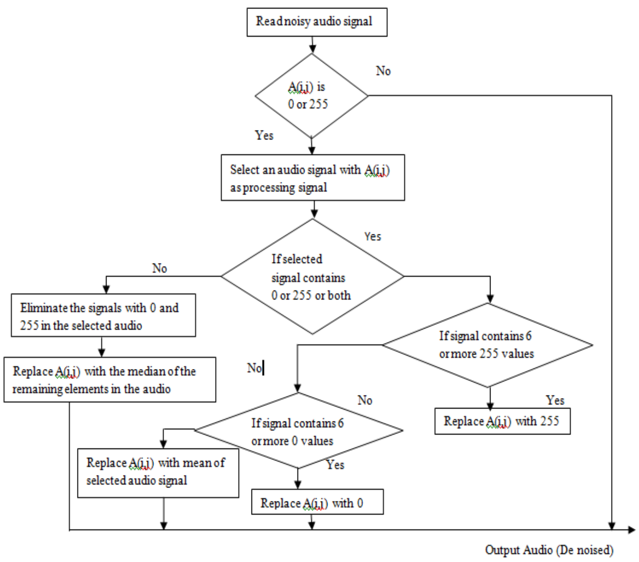

Fig - 2: Flow chart of the proposed system

The processing sound is checked whether it is noisy or noisy free. For example let us consider the audio signal is noisy only if it is 0 or 255dB, all the remaining sounds are left unaltered. Fig - 2 shows the flow diagram of the proposed system process. The steps are explained below

Step 1: If selected audio signal is other than 0 or 255dB, it is uncorrupted signal and it is left unaltered. Else process the signal.

Step 2: Let A(i,j) be the audio signal being processed.

Step 3: If selected audio signal consists of all 0 or 255dB or both then

a. If signal has 6 or more 255's then replace A(i,j) with 255.

b. Else if signal has 6 or more 0's then replace A(i,j) with 0.

c. Else replace A(i,j) with mean of elements in the audio signal.

Step 4: If selected audio signal consists of other values as well, then eliminate 0's and 255's and find the median of remaining elements. Replace A(i,j) with this median value.

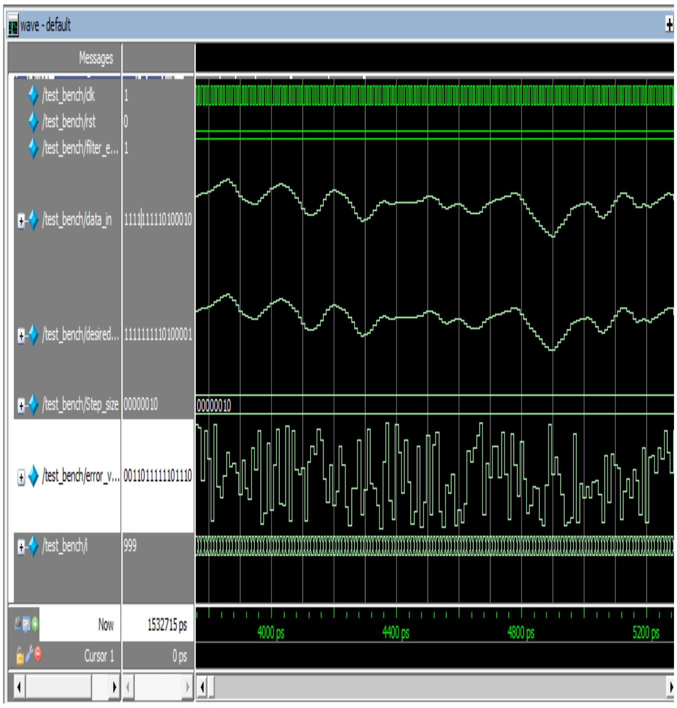

Fig – 3: Output

The process is simulates in ModelSim-Altera 6.4a and the simulated output is shown in Fig - 3. The input audio signal is represented as data in. The noise cancelled is labeled as the error. The random noise is cancelled and the desired output is obtained. In the output waveform it is shown that the input and the output signal are symmetrical with very little difference. This shows that the proposed model is good in reproducing the given input audio signal.

4. CONCLUSION

The existing multiband filter has a total power consumption of .154W and 41 logic elements. Here the coefficients have to be stored and for other bands process like inversion and transformation has to be done. This takes space and power. To overcome this trimmed mean filter is proposed. It saves processing time, space and power consumed. The proposed model is simulated in the ModelSim- Altera 6.4a and the power and area are calculated using Altera Quartus software.

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