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Implementation of Adaptive Filtering Algorithm for Speech Signal on FPGA

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Abstract: This project gives the study of the principles of Adaptive Noise Cancellation (ANC) and its Applications. Adaptive noise Cancellation is an alternative technique of estimating signals corrupted by additive noise or interference. In signal processing methods of removing noise, levels of noise rejection are not attainable without prior knowledge about speech signal and noise. But in this method of noise cancellation with no a priori knowledge of signal or noise, noise rejection can be achieved satisfactorily. FPGA implementation of adaptive filtering algorithm is studied in this project work. Two adaptive Filtering Algorithms are implemented LMS and wiener. LMS filter is designed in VHDL. Here wiener filter is implemented in adaptive manner to accommodate the varying nature of speech signal. The adaptive wiener filter is implemented in time domain rather than in frequency domain. This adaptive wiener filer is uses two method for speech enhancement TSNR and HRNR. The basic principle of wiener filter is to obtain the estimate of speech signal corrupted by noise. The noise reduction process applies spectral gain to short time spectrum value of noisy speech signal. This gain is expressed as function of priori SNR which is estimated using decision-directed approach. TSNR is used to eliminate the drawback of decision directed approach and retains its advantage. But in noise reduction process some harmonics which a part of original speech signals are suppressed. For that HRNR method is used to recover these harmonics. The resulting artificial signal is produced in order to refine the *a priori* SNR used to compute a spectral gain able to preserve the speech harmonics.

Keywords: Adaptive noise cancellation, LMS Filter Weiner filter, adaptive wiener filter, TSNR, HRNR

I. INTRODUCTION

Speech is a very basic way for humans to convey information, it has a bandwidth of only 4 kHz; it can convey information with the emotion of a human voice. Certain properties of the speech signal are, it is a one-dimensional signal, with time as its independent variable, it is random in nature, it is non-stationary, and the frequency spectrum is not constant in time. Although human beings have an audible frequency range of 20 Hz to 20 kHz, the human speech has significant frequency components only up to 4 kHz.

The most common problem in speech processing is the effect of interference noise in the signals. This noise masks the speech signal, reduces its intelligibility and also in noisy environment speech communication is greatly affected by the presence of background acoustic noise.

The presence of background noise in speech significantly reduces the intelligibility of speech. Noise reduction algorithms are used to suppress such background noise and improve the perceptual quality and intelligibility of speech. Removing various types of noise is difficult due to the random nature of the noise and the inherent complexities of the speech. Noise reduction techniques usually have a trade-off between the amount of noise removal and speech distortions introduced due to processing of the speech signal. Several techniques have been proposed for this purpose in the area of speech enhancement, like spectral subtraction approach, wiener filter and kalman filter. The performances of these techniques depend on the quality and intelligibility of the processed speech signal. The improvement in the speech signal to noise ratio is the target of most techniques.

II. SPECTRAL SUBTRACTION

The spectral subtraction methods are the most widely used due to the simplicity of implementation and also due to low computational load. It reduces the effect of background noise based on the STSA estimation technique. The basic power spectral subtraction technique is popular due to its simple underlying concept and its effectiveness in canceling noise in speech signal.

The basic principle of the spectral subtraction method is to subtract the magnitude spectrum of noise from that of the noisy speech.

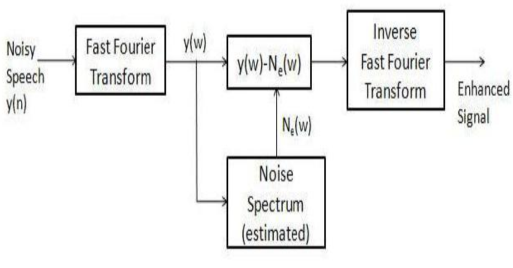

Fig.1 Block Diagram of spectral subtraction.

The noise is assumed to be uncorrelated and additive to the speech signal. An estimate of the noise signal is measured during silence or non-speech activity in the signal. The phase of the noisy speech is kept unchanged, since it is assumed that the phase distortion is not perceived by human ear. However the subtraction type algorithms have a serious drawback in that the enhanced speech is accompanied by unpleasant musical noise artifact, which is characterized by tones with random frequencies.

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A. Drawbacks of Spectral Subtraction Residual noise:

It is obvious that the effectiveness of the noise removal process is dependent on obtaining an accurate spectral estimate of the noise signal. The better the noise estimate, the lesser the residual noise content in the modified spectrum. However, since the noise spectrum cannot be directly obtained, we are forced to use an average estimate of the noise. Hence there are some significant variations between the estimated noise spectrum and the actual noise content present in the instantaneous speech spectrum.

The subtraction of these quantities results in the presence of isolated residual noise levels of large variance. This residual spectral content manifest themselves in the reconstructed time signal as varying tonal sounds resulting in a musical disturbance of an unnatural quality. This musical noise can be even more disturbing and annoying to the listener than the distortions due to the original noise content.

B. Roughening Of Speech Due To the Noisy Phase:

The phase of the noise-corrupted signal is not enhanced before being combined with the modified spectrum to regenerate the enhanced time signal. This is due to the fact that the presence of noise in the phase information does not contribute immensely to the degradation of the speech quality. This is especially true at high SNRs (>5dB). However, at low SNRs (<0dB), the noisy phase can lead to a perceivable roughness in the speech signal contributing to the reduction speech quality.

III. ADAPTIVE NOISE CANCELLATION

Speech is the most primary human communication. For that reason there exists a big trend to increase and improve telecommunications. However, the background noise is an important handicap. If it is joined with other distortions, it can seriously damage the service quality.

In all applications that require at least one microphone, the signal of interest is usually contaminated by background noise and reverberation. As a result, the microphone signal has to be 'cleaned' with digital signal processing tools before it is played out, transmitted or stored.

So, it is important to cancel the noise which may combine the signal in order to obtain a good quality signal, this may be achieved using adaptive Noise Cancellation which improves the Signal-to-Noise Ratio at the received noisy signal.

Adaptive noise cancellation is an alternative technique of estimating signals corrupted by additive noise or interference. It has advantage that, with no priori knowledge of signal or noise, levels of noise rejection are obtainable that would be difficult to achieve by other signal processing methods of noise removing.

The principle of adaptive noise cancellation is to obtain an estimate of the noise signal and subtract it from the corrupted signal.

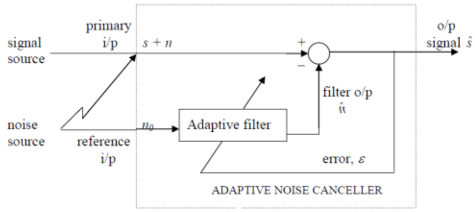

Fig.2 Adaptive Noise Canceller

As shown in the figure, an Adaptive Noise Canceller (ANC) has two inputs – primary and reference. The primary input receives a signal s from the signal source that is corrupted by the presence of noise n uncorrelated with the signal. The reference input receives a noise n0 uncorrelated with the signal but correlated in some way with the noise n.

The noise no passes through a filter to produce an output nˆ that is a close estimate of primary input noise. This noise estimate is subtracted from the corrupted signal to produce an estimate of the signal at sˆ, the ANC system output

A. Adaptive Filters

The adaptive noise cancellation technique uses adaptive filters for signal processing. The adaptive filter constitutes an important part in statistical signal processing. Whenever there is a requirement to process signals that result from operation in an environment of unknown statistics, the use of an adaptive filter offers an attractive solution to the problem as it usually provides a significant improvement in performance over the use of a fixed filter designed by conventional methods. Furthermore, the use of adaptive filters provides new signal-processing capabilities that would not be possible otherwise.

We thus find that adaptive filters are successfully applied in such diverse fields as communications, control, radar, sonar, seismology, and biomedical engineering.

An adaptive filter is very generally defined as a filter whose characteristics can be modified to achieve some end or objective and is usually assumed to accomplish this modification or adaptation automatically.

B. Adaptive Filtering Algorithm

There are many algorithms used to adjust the coefficients of the digital filter in order to match the desired response as well as possible. This includes the following.

LMS Algorithm:

The simplicity of the Least Mean Square (LMS) algorithm and ease of implementation makes it the best choice for many real-time systems.

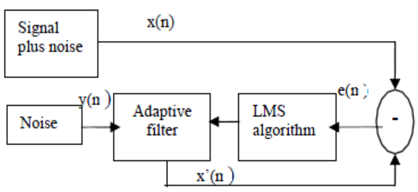

Fig 3: Adaptive LMS Filter

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The implementation steps for this algorithm can be stated as;

(1) Define the desired response and set each coefficient weight to zero

$$w(n) = 0, n = 1,2,3,\ldots,N$$

For each sampling instant (n) carry out the following steps;

(2) Move all samples in the input array one position to the right, now load the current data sample (n) into the first position in the array. Calculate the output of the adaptive filter by multiplying each element in the array of filter coefficients by the corresponding element in the input array and all the results are summed to give the output corresponding to that data that was earlier loaded into the input array, such that the output y(n) is;

$$y(n) = \sum_{n}^{N}{}_{0}^{1} w(n)x(n)$$

(3) Before the filter coefficients can be updated, the error must be calculated, simply find the difference between the desired response, d(n), and the output of the adaptive filter, y(n).

$$e(n) = y(n) - d(n)$$

(4) To update the filter coefficients multiply the error by the learning rate parameter, μ, and then multiply the results by the filter input and add this result to the values of the previous filter coefficients.

$$\overline{w}(n+1) = \overline{w}(n) + \mu.e(n).\overline{x}(n)$$

The resources required to implement the LMS algorithm for a transversal adaptive FIR filter of L coefficients in real the computations given are those required to process one sample.

Wiener Filtering Algorithm:

Wiener filter theory provides a convenient method of mathematically analysing statistical noise cancelling problems.

The Wiener filter is a popular technique that has been used in many signal enhancement methods. The basic principle of the Wiener filter is to obtain estimate of speech signal from that corrupted by additive noise. This estimate is obtained by minimizing the Mean Square Error (MSE) between the desired signal s(n) and the estimated signal ˆs(n). It is based on a statistical approach.

The Wiener filter weights noisy signal spectrum according to SNR at different frequencies

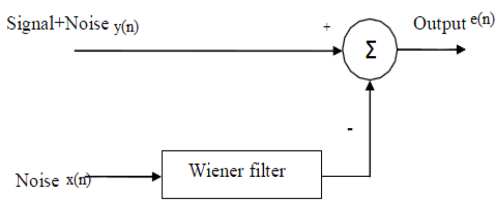

Fig.4 Basic of Wiener Filter

Typical filters are designed for a desired frequency response. However, the design of wiener filter takes a different approach. One is assumed to have knowledge of the spectral properties of the original signal and the noise, and one seeks the linear time invariant filter whose output would come as close to the original signal as possible.

(1) For this transfer function of wiener filter is used in frequency domain which is expressed as follows:

$$H(\omega) = \frac{Ps(\omega)}{Ps(\omega)+Pd(\omega)}$$

Where, $Ps\ (\omega)$ and $Pd\ (\omega)$ are power spectral densities of clean and noisy speech signals respectively.

(2) In wiener filter, the the speech signal and noise is assumed uncorrelated and stationary, and the SNR is given by:

$$SNR = \frac{Ps(\omega)}{Pd(\omega)}$$

(3) Using this definition of SNR, the transfer function of Wiener filter can be given

$$H(\omega) = \left[1 + \frac{1}{SNR}\right]^{-1}$$

From the above definition of transfer function, it can be interpreted that the Wiener filter has fixed frequency response at all frequencies and also needs an estimation of the power spectral density of clean signal and noise prior to filtering.

C. Adaptive Wiener Filter

This section presents an adaptive implementation of the Wiener filter which benefits from the varying local statistics of the speech signal. The designed adaptive wiener filter depends on the adaptation of the filter transfer function from sample to sample based on the speech signal statistics (mean & variance).

A block diagram of the approach is as shown in figure below.

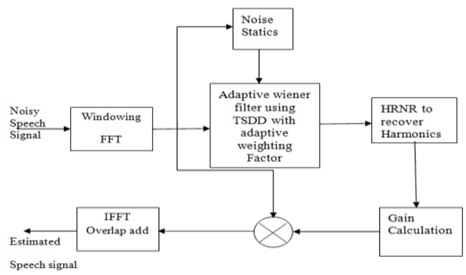

Fig.5 Block Diagram of Adaptive Wiener Filter

As we seen above main aim of wiener filter is to find out the signal estimate. This signal estimate is calculated by multiplying spectral gain with noisy speech spectrum. This spectral gain depends upon the *priori* SNR. This priori SNR follows the shape of posterior SNR but with delay of one frame. In practical implementations of speech enhancement systems, the power spectrum density of the speech and the noise are unknown as only the noisy speech is available. Then, both the *instantaneous* SNR and the *a priori* SNR have to be estimated.

This *instantaneous* SNR and the *a priori* SNR can be estimated using decision directed approach. The behaviour of the estimator of the *a priori* SNR controlled by the parameter α .The multiplicative gain function is obtained by multiplying functions of priori SNR with instantaneous

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SNR. The speech signal spectrum is calculated by multiplying this multiplicative gain with noisy speech spectrum.
The Multiplicative gain used here is of wiener transfer function.

D. Implementation of Adaptive Filtering Algorithm for speech Signal on FPGA

In this project implementation, two types of adaptive filtering algorithms are implemented one is Least Mean Square algorithm (LMS) which is implemented in VHDL language and second is adaptive wiener filtering algorithm (AWF) is implemented in MATLAB.

To evaluate the performance of the adaptive filtering algorithm and extensive simulations have been performed. The implementation process comprises of the following steps:

(1) To implement and evaluate the performance of adaptive filtering algorithm , noisy speech signal is passed through the adaptive Filter to remove the noise from signal.
(2) Least Mean Squares (LMS), one of the widely used algorithms in many signals processing environment, is implemented for adaption of the filter coefficients. The cancellation system is implemented in VHDL .The simulation of VHDL design of adaptive filter is performed and analysed on the basis of Signal to Noise ratio (SNR) and Mean Square Error (MSE).
(3) The adaptive Wiener filter is implemented in time domain rather than in frequency domain. This is done to accommodate the random nature of speech signal. The cancellation system is implemented in MATLAB .The simulation of MATLAB design of adaptive filter is performed and analysed on the basis of Signal to Noise ratio (SNR). Two methods TSNR and HRNR are implemented to enhance the speech signal

E. Experimentation Result

Result of LMS Filter

In this section, result of the LMS Filter is given in the form of Architecture, RTL schematic, net list and simulation results test bench waveform.

- Architecture

```
architecture arch of LMS is signal x1,x2,w1,w2: std_logic_vector(15 downto 0):="0000000000000000";
begin process--(clk)
variable w11,w22: std_logic_vector(15 downto 0):="0000000000000000";
variable y,e: std_logic_vector(31 downto 0);  --(1,19,12)
variable k: std_logic_vector(12 downto 0):="0000000000000";
variable t1,t2: std_logic_vector(63 downto 0):=(others=>'0');
```

- RTL Schematic

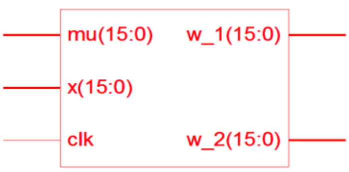

Fig.6 RTL Schematic for LMS filter

- Net List

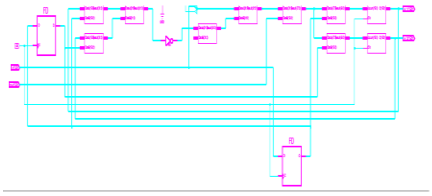

Fig.7 Net list

- Simulation Results

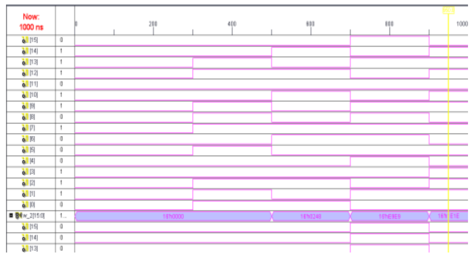

Fig.8 Simulation of LMS Filter

Result of Adaptive Wiener filter

In this section, result of the Adaptive Wiener Filter is given in the form of Noisy speech signal waveform, Spectrogram of clean speech signal.

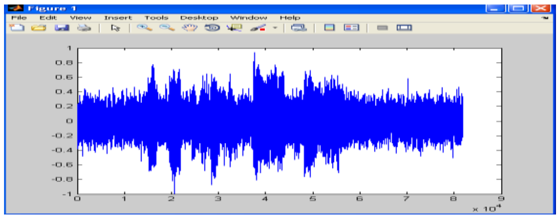

Fig.9 Waveform of Noisy speech signal

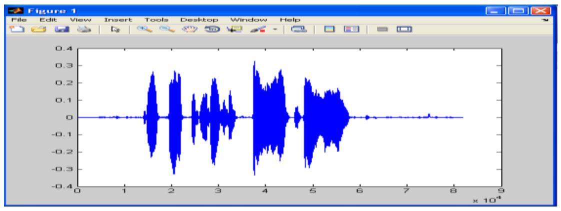

Fig.10 Clean speech signal Using AWF method TSNR

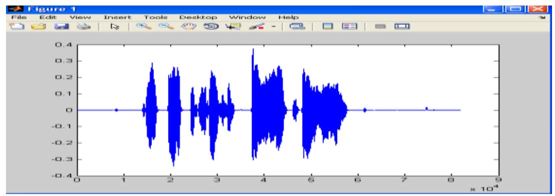

Fig.11 Clean speech signal Using AWF method HRNR

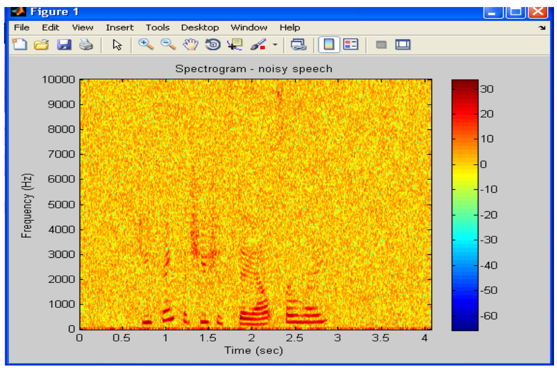

Fig.12 Spectrogram of Noisy speech signal

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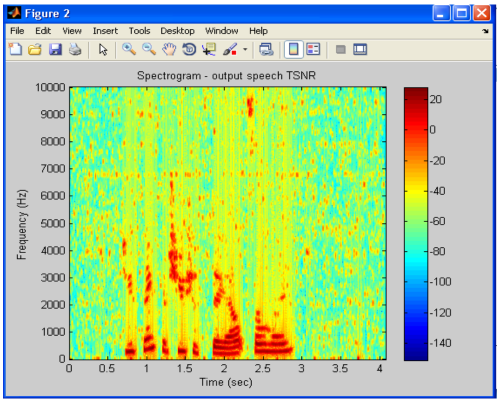

Fig. 13 Spectrogram of Output speech signal using TSNR

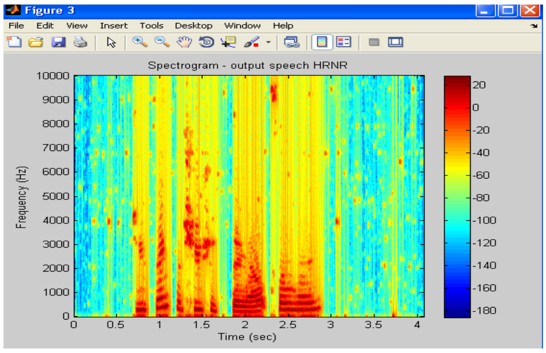

Fig. 14 Spectrogram of Output speech signal using HRNR

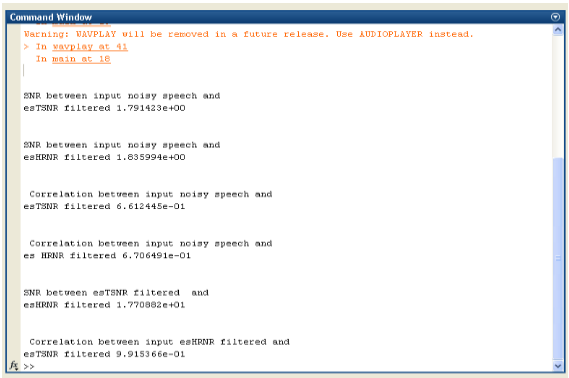

Fig.15 SNR Values between input noisy speech signal and clean speech signal

IV. CONCLUSION

Because of the high level of noise (especially due to traffic), now-a-days, it is difficult to record a clean speech signal. Depending upon the SNR, the complete signal can be fade-in by noise. Therefore, Adaptive filtering Algorithm of FPGA is implemented in this project work. Filtering is done in adaptive manner so as to contribute varying nature of speech signal. LMS, a popular Adaptive Filtering Algorithm and Adaptive wiener filter which is capable of estimating signal even in situations of very high noise or low SNRs is Implemented on FPGA . FPGA has become the best choice for the design of signal processing system due to their greater flexibility and higher bandwidth, resulting from their parallel architecture. The performance of this algorithm is evaluated in terms of various analytical parameters. From the results obtained, it is concluded that the algorithm maintains the intelligibility of the original speech signal.

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