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IMPLEMENTATION OF INTERFERENCE CANCELLATION BY ADAPTIVE FILTERS

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ABSTRACT
This paper investigates on the development and implementation of adaptive noise cancellation (ANC) algorithm meant for mitigating the high machinery noise in factory plants ,which makes the speech signal unintelligible.This opens up the need for an adaptive filter that cancels this interference of noise.An adaptive filter is computational device that attempts to model the relationship between two signals in real time in an iterative manner.An adaptive filter self adjusts the filter coefficients according to an adaptive algoritm.A comparative study of Gradient based adaptive Infinite Impulse Response(IIR) algorithm and its modified version is performed using MATLAB simulator interms of converging speed.From the simulation result the best IIR algorithm is used for implementation in Performance Optimized with Enhanced RISC PC (Power PC) 7448.

KEYWORDS: Adaptive noise cancellation (ANC),IIR filter,Power PC 7448,Adaptive Line Enhancer(ALE),Mean square error (MSE)

INTRODUCTION
In the field of signal processing, there is a significant need for a special class of digital filters known as adaptive filters. Adaptive filters are used commonly in many different configurations, and for certain applications these filters have a great advantage over the standard digital filters. They can change their filter coefficients according to preset rules. An adaptive filter is a digital filter that can adjust its coefficients to give the best match to a given desired signal[1]. When an adaptive filter operates in a changeable environment the filter coefficients can adapt in response to changes in the applied input signals. Based on recursive algorithms adaptive filters update their coefficients and train them to reach the optimum solution. Adaptive filters are mainly used in applications like system identification, linear prediction, inverse system identification, noise cancellation etc[6]. Adaptive signal processing is one of the most important classes of algorithms for modern communication systems. The concept of adaptive noise cancellation has become widespread in the areas of communications and signal processing. The noise-cancellation problem involves two received signals, commonly referred to as the primary and the reference. The goal is to cancel (that is, subtract out) the portions of the two signals that are mutually correlated [2].

There are different adaptive algorithms (eg. Least Mean Square(LMS), Normalized Least Mean Square (NLMS) etc.) that can be used in time domain. Some widely used methods with Adaptive FIR filter have been explained by S. Haykins [6] & B.Widrow [7]. FIR filters are more stable than IIR filters.Adaptive filters are attractive: many fewer coefficients may be needed to achieve the desired performance in some applications[3]. However, it is more difficult to develop stable IIR algorithms, they can converge slowly. Adaptive IIR algorithms are used in some applications (such as low frequency noise cancellation) where the need for IIR-type responses is great. In some cases, the exact algorithm used by a company is a tightly guarded trade secret.However a gradient-based Adaptive IIR algorithm, with some additional features that enable it to adapt more quickly is explained by Don R.Hush [2]. A modified version of the above algorithm gives even faster

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convergence. So the modified algorithm can be used for implementation.

ADAPTIVE SIGNAL PROCESSING

The term Adaptive can be understood by considering a system which is trying to adjust itself so as to respond to some phenomenon that is taking place in its surroundings.This is what adaptation means. Moreover there is a need to have a set of steps or certain procedure called algorithm by which this process of adaptation is carried out[4] .

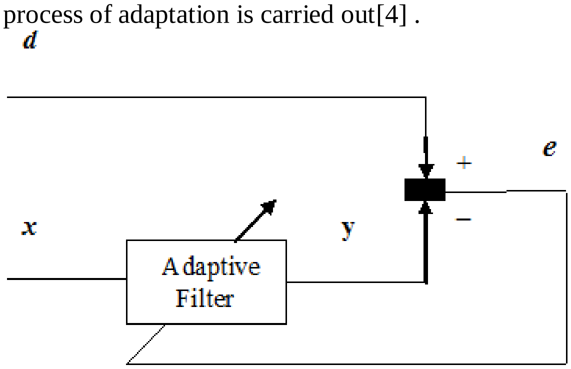

Figure:**1** ***Basic adaptive filter structure***

The adaptive filter structure shown in the figure 1in which the filter's output y is compared with a desired signal d to produce an error signal e, which is fed back to the adaptive filter. The error signal is given as input to the adaptive algorithm, which adjusts the variable filter to satisfy some predetermined rules. In stochastic frame work (based on Wiener filter theory), the optimum coefficients of a linear filter are obtained by minimization of its mean-square error (MSE). All the adaptive algorithms take the output error of the filter, correlate that with the samples of filter input in some way, and use the result in a recursive equation to adjust the filter coefficients iteratively

ADAPTIVE ALGORITHMS

There are many adaptive algorithms available. Two main algorithms are LMS and NLMS algorithms[7]..

Least Mean Square Algorithm (LMS)

The LMS algorithm is one of the most widely used adaptive filtering algorithms due to its simplicity and robustness to signal statistics. LMS algorithm uses the instantaneous value of the square of the error signal as an estimate of the MSE.

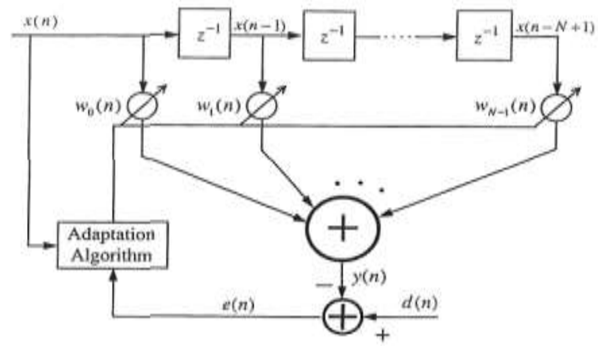

***Figure**:**2** An N tap transversal adaptive filter[8]*

Figure 2 depicts an N-tap transversal adaptive filter. The filter input, x (n), desired output, d (n) and the filter output,

The equations are derived inline with[8]

$$y(n) = \sum_{i=0}^{N-1} w_i(n)x(n-i) \qquad (1)$$

are assumed to be real-valued sequences. The tap weights w_0 (n), w_1 (n) ... ,w_{N-1}(n) are selected so that the difference (error),

$$e(n)=d(n)-y(n) \qquad (2)$$

is minimized in some sense.

It may be noted that the filter tap weights are explicitly indicated to be functions of the time index 'n'. The LMS algorithm changes (adapts) the filter tap weights so that *e(n)* is minimized in the mean-square sense, thus the name least mean square. It is a sequential algorithm which can be used to adapt the tap weights of a filter by continuous observation of its input, *x(n)*, and desired output, *d(n).*

The recursive weight update equation is given by

$$\mathbf{w}(n+1) = \mathbf{w}(n) - \mu \nabla e^2(n) \qquad (3)$$

Where $\mathbf{w}(n) = [w_0(n)\ w1(n)\ ....w_{N-1}(n)]^T$, μ is the algorithm step-size parameter and ∇ is the gradient operator .

$$\nabla e^2(n) = -2e(n)\mathbf{x}(n) \qquad (4)$$

Where $\mathbf{x}(n) = [x(n)\ x(n-1)\ ...\ x(n-N+1)]^T$.

Substituting the result in equation (3)

$$\mathbf{w}(n+1) = \mathbf{w}(n)+2\mu e(n)\mathbf{x}(n). \qquad (5)$$

This is referred to as the LMS recursion. It suggests a simple procedure for recursive adaptation of the filter coefficients after the arrival of every new input sample, *x(n)*, and its corresponding desired output sample, *d(n).*

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Normalized Least Mean Square Algorithm (NLMS)

The normalized (NLMS) algorithm may be viewed as a special implementation of the LMS algorithm which takes into account the variation in the signal level at the filter input and selects a normalized step size parameter which results in a stable as well as fast converging adaptation algorithm. NLMS algorithm provides optimized performance keeping the stability. The only difference of NLMS algorithm with that of LMS algorithm is that the step size adjusts based on the input signal, thereby converging faster than LMS algorithm.

Consider the LMS recursion

$$\mathbf{w}(n+1) = \mathbf{w}(n) + 2\mu\, e(n)\mathbf{x}(n). \qquad (6)$$

But the step size μ changes based on input $\mathbf{x}(n)$ such that

$$\mu = \frac{1}{2\mathbf{x}(n)\mathbf{x}^T(n)} \qquad (7)$$

Substitute the value in equation (6)

$$\mathbf{w}(n+1) = \mathbf{w}(n) + \frac{1}{\mathbf{x}^T(n)\mathbf{x}(n)}\, e(n)\mathbf{x}(n). \qquad (8)$$

This is the NLMS recursion. When this is combined with the filtering equation (1) and the error estimation equation (2) NLMS algorithm is obtained.

ADAPTIVE NOISE CANCELLATION

The ANC method uses a "primary" input containing the corrupted signal and a "reference" input containing noise correlated in some unknown way with the primary noise. The reference input is adaptively filtered and subtracted from the primary input to obtain the signal estimate.

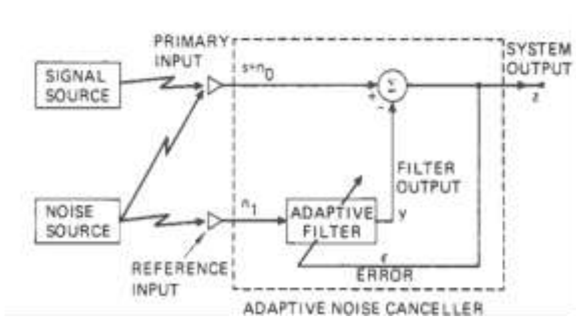

Figure:3 ***Adaptive noise cancelling concept***

A signal s is transmitted over a channel to a sensor that also receives a noise n_0 uncorrelated with the signal. The combined signal and noise $s+n_0$ form the primary input to the canceller. A second sensor receives a noise n_1 uncorrelated with the signal but correlated in some unknown way with the noise n_0. This sensor provides the reference input to the canceller. The noise n_1 is filtered to produce an output y that is as close a replica as possible of n_0. This output is subtracted from the primary input $s+n_0$ to produce the system output z=. $s+n_0-y$ Hence the output of the system contains the signal alone. Adaptive noise cancellation has been used for several applications like cancelling the Maternal ECG in Fetal Electrocardiography,Cancelling noise in speech signals,Cancelling periodic interference without an external reference source.

ADAPTIVE NOISE CANCELLATION USING FIR FILTER

In adaptive noise cancellation an adaptive filter is used to determine the relationship between the noise reference signal *x(n)* and the component of this noise that is contained in the measured signal(desired) *d(n)*. After subtracting out this component adaptively the error signal *e(n)* gives the signal of interest. Hence adaptive noise cancellation is a method of estimating signals corrupted by additive noise or interference.

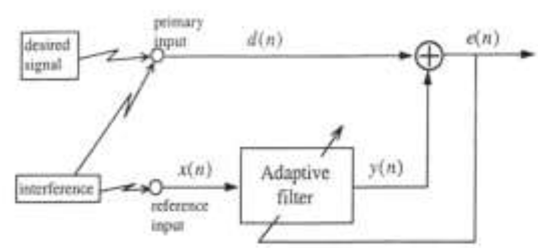

Figure:4 ***Adaptive filter as a noise canceller[9]***

Simulation Results

For simulation sinusoidal signal is taken as the pure signal. Then add noise with this pure signal. Adaptive noise cancellation is performed using above algorithms and compare the performance of algorithms. Figure 4 shows the pure sinusoidal signal, figure 5 shows the noise which is added to the pure signal to make the noise corrupted signal. The sinusoidal signal is distorted by the addition of noise. The noise corrupted signal is shown in figure 6.

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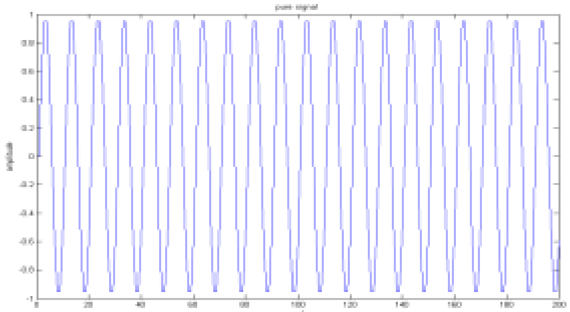

Figure:5 ***Pure sinusoidal signal***

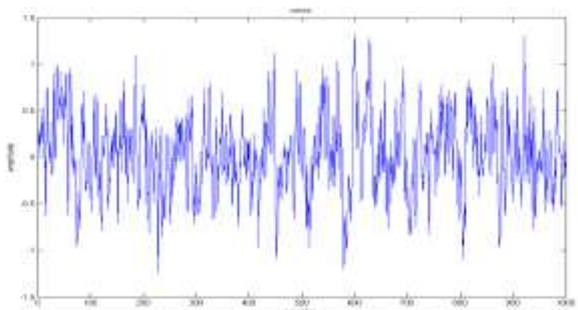

Figure:5 ***Noise Signal***

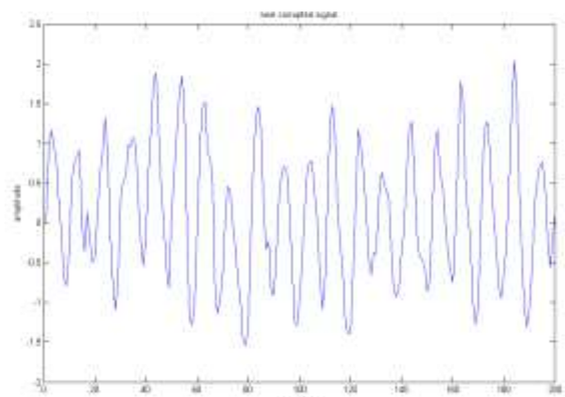

Figure:6 ***Noise corrupted signal***

The noise filtered signals (nfs) obtained using the two algorithms shown in the figures below.

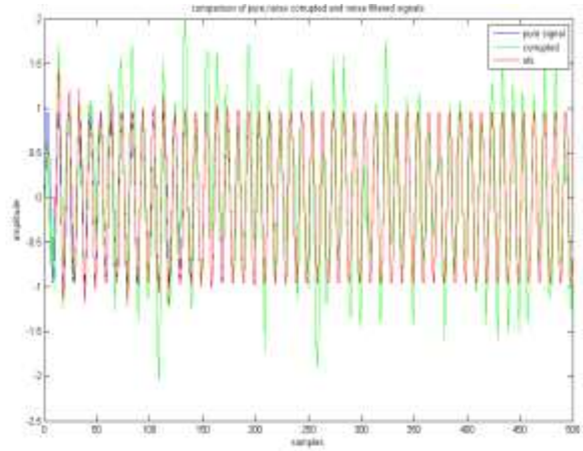

Figure:7 ***Noise filtered signal using LMS algorithm***

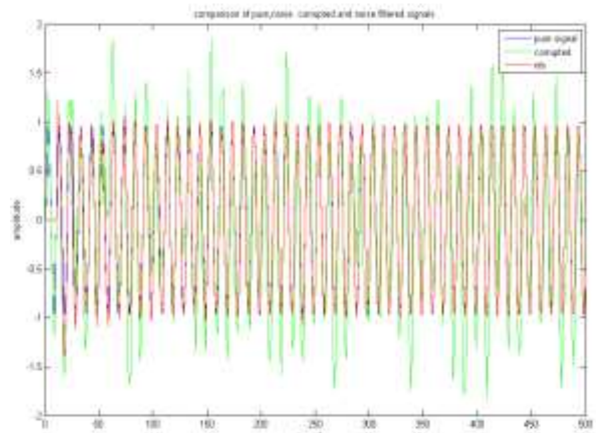

Figure:8 ***Noise filtered signal using NLMS algorithm***

The figure 7 and 8 show noise filtered signals using LMS and NLMS algorithms respectively. From the simulation result it is clear that using the adaptive algorithms the signal can be recovered. The NLMS algorithm converges faster than the conventional LMS algorithm

ADAPTIVE LINE ENHANCER USING IIR STRUCTURE

The Adaptive Line Enhancer enhances the sinusoidal component of the reference input so that output has high signal to noise ratio SNR. From the ALE we also obtain an estimate of the sinusoidal frequency. This ALE can be utilizes as noise canceller. Several forms of the ALE are available. The most popular is the FIR ALE. It has the advantage of being inherently stable and easy to adapt. However, it often requires a

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large number of filter weights to provide adequate enhancement of narrow-band signals. In an effort to reduce the number of weights, several forms of adaptive recursive filters (IIR) can be used.

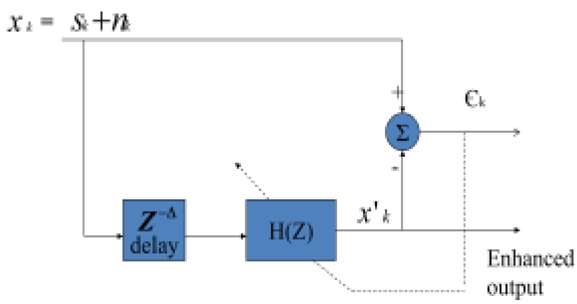

Figure:9 ***Adaptive Line Enhancer[2]***

The purpose of Δ is to decorrelate the input noise and its delayed version present at the filter input. This causes the adaptive process to respond only to the sinusoid. As such, H (z) forms a band pass filter around the sinusoid, applying at the same time a phase shift to compensate for the delay so that the sinusoidal component of x_k is removed at the summer. At the output of the adaptive filter x'_k, the enhanced sinusoid is obtained. To simplify the analysis assume n_k is white noise so that $\Delta = 1$ is sufficient for decorrelation.

GRADIENT BASED ADAPTIVE ALGORITHM FOR IIR ALE

The technique used here is a gradient-based algorithm with some additional features that enable it to adapt more quickly. The filter output is given by

$$x'_k = \{\frac{1-r^2}{1+r^2}\} w_k x_{k-1} - (1-r^2)x_{k-2} + w_k x'_{k-1} - r^2 x'_{k-2} \qquad (9)$$

Where $0 << r < 1$
$-2r < w < 2r$, r is the radius of the circle just inside the unit circle

Also from figure 9, the error output is defined to be

$$\varepsilon_k = x_k - x'_k \qquad (10)$$

The coefficient update for the adaptive algorithm can be expressed as

$$w_{k+1} = w_k + \rho \varepsilon_k \alpha_k \qquad (11)$$

Where ρ is a parameter which controls the rate of convergence, ε_k is the error, and α_k is the partial derivative

$$\alpha_k = \frac{\partial x'_k}{\partial w_k} \qquad (12)$$

From (9) and the definition of (11) , recursive relationship for α_k

$$\alpha_k = w_k \alpha_{k-1} - r^2 \alpha_{k-2} + (\frac{1-r^2}{1+r^2}) x_{k-1} + x'_{k-1} \qquad (13)$$

The coefficient update is then defined by (10),(11) and (13).

The normalization factor is incorporated by reexpressing the coefficient update in (11) as

$$w_{k+1} = w_k + \mu_1 \frac{\varepsilon_k \alpha_k}{\psi_k} \qquad (14)$$

Where ε_k and α_k are as defined above and

$$\psi_k = v \psi_{k-1} + (1-v)\alpha_k^2 \qquad (15)$$

The "forgetting factor "v is in the range 0<<v<1

The performance of this algorithm can be enhanced even further by using an approximate partial derivative in place of α_k in (13). The modified derivative, denoted by α_{mk},is obtained from (13) by suppressing the recursive terms, i.e.,

$$\alpha_{mk} = (\frac{1-r^2}{1+r^2}) x_{k-1} + x'_{k-1} \qquad (16)$$

The resulting algorithm is then defined by the same set of equations as before, with α_k replaced by α_{mk}. The effect of this modification is to produce a modified gradient estimate which has a higher probability of taking on the correct sign (direction) than the true gradient. This in turn leads to a faster convergence rate.

Simulation Results

After performing the adaptive process, at the output of the filter we get the enhanced version of the sinusoid. At the output of the summer, (error signal) sinusoidal component of input is removed. The simulation result shows the Power Spectral Density (PSD) of pure signal, noise corrupted signal, comparisons of PSD of error signals and enhanced signals using both the algorithms. Mean Square Error (MSE) of two algorithms is also shown.

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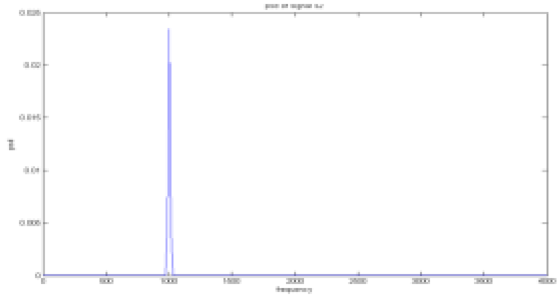

Figure:10 PSD of pure signal

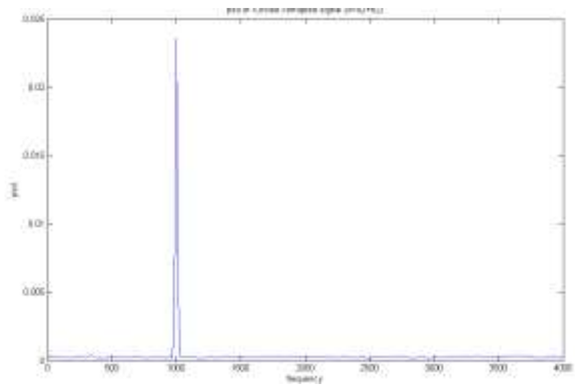

Figure:11 PSD of noise corrupted signal

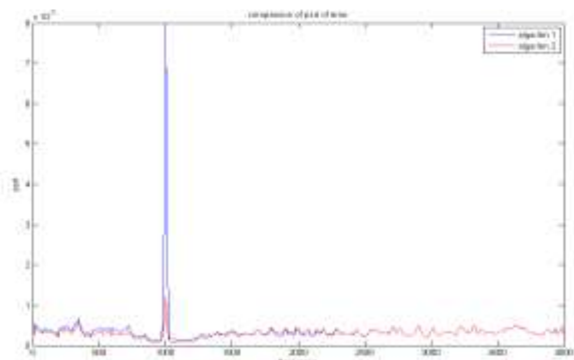

Figure:12 Comparison of PSDof error

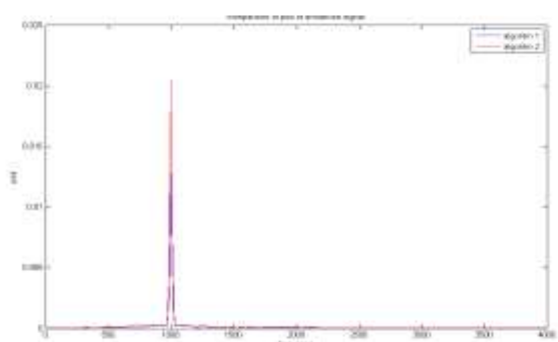

Figure:13 Comparison of PSD of enhanced signal

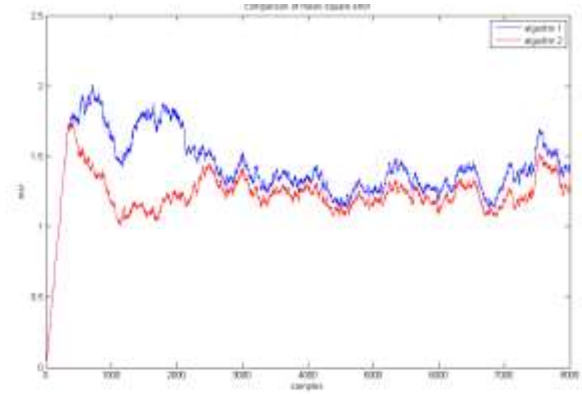

Figure:14 Comparison of Mean square error

From the PSD of error signal it is clear that a very large reduction of power occur at the output of the summer. The modified algorithm provides more reduction in power. The figure 15 shows the reduction in power for different Signal to Noise Ratio (SNR) with both algorithms. The figure 16 shows the convergence time of both the algorithms.

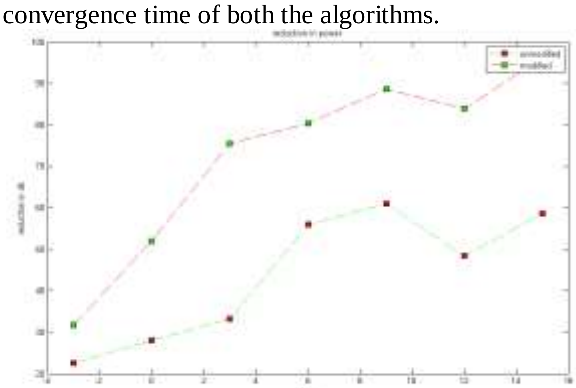

Figure:15 Reduction of Power in db

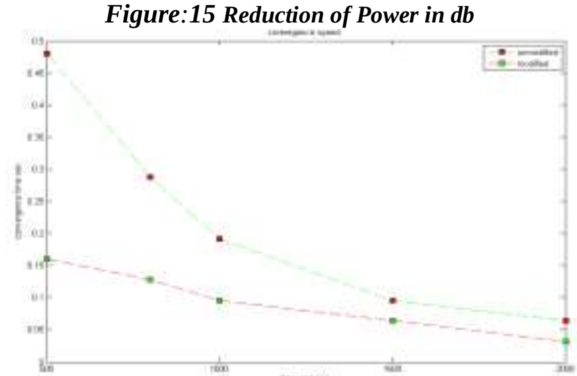

Figure:16 Comparison of convergence time

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ADAPTIVE IIR NOISE CANCELLATION

ALE can be utilized as noise canceller with experimental data. ALE is used as the basic structure for noise cancellation.

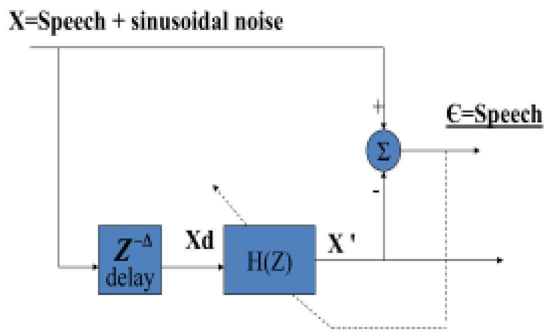

Figure:17 ***IIR noise canceller***

Here the input signal X contains the speech signal and sinusoidal noise. The purpose of Δ is to decorrelate the input speech and its delayed version present at the filter input. This causes the adaptive process to respond only to the sinusoid As such, H (z) forms a band pass filter around the sinusoid, applying at the same time a phase shift to compensate for the delay so that the sinusoidal component of **X** is removed at the summer. By taking the autocorrelation of speech signal one can find out suitable value for Δ.So at the output of the summer (error signal) contains the speech alone. i.e., the sinusoidal noise is adaptively cancelled out. Let the signal to be interfered with noise such that the SNR is controllable and verify the performance of IIR structure in highly noise environment. The system is provided with the permission to input a particular SNR (user's choice), a noise with corresponding amplitude is generated and added to signal. This noise corrupted signal will be processed

Table 1. SNR and Power Levels

SNR in dB	Power Level between signal and noise
0 dB	Signal power = noise power
-3 dB	Signal power =(1/2) noise power
-5 dB	Signal power << noise power

Simulation Results

Verify the performance of algorithm in highly noise environment . Produce the simulation result with SNR = -5 db, the speech power is too low compared to sinusoidal noise.

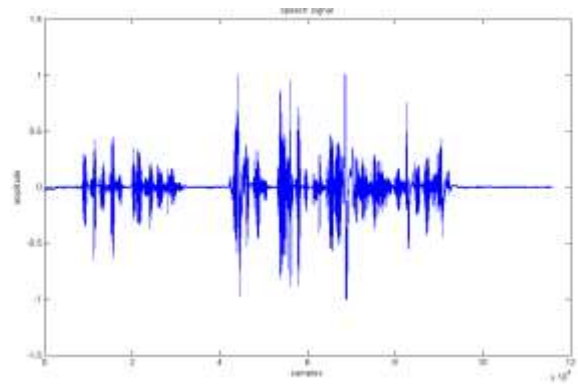

***Figure**:18 Speech signal*

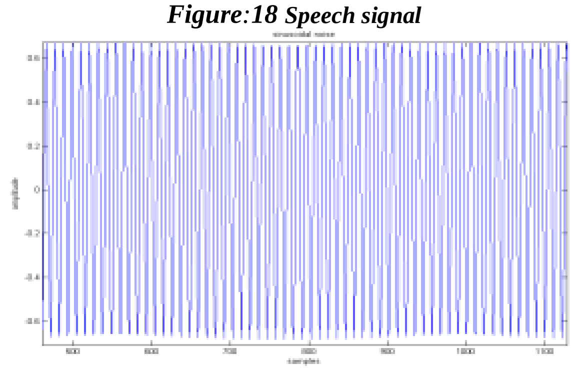

***Figure**:19 Sinusoidal noise*

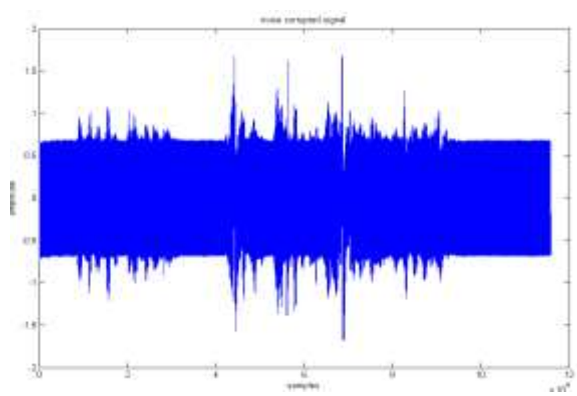

***Figure**:20 Noise corrupted signal*

The figure 18 shows the pure speech and the figure 19 shows the sinusoidal noise. Figure 20 shows the noise corrupted signal. As in ALE here also two adaptive algorithms for noise cancellation were performed, a gradient based algorithm and a modified version of the same. The figure 21 shows the comparison among the pure speech and noise filtered signals obtained from two algorithms. From the simulated result it is clear that even if the signal is interfered by a noise whose power is very high compared to signal power, adaptive IIR filter is able

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to pick out the noise. The blue colored signal is obtained from normal algorithm. The red colored signal is produced by modified algorithm. From the figure below it is clear that the modified algorithm is converging faster than the normal one. Hence modified version of the gradient based algorithmis used for implementation on PowerPC.

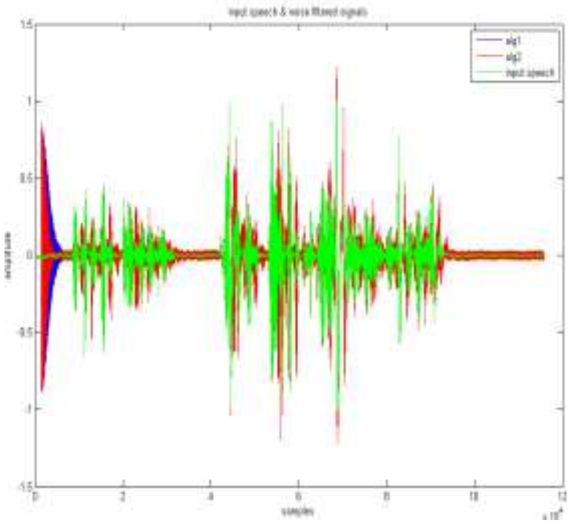


Figure:21 ***Input speech and noise filtered signals***

IMPLEMENTATION

Interference cancellation using adaptive IIR algorithm is implemented in PowerPC 7448 SBC(Single Board Computer).

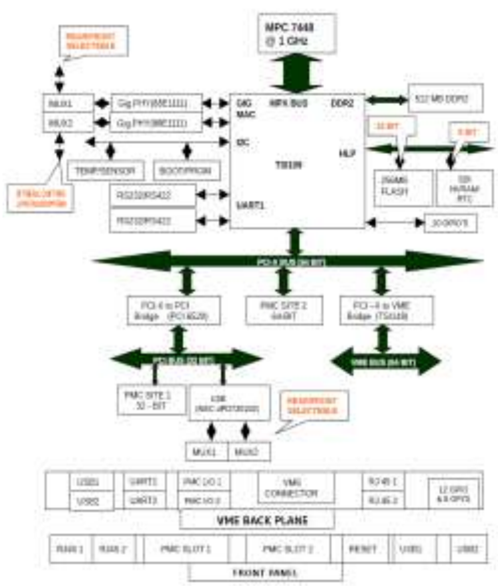
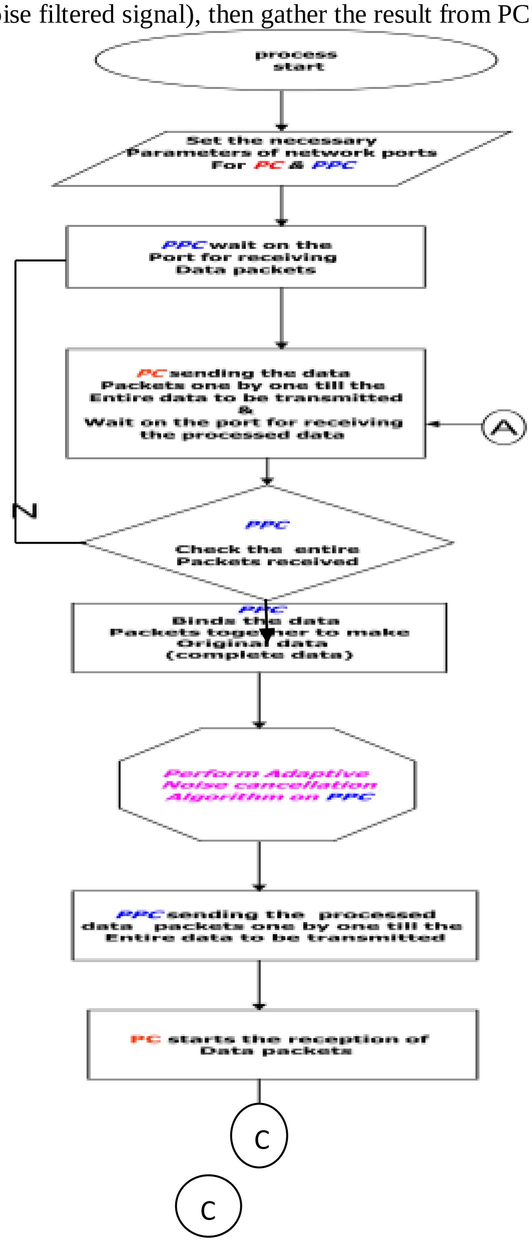


Figure:22 ***PowerPC 7448 SBC block diagram***

Implementation process flow

The noise corrupted signal (Speech signal in the presence of machinery noise)is captured by PC and the data would be transferred to PowerPC through network as packets. The PowerPC receives the data and perform the adaptive algorithm, and the result is send back to PC, the PC receives the processed data (noise filtered signal), then gather the result from PC



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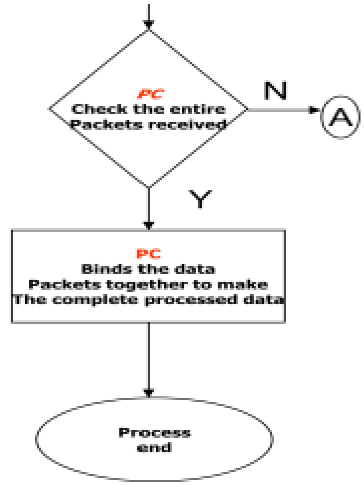

Figure:23 ***Implementation flow chart***

RESULTS AND DISCUSSION

The interference cancellation using IIR filter has been implemented successfully on PowerPC 7448 with Modified Gradient Based Adaptive algorithm and verified the output. Here the signal applied to the input is speech corrupted by machinery noise. After adaptive filtering process speech alone is produced at the output.

Since the sound card for the new PowerPC 7448 has not yet developed and integrated to the board, for the real time implementation sound cannot be directly captured in PowerPC .So the following procedure is used. The noise corrupted signal is captured by PC and the data would be transferred to PowerPC through network as packets. The PowerPC receives the data and perform the algorithm, and the result is again send to PC, the PC receives the processed data (noise filtered signal), then playback the result from PC. Figure 24 and Figure 25 shows the implemented result.

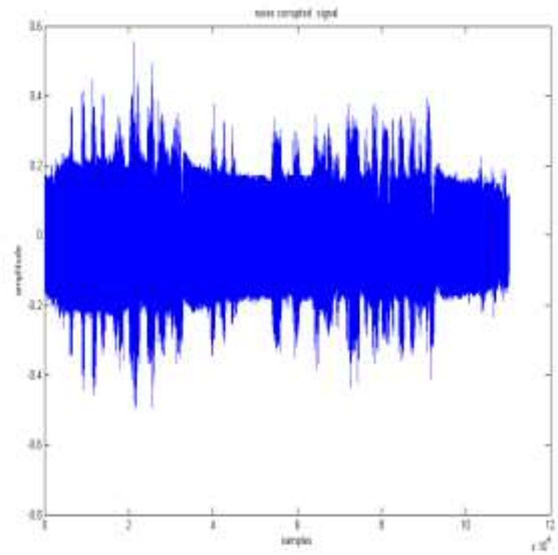

Figure:24 ***Noise corrupted Speech signal***

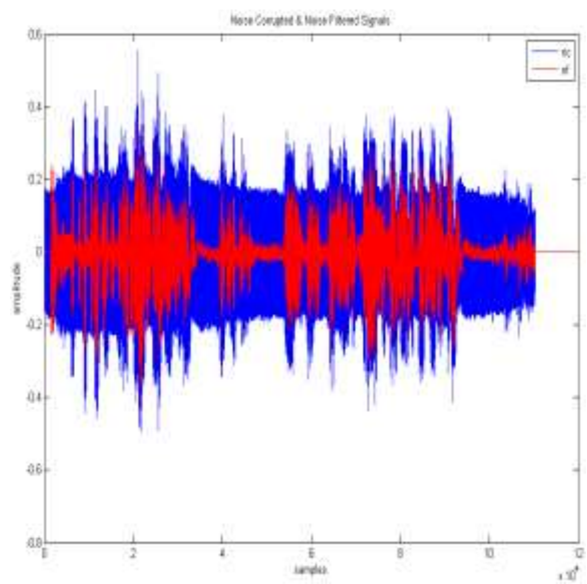

Figure:25 ***Noise corrupted and Noise filtered Speech signals***

Figure 25 shows the noise corrupted signal and noise filtered signal together. The Adaptive filter filtered the machinery noise and produced the speech alone at the output.

CONCLUSION

A detailed survey on ANC is carried out and observed that many fewer coefficients may be needed to achieve the desired performance for IIR filter when compared to FIR.From simulation under

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different noise environment, it is concluded that the modified gradient based algorithm is more suitable for Adaptive IIR noise cancellation. Even if the signal is interfered by a noise whose power is very high compared to signal power, adaptive IIR filter is able to pick out the noise. i.e., even in the highly noise environment adaptive IIR filter is powerful.

FUTURE SCOPE

Future work includes the implementation of the pipelined version of the filter for improved speed, and to optimize the programmable processing element at circuit level for efficient ASIC implementation of the reconfigurable fabric.

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