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# An Adaptive Noise Cancellation Using Wavelet Based Grazing Estimation of Signal Method: Special Reference to Speech Signal



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## Abstract:

This paper introduce the reducing the content of noise present in the received Speech signals for wireless communication medium by using Wavelet based Grazing Estimation of Signal (WGES) Method. Due to mixing of white Gaussian noise, the received signal is degraded. This proposed method is designed based on the superposition principle with eight possible cases. By conducting multiple possible cases of signal movement the noise signal is moved to opposite direction of original signal. This output is cascaded with wavelet transforms techniques with compare the available control algorithms output error signals. Compared to other available control algorithms the proposed method is Simple to implement, yields good performance and converges quickly. This proposed technique is implemented using Matlab software and DSP processor. This computer output simulation results confirm the effectiveness of our proposed algorithm.

## Keywords:

Active Noise Control, Filtered-X LMS algorithm, Grazing estimation signal, Minimum mean square, Pseudo Code, SNR, PSNR, Wavelet and Matlab 6.5.

## I.INTRODUCTION:

Active noise control has received much attention in the recent research for industrial applications. Based on the superposition principle the undesired noise can be reduced by adding another noise with same amplitude but opposite sign. Basically the noise cancellation is designed using adaptive techniques, which is working based on weight updating concepts for all iterations. The design and weight updating process increases the complexity of the control system implementation The proposed method using Grazing Estimation of Signal concept reduces the iteration time as compared with existing control algorithms. The proposed method is tested for various cases of change in noise direction with help of signal movement techniques. This system is designed using superposition principle.

## II.PROBLEM STATEMENT:

To achieve an immaculate commotion lessening for discourse signals transmitted through the remote medium utilizing versatile calculations. In such a correspondence, the whole the clamor (a loud and confused noise ) is included the channel. The commotion is profoundly irregular. Here there is no hotspot for getting a related clamor at the less than desirable end. Just the got sign can recount the tale of the commotion added to it subsequently some way or another, just on the off chance that it is conceivable concentrate the clamor from the got signal, through a few means, then the aforementioned versatile strategies to separate the sign to clamor proportion (SNR) of the got signal.

## III.MATHEMATICAL MODELING OF GRAZING ESTIMATION OF SIGNAL METHOD:

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The versatile strategies to lessen clamor are successful, when the reference commotion is exceedingly interrelated to the tainting commotion. In any case, attributable to the very irregular nature of the tainting clamor, it is hard to gauge it. Here, to produce a successful introduction commotion from the got signal itself, this can be then used to diminish the clamor substance of the coordinating got signal. This procedure depends on having initial two specimens of the first flag accurately. Beside assessment the third specimen utilizing the initial two examples. This is finished by finding the slant between the initial two examples and develops the same for the third specimen. This next assessed test is subtracted with the quality third specimens are utilized to gauge the fourth example similarly just like the third example found.

$$m = ES_{n-1} - ES_{n-2} \quad 1$$
$$ES_n = ES_{n-1} + m \quad 2$$
$$N'_n = X_n - ES_n \quad 3$$

Where, ES speaks to the evaluated signal, X speaks to the got sign, and N' speaks to the assessed commotion of the primary stage additionally an edge for the assessed clamor set. This edge depends on the imaginable level of commotion. The limit level can be close around 0.5 times the maximum outright esteem that the clamor can take. At whatever point the total estimation of the evaluated commotion level crosses this preset edge level, the assessed signal worth right then and there is reset.

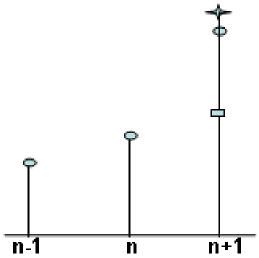
i.e. when $N'_n >$ threshold, then

$$ES_n = ES_n + N' \quad 4$$

This guarantees we don't simply continue moving in a solitary heading. At whatever point there is more than the normal deviation, we attempt to bring the evaluated signal quality in nearness of the sign value. Thusly we attempt to keep our assessed signal specimens in close nearness to the first flag over the span of estimation. We appear underneath all the conceivable cases.

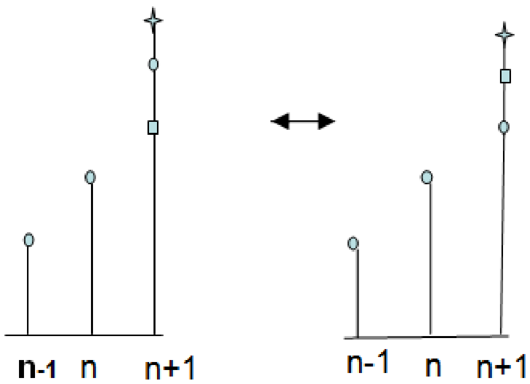
## IV.ROPOSED METHOD FOR POSSIBLE CASES:

### Case 1:




In the above case, see that the estimated falls below the actual value and the noise is positive. Thus in this case the estimated signal, the actual value and the noise has right then and there in the got signal [40]. This quality gives the assessed clamor test right then and there and we call it the as the evaluated commotion. Presently the second and the the same sign. The estimated signal, clean signal (actual signal) value and the noise all are in the same direction. This implies that at $(n+1)^{th}$ instant, there is some level of correlation between them.

### Case 2:




This case is similar to case 1 except that the noise here is of high magnitude. Due to this, the difference between the received value and the estimated value happens to be greater than the threshold. In this way the evaluated esteem must be balanced, as appeared in the figure above. For this situation we see that the spotless sign esteem, the evaluated signal worth and the got signal esteem all have in the same course. This as clarified on the off chance that 1, the evaluated sign

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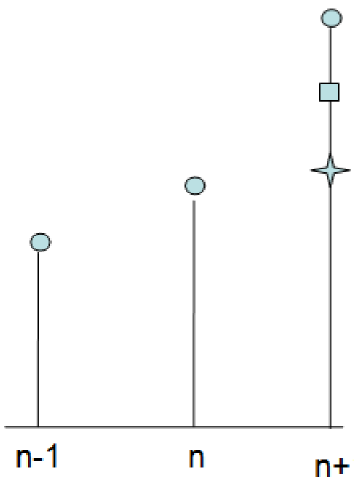
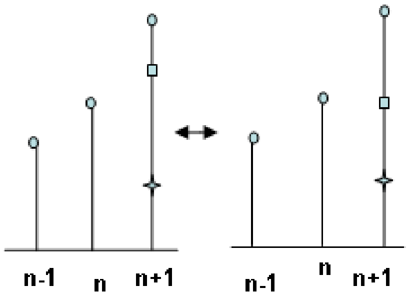
is corresponded to both the spotless sign and the commotion at the (n+1)th instant.

**Case 3:**



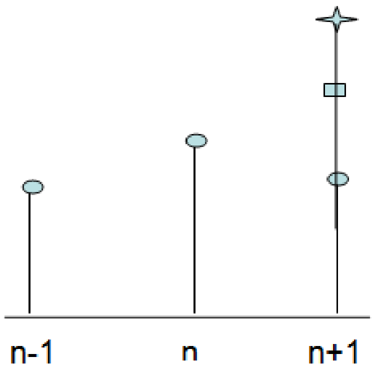

This case is like case 1, aside from that the clamor included at moment n+1 is negative. Again for this situation the evaluated signal and real flag have the same sign, and inverse to that to the commotion.

**Case 4:**



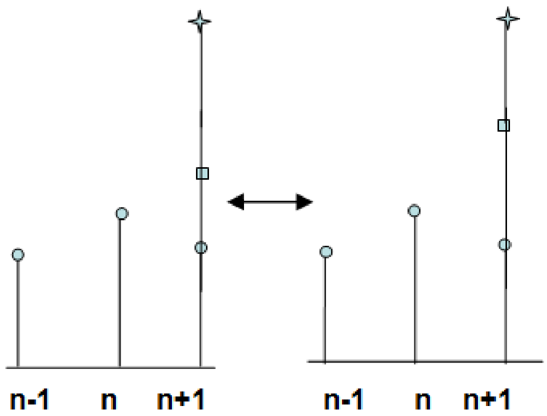

This case the noise is high negative, so that the difference between the estimated value and the received vale is more than the threshold. Hence we read just the estimated value.

**Case 5:**




Presently we will consider situations where there is a change of slant between the (n+1)th and nth specimens when contrasted with that of in the middle of nth and (n-1)th examples. For this situation the commotion is sure, as appeared in the figure above. As can be seen, the assessed signal and the genuine sign and the commotion for this situation will be in the bearing.

**Case 6:**




In this case the signal has high positive value, due to which the difference between the received value and the estimated value is more than the threshold. Thus as shown, we readjust the estimated value. In this case, the estimated, noise and the clean signal are in the same direction.

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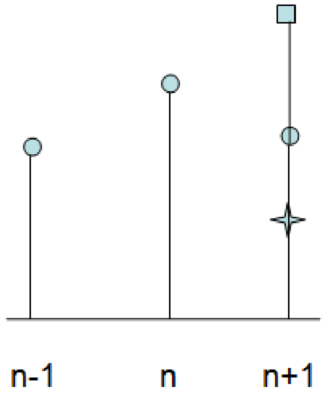
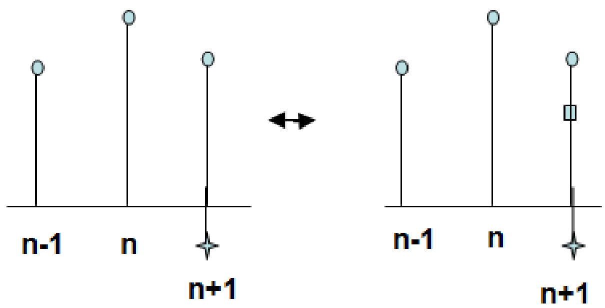
**Case7:**




For this situation the commotion getting included is negative. Along these lines as demonstrated, the evaluated and the clean are in the same course, and inverse to that of the clamor.

**Case 8:**




In this case the noise getting added is high negative which plunges the received value negative. If the noise Magnitude is large, than we move to adjust the estimated value as shown in the figure above, else we may not adjust the estimated value. In either case we see that the evaluated and the spotless sign are in the same course, while the commotion will have inverse bearing. Therefore we have seen all the conceivable cases for the qualities in the positive bearings. In every one of the cases, the evaluated signal and the perfect sign are in the same heading. Just at the, moments, when the sign worth crosses the time pivot, there are odds of the evaluated signal and the spotless sign to be inverse headings. These have less risks

## V.GENERALIZED PROCEDURE:

By using this procedure we can generate the well correlated signal to the actual signal indented to transmit. After generating the estimated signal and also called reference signal in the interference cancellation application of the adaptive signal processing. Since the clean signal and noise are uncorrelated, the output of this process is die noise signal itself. The generated noise and the actual noise depends on the level of the correlation between the estimated signal and the clean signal.

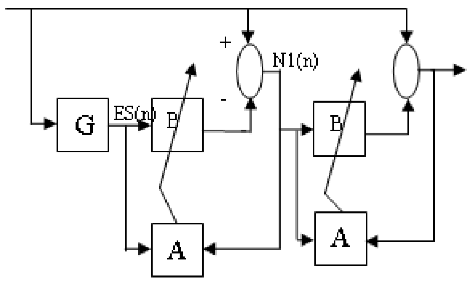



**Figure 1: Block diagram of Grazing estimation method**

Where the letters denote the following blocks A-Adaptive algorithm, B = Filter, D =delay, G=Grazing Estimation

The basic aim of generating a correlated noise to the noise corrupting the signal has been now achieved. It is now used this generated noise to cancel the noise In the received signal by using the interference cancellation application again. The received signal is passed through the grazing estimation block, the output of which is an estimated signal,

## VI.PROPOSED GES ALGORITHM:

Grazing {rec. Signal(1) signal{2)),

Load index(1), Indcx {2)=signal(1), signal(2) Noise1,,noise(2)= rec (1)-index(1},

rec(2)-index{2)

Len = length of the signal For n=3 to Len

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slope= index (n-2)-index{n-1)

index(n}=slope +index(n -2)
noise (n)= rec(n)-index(n)

if absolute value of noise(n)>threshold,
index(n) - index(n) +noise(n)/2
end

It is assumed that the sampled noisy speech signal y

$$Y_K = S_K + s_K . n_K$$

Where *k s* is the clean speech signal, *k n* represents an independent noise source with unit variance ($s_n^2$=1) and s $_K$ is the noise level. Wavelet denoising is an n on – Parametric estimation method that has been proposed in recent years for speech enhancement applications. The goal of wavelet denoisinig is to optimize the mean-squared error Subject to the side condition that with high probability, the estimation, $s^\wedge$ is at least as smooth as s .This constraint provides an optimal trade-off between the bias and variance of the estimate by keeping the two terms the same order of magnitude. The implementation of wavelet denoising is a three step procedure involving wavelet decomposition, nonlinear threshold and wavelet reconstructing although wavelet denoising, provides a theoretical framework to the estimation problem attributes specific lo speech must still be exploited to achieve good performance for the speech enhancement application. Here, the perceptual speech wavelet denoising system using adaptive

## VII.EMBEDDING GES WITH WAVELET TRANSFORM TECHNIQUE [WGES]:

The purpose of preprocessing is to initially lower the noise level *of yk* while minimizing the distortion in $s_k$ ( $y_k$ ) denotes the output of this preprocessing stage. For this the grazing signal estimation method is applied. The entire structure of implementing wavelet denoising is as given in figure 2 below

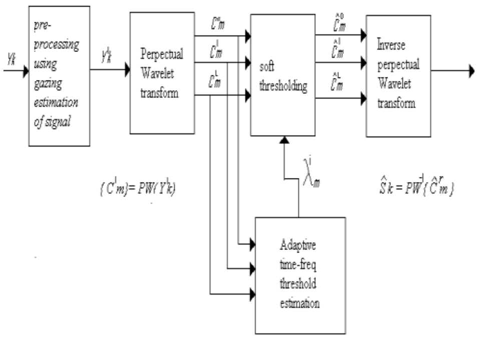


**Fig 2:Block diagram of wavelet de noising**

Quintile-based noise spectrum estimator to track the slowly varying non-stationary noise statistics. Simulation results show that the grazing estimation of the signal technique achieves modern's levels of noise suppression.

## VIII.SIMULATION RESULTS

**Table 1: Comparisons of Various Methods for Ding Sound Signal**

| Received Signal PSNR (db) | PSNR of recovered signal (db) | | |
|---|---|---|---|
| | Grazing Estimation | Wavelet de-noising | Grazing estimation + wavelet de-noising(Proposed Method) |
| 51.3451 | 64.6823 | 60.2440 | 70.4061 |
| 56.4513 | 65.0934 | 65.2465 | 71.0318 |
| 61.3707 | 65.8718 | 69.9194 | 72.6852 |
| 66.3772 | 68.1647 | 73.9194 | 74.6747 |
| 71.3298 | 71.858 | 76.9806 | 76.6095 |
| 76.3978 | 75.9917 | 78.5975 | 77.6586 |

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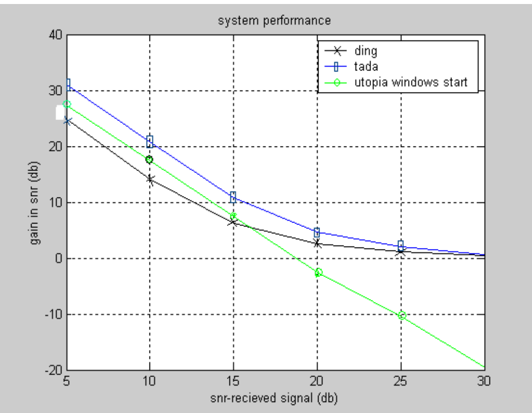
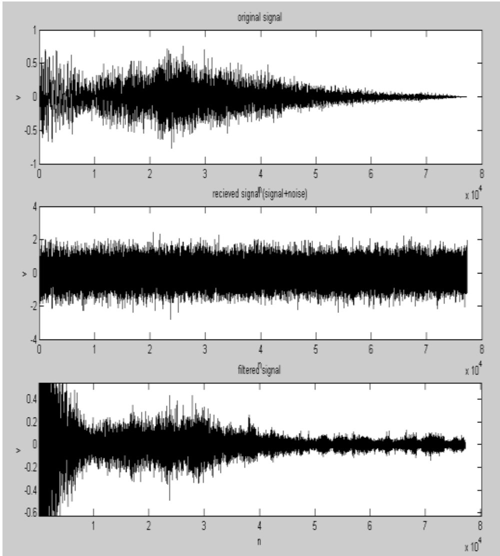
**Table 2: Comparisons of various methods for s1omwb speech signal**

| Received Signal PSNR (db) | PSNR of recovered | | |
|---|---|---|---|
| | Grazing Estimation | Wavelet de-noising | Grazing estimation + wavelet de-noising(Proposed Method) |
| 51.3451 | 64.6823 | 60.2440 | 70.4061 |
| 56.4513 | 65.0934 | 65.2465 | 71.0318 |
| 61.3707 | 65.8718 | 69.9109 | 72.6852 |
| 66.3772 | 68.1647 | 73.9194 | 74.6747 |
| 71.3298 | 71.8587 | 76.9806 | 76.6095 |
| 76.3978 | 75.9917 | 78.5975 | 77.6586 |

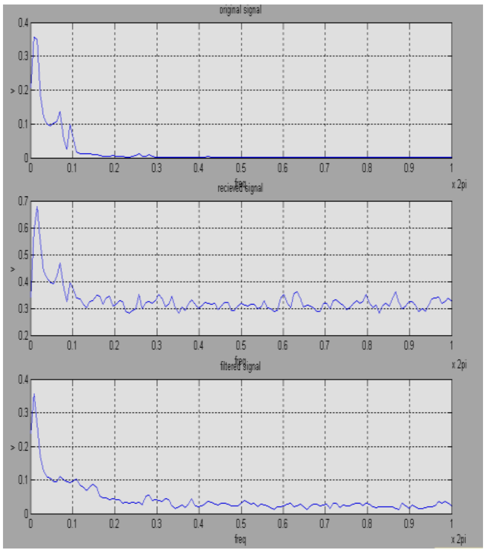



**Represents the gain in the PSNR**

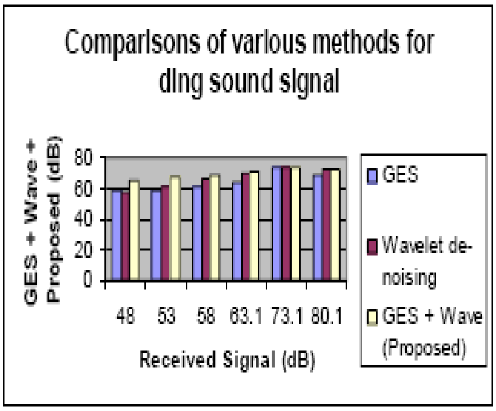
The table1 and 2 shows the analysis of proposed method using different speech signals. The proposed algorithm is more effective as compare to the existing algorithms.




**System performance in time domain**




**System performance in Frequency domain**




**Analysis of proposed method using Ding sound**

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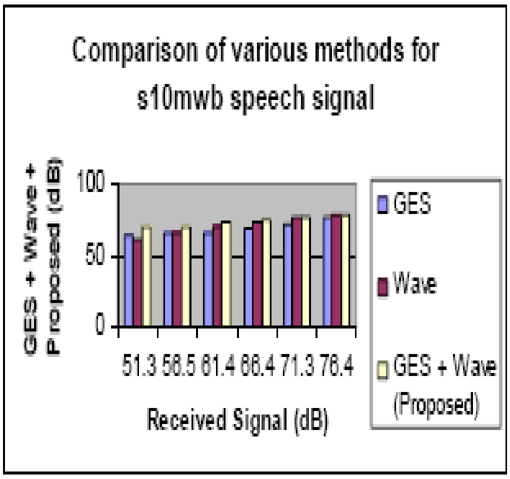


**Analysis of proposed method using S10mwb Speech signal**

## IX.CONCLUSION:

This proposed technique called grazing estimation of sign strategy has been acquainted with produce clamor at the less than desirable end, which is intelligible to the commotion adulterating the sign. In this way decrease of clamor level in got signal utilizing versatile sign handling method was conceivable. It could be seen that by and large PSNR upgrade of 15 – 20 db is conceivable, when the sign is profoundly drenched in clamor. It can likewise be seen that the addition in PSNR is high when the sign is all the more profoundly inundated in commotion. This gives it leverage of permitting the transmission of sign with low power

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