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Design of Adaptive Noise Cancellation for Speech Signals Using GES Method

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Abstract: This study introduced a new method for reducing the content of noise present in the received signal for wireless communication medium using Grazing Estimation of Signal method (GES). The received signal is corrupted due to mixing of white Gaussian noise. This proposed method is designed based on super position principle and 8 cases of signal varied positions. This new method output is fed to the wavelet technology compare the existing LMS, RLS algorithm using Matlab 6.5 simulation results. Final computer simulation shows the more effective and robustness with comparison of .existing method.

Key words: Active noise control, signal to noise ratio, Grazing estimation method, Pseudo code, Matlab 6.5

INTRODUCTION

The signal traveling over the medium gets corrupted by noise, which degrades the signal quality. Apart from noise the signal undergoes through lot many degrading effects that are inherent of the medium, which adds to signal degradation. These degrading components are highly random in nature. The filtering of the informative signal from this highly corrupted signal is a difficult task. The adaptive techniques are efficient in canceling noise when the reference noise used is correlated to the noise corrupting the signal is presented by Widrow and Stearns (1985). Since the noise gets added in the channel and is totally random, hence there is no means of creating a correlated noise, at the receiving end. Only way possible is to extract the noise from the received signal itself as only the received signal can say the Story of the noise added to it.

The technique used in this study is a two way process. As the first step try to estimate a signal correlated to the actual signal i.e., the information bearing component of the received signal. The second method used for this generation is presented. Since the signal and the noise are non-coherent to each other, this signal is used to extract noise from the received signal using the interference cancellation technique of the adaptive signal processing, thus giving us noise which to a good extent will be correlated to the noise in the received signal proposed by Kuo and Morgan (1996). A technique to generate a signal correlated to the actual signal, which is thus the foremost step in generating correlated noise, is presented next.

PROBLEM STATEMENT

Our aim is to achieve noise reduction for signals transmitted through the wireless medium. In such a communication, all the noise is added in the channel. The noise is highly random. Here there is no source for obtaining a correlated noise at the receiving end. Only the received signal can tell the story of the noise added to it. Only if it is possible extract the noise from the received signal through some means then the above mentioned adaptive techniques to enhance the signal to noise ratio of the received signal. A method to obtain a correlated noise from the received signal itself.

MATHEMATICAL MODELLING OF GRAZING ESTIMATION OF SIGNAL METHOD

The adaptive techniques to reduce noise are effective when the reference noise is highly correlated to the corrupting noise invented by Morgan (1980). But owing to the highly random nature of the corrupting noise, it is difficult to estimate it. Here, to generate an effective reference noise from the received signal itself, this can be then used to reduce the noise content of the same received signal. The technique used is that of trying to graze through the informative signal and thus trying to find the approximate noise and information content at every instant is obtained from Kuo and Morgan (1999).

This technique is based on having first two samples of the original signal correctly. Next to estimate the third sample using the first two samples. This is done by finding the slope between the first two samples and

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extending the same for the third sample. This next estimated sample is subtracted with the value at that instant in the received signal. This value gives the estimated noise sample at that instant and we call it the as the estimated noise. Now the second and the third samples are used to estimate the fourth sample in a similar way as was the third sample found. The same method is carried on in generating all the higher samples.

$$m = ES_{n-1} - ES_{n-2} \quad (1)$$

$$ES_n = ES_{n-1} + m \quad (2)$$

$$N'_n = X_n - ES_n \quad (3)$$

Where, ES represents the estimated signal, X represents the received signal and N' represents the estimated noise of the first stage also a threshold for the estimated noise set. This threshold is based on the likely level of noise. The threshold level can be near around 0.5 times the max absolute value that the noise can take. When ever the absolute value of the estimated noise level crosses this preset threshold level, the estimated signal value at that instant is reset. i.e. when

$$N'_n > \text{threshold, then}$$

$$ES_n = ES_n + N'_n \quad (4)$$

This ensures that we don't just keep moving in a single direction. When ever there is more than the expected deviation, we try to bring the estimated signal value in proximity of the signal value. This way we try to keep our estimated signal samples in close proximity to the original signal through out the course of estimation presented by Manikandan.

PROPOSED METHOD FOR POSSIBLE CASES

To understand how the above technique works, let us consider the following cases given by Vijayan (1994), Here:

- • Denotes the actual sample of the signal.
- □ Denotes the estimated sample at that instant.
- ✧ Denotes the value due to addition of noise at that instant i.e., it denotes the sample of the received signal.

Case 1: In the Fig. 1, 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 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.

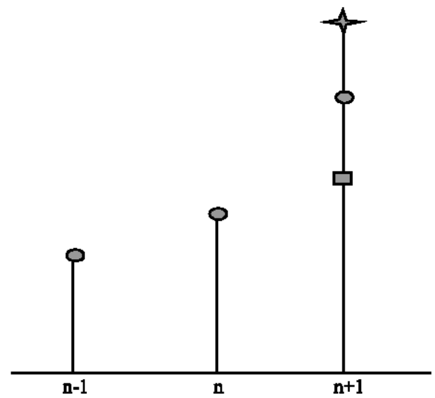

Fig. 1: Case 1

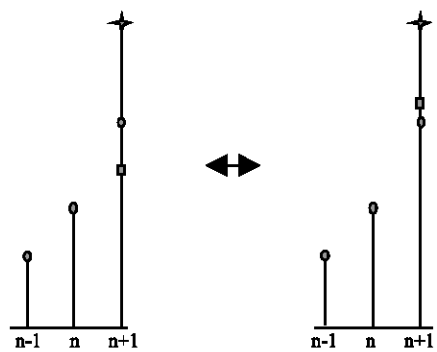

Fig. 2: Case2

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. Thus the estimated value has to be adjusted, as shown in the Fig. 2. In this case we see that the clean signal value, the estimated signal value and the received signal value all have in the same direction. This as explained in case 1, the estimated signal is correlated to both the clean signal and the noise at the $(n+1)^{th}$ instant.

Case 3: This case is similar to case 1, except that the noise added at instant n+1 is negative. Again in this case the estimated signal and actual signal have the same sign and opposite to that to the noise (Fig. 3).

Case 4: In this case the noise is high negative, so that the difference between the estimated value and the received

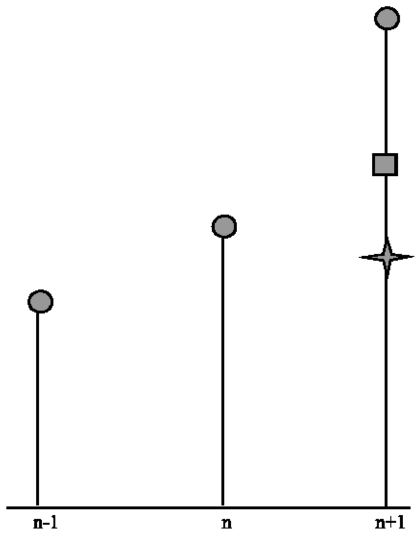

Fig. 3: Case 3

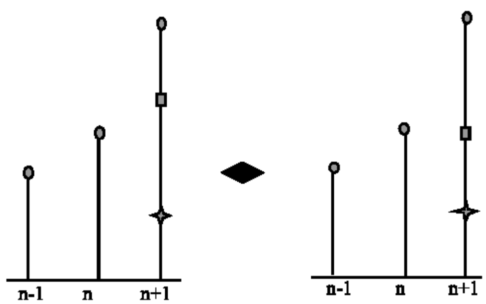

Fig. 4: Case 4

value is more than the threshold. Hence we read just the estimated value (Fig. 4).

Again the estimated signal has the same sign as the actual signal and opposite to that of the noise.

Case 5: Now we will consider cases where there is a change of slope between the $(n+1)^{th}$ and n^{th} samples as compared to that of between n^{th} and $(n-1)^{th}$ samples. In this case the noise is positive, as shown in the figure above. As can be seen, the estimated signal and the actual signal and the noise in this case will be in the direction (Fig. 5).

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 (Fig. 6).

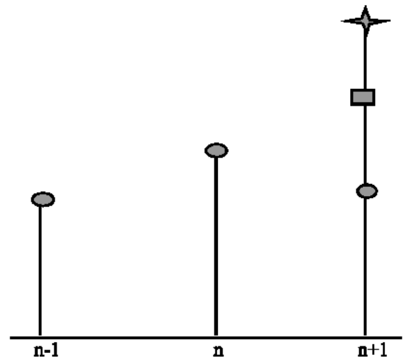

Fig. 5: Case 5

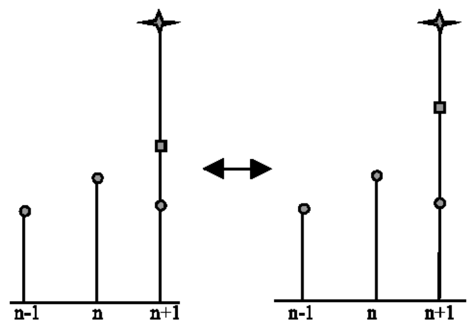

Fig. 6: Case 6

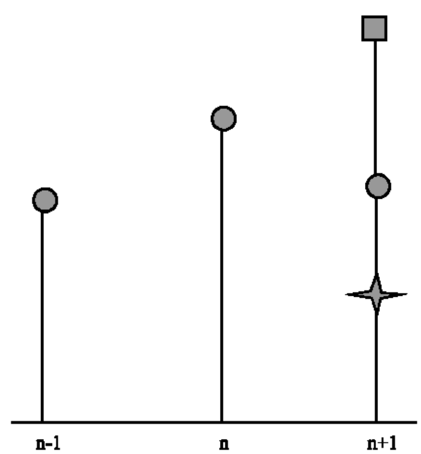

Fig. 7: Case 7

Case 7: In this case the noise getting added is negative. Thus as shown, the estimated and the clean are in the same direction and opposite to that of the noise (Fig. 7).

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 (Fig. 8).

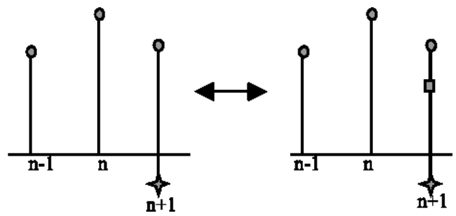

Fig. 8: Case 8

In either case we notice that the estimated and the clean signal are in the same direction, whereas the noise will have opposite direction.

Thus we have seen all the possible cases for the values in the positive directions. In all the cases, the estimated signal and the clean signal are in the same direction. Only at the, instants, when the signal value crosses the time axis, there are chances of the estimated signal and the clean signal to be opposite directions. These have very less chances. Compared to this, the noise and estimated signal are in same direction in only half of the cases. Also the level of correlation between them is low. For the whole range of the signal, the estimated signal closely follows the clean signal. Based on the above discussion, the model of how the estimated signal follows the clean signal is shown in the Fig. 9 (Elliott and Nelson, 1993).

Thus using this scheme, we can generate the well correlated signal to the actual signal intended to transmit. After generating the estimated signal, we use it as the 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 the noise signal itself. The level of correlation of this generated noise and the actual noise depends on the level of the correlation between the estimated signal and the clean signal. Thus we have now achieved our basic aim of generating a correlated noise to the noise corrupting the signal. This generated noise to cancel the noise in the received signal by using the interference cancellation application again introduced by Davis (2002). This is based on the assumption that speech signal occupy low frequency regions and hence there will not be many sudden changes in it in the time domain, i.e. the signal will be smooth to the required extent.

There are least chances that the estimated sample coincides with the actual value of the original signal at each instant. But it is sufficient to serve the purpose of generating signal which is correlated to information signal to good extent. The block diagram of the above process is shown in Fig. 10. The received signal is passed through the grazing estimation block, the output of which is an

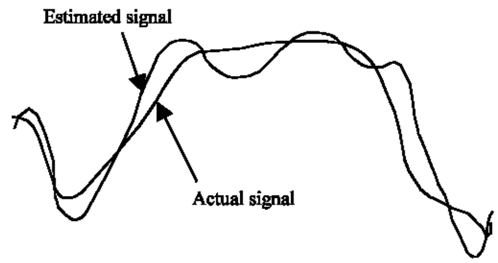

Fig. 9: Depiction of estimated signal

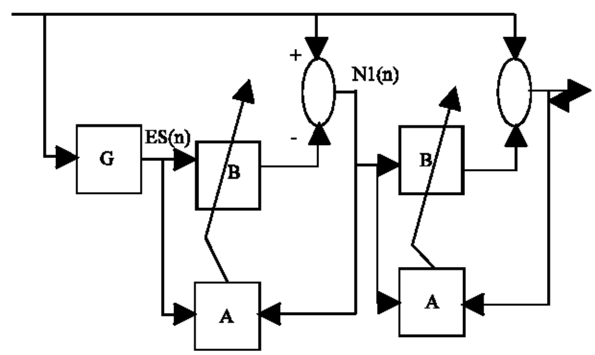

Fig. 10: Block diagram of Grazing estimation method

estimated signal, denoted by ES (n), which will be correlated to the original signal. Next, this ES (n) is passed through adaptive filtering where it cancels the information content of the received signal, which is totally corrupted with noise prepared by Eriksson *et al.* (1994). The efficiency of this cancellation depends on the correlation level of the estimated signal with that of the original signal. With the above suggested method, it was seen that a good cancellation was possible. Since the noise and the information signal are uncorrelated what we get at the output is something, which will be well correlated to the corrupting noise in the received signal, which is the aim of the paper. This generated noise is denoted as N' (n) in Fig. 1. N' (n) is again passed through adaptive signal processing techniques to cancel the noise content present in the received signal, as shown in Fig. 1. Reduction of noise content to an extent of 15-20 db. was achieved. The above technique was simulated on matlab-6.5 software by Elliott and Nelson (1993). The codes for grazing estimation are presented.

PROPOSED GES ALGORITHM

Grazing (rec, signal (1), signal (2)),
Load index (1), index (2) = signal (1), signal (2)
noise1, noise (2) = rec (1)-index (1),
Rec (2)-index (2)

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Len = length of the signal
For n = 3 to Len
 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
x (n) = s(n)+N(n)
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Where the following letters denote the following blocks:

A = Adaptive algorithm
B = Filter
G = Grazing Estimation

RESULTS AND DISCUSSION

The matlab simulation output results is shows the Fig. 11, where 'utopia windows start' wav. File is used. It represents all the stages of the process, viz. the original, the noise corrupted signal, the estimation of the signal using grazing estimation and finally the output signal.

It can be seen that the o/p signal is very close to the original one. The gain between the received and filtered signal is 30db (approx). Similar results were obtained for different test signals of standard wav. Files viz. Ding and Tada. The results in frequency domain are as in the Fig. 12.

It can be seen that the frequency components representing noise has been well reduced in the filtered signal Compared to that in the received signal, thus emphasizing the efficiency of the method. Ratio (in db) of the filtered signal, against the PSNR ratio (in db) of the received signal. It can be seen that lesser the PSNR of the received signal more is the improvement in the filtered signal (Fig. 13). This is so because, in our grazing estimation, we estimate the next sample, which is definitely going to be different from the original sample at that instant by Burgess (1981).

We can assume again this estimated signal, as, original signal with some noise, which is correlated to the original noise to some extent. But when the received signal is deeply buried in noise, then the noise component in the estimated signal will be a small fraction of the original noise. Thus in the first set of adaptive cancellers, to cancel the signal, this estimated noise will have insignificant effect. Thus in such cases, the generated noise, at the o/p of the first stage of adaptive canceller, will be well correlated to the actual corrupting noise. This

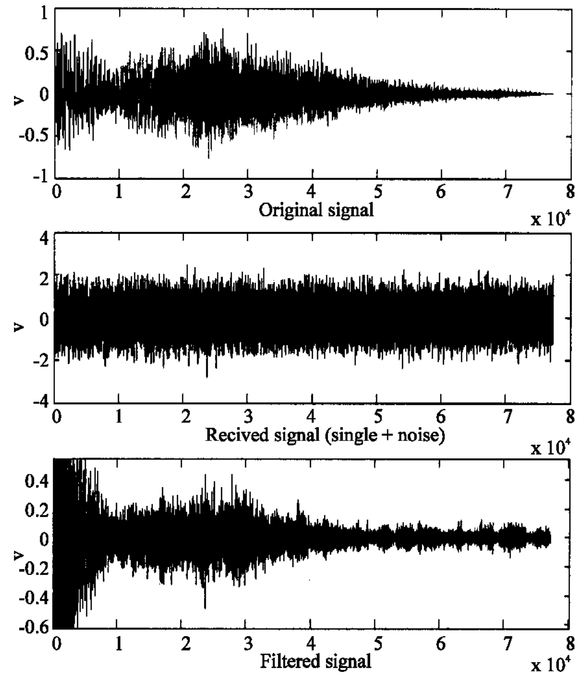

Fig. 11: System performance in time domain

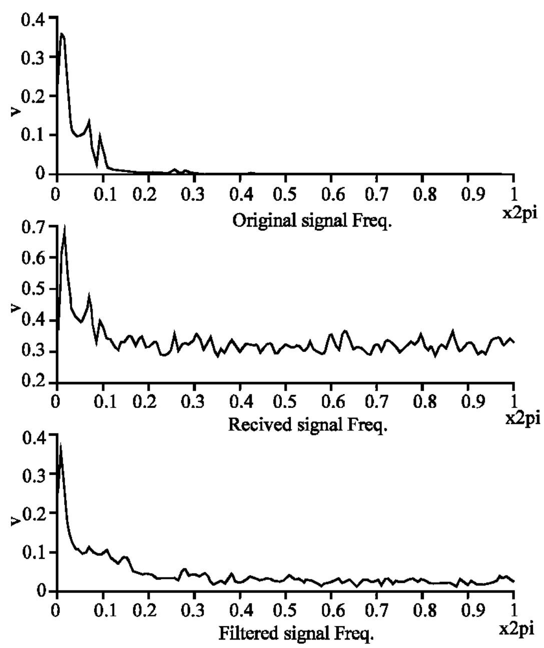

Fig. 12: System performance in frequency domain

permits us to transmit signal at low voltages. This can be considered as one or the advantages of this method presented by Boucher *et al.* (1990).

Table 1 shows the perfect noise reduction obtained using proposed GES + Wavelet method. The Improved signal [>20db] is obtained by maximum PSNR value using this method.

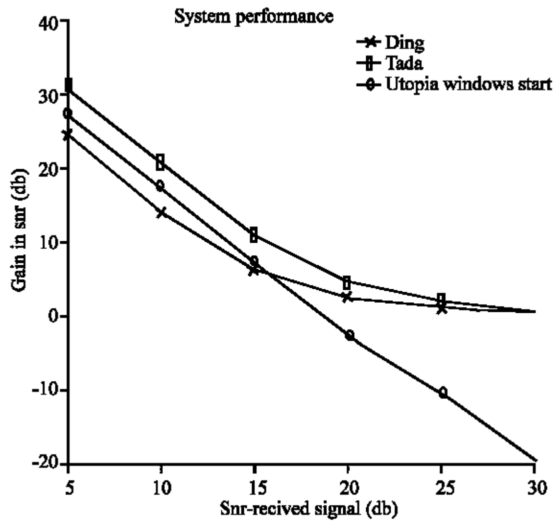

Fig. 13: Represents the gain in the PSNR

Table 1: Comparisons of various methods for s1omwb speech signal and ding sound signal

Received signal PSNR (db)	PSNR of recovered Grazing estimation	Wavelet de-noising	Grazing estimation + wavelet de-noising
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

CONCLUSION

This is an introduction of a new approach. A method called signal grazing was introduced to generate noise at the receiving end, which is coherent to the noise corrupting the signal. Thus reduction of noise level in received signal using adaptive signal processing technique was possible. It could be seen that in general PSNR enhancement of 15-20 db is possible, when the signal is deeply immersed in noise. It can also be seen that the gain in PSNR is high when the signal is more deeply immersed in noise. This gives it an advantage of allowing the transmission of signal with low power. The above given system could be cascaded with other PSNR enhancement technique at the receiving end, viz. de-emphasis in case of analog communications and before matched filter in digital based communication to obtain an overall optimum PSNR enhancement. But all this would need high speed processors. In future this new method is planning to implement on real-time using DSP processor.

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REFERENCES

Boucher, C., S.J. Elliott and P.A. Nelson, 1990. The effect of modeling errors on the Performance and stability of active noise control systems. Proc. Recent Adv. Active Control of Sound Vibrat., pp: 290-301.

Davis, G.M., 2002. Noise Reduction in Speech Applications. CRC Press, London, UK.

Eriksson, L.J., M.C. Allie, D.E. Melton, S.R. Popovich and T.A. Laak, 1994. Fully Adaptive Generalized Recursive Control Systems for Active Acoustic Attenuation. Proceeding of ICASSP, Vol. II.

Elliott, S.J. and P.A. Nelson, 1993. Active Noise Control. IEEE. Signal Processing Magazine.

Kuo, S.M. and D.R. Morgan, 1999. Active Noise Control: A Tutorial Review. Proceedings of the IEEE, Vol. 87.

Kuo, S.M. and D.R. Morgan, 1996. Active Noise Control Systems: Algorithms and DSP Implementations. John Wiley and Sons, New York.

Morgan, D.R., 1980. An Analysis of Multiple Correlation Cancellation Loops with a Filter in the Auxiliary Path. IEEE Transactions of Acoustic, Speech and Signal Processing, Vol. ASSP-28.

Vijayan, D., 1994. Feedback active noise control systems, masters thesis. Northern Illinois University.

Widrow, B. and S.D. Stearns, 1985. Adaptive Signal Processing. Prentice-Hall, Inc. Englewood Cliffs, N.J.