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Study of White Gaussian Noise with Varying Signal to Noise Ratio in Speech Signal using Wavelet

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Abstract: *The present work provides the wavelet based mechanism to analyze the effect of white Gaussian noise in the input speech signal. The white Gaussian noise (WGN) is imposed in the captured input speech signal and the signal is denoised using wavelet tree decomposition, filtration and reconstruction process. The mean square error (MSE) and mean absolute error (MAE) have been calculated in denoising process. The process is repeated many times for different values of signal to noise ratio (SNR) in additive white Gaussian noise. The comparative analysis of the mean square error and mean absolute error has been produced for all the cases. All the graphical and experimental works have been implemented in MATLAB.*

Keywords: *White Gaussian Noise (WGN), Signal to Noise Ratio (SNR), Mean Square Error (MSE), Mean Absolute Error(MAE), Wavelet.*

I. INTRODUCTION

The audio signals are produced from a sound, which generates the vibrations in the audible frequency range to form pressure waves. The human ear receives these pressure signals and sends them to evoke the brain. The attenuation, noise and distortion always affect the sound until the system is made prone to these factors. Speech signal synthesis is very important for various applications[1].

The existence of noise is inevitable in real applications of speech processing. In fact, background noise is one of the major factors that adversely affect the perceived grade of service in speech communication system. It is well known that the additive noise affects mainly the performance of the system and reduces the Signal to Noise Ratio (SNR) and the speech intelligibility. A noise reduction scheme, capable of handling a wide variety of noise situations with varying characteristics and noise levels, becomes necessary. The traditional approach to noise cancellation lay in utilizing standalone noise cancellation modules on the near-side or transmit path. This approach works well under constant conditions, but as environment changes, the performance gets degraded and the system struggles to adapt[2].

II. BASIC TERMINOLOGY

A. White Gaussian Noise

- Gaussianity refers to the probability distribution with respect to the value.
- The probability of the signal falling within any particular range of amplitudes.
- The term 'white' refers to the way the signal power is distributed independently over time or among frequencies.

B. Noise Cancellation

The usual method of estimating a signal corrupted by additive noise is to pass it through a filter that tends to suppress the noise leaving the signal relatively unchanged i.e. direct filtering. The design of such filters is the domain of optimal filtering, which was originated with the pioneering work of Wiener and was extended by Kalman, Bucy and Others. Filters used for direct filtering can be either fixed or adaptive[3,4].

B.1 Fixed Filters

The design of fixed filters requires a priori knowledge of both the signal and the noise, i.e. if we know the signal and noise beforehand, we can design a filter that passes frequencies contained in the signal and rejects the frequency band occupied by the noise.

B.2 Adaptive Filters:

Adaptive filters, on the other hand, have the ability to adjust their impulse response to filter out the correlated signal in the input. They require little or no a priori knowledge of the signal and noise characteristics. If the signal

is narrowband and noise broadband, which is usually the case, or vice versa, no a priori information is needed, otherwise they require a signal(desired response) that is correlated in some sense to the signal to be estimated. Moreover adaptive filters have the capability of adaptively tracking the signal under non-stationary conditions.
Noise cancellation is a variation of optimal filtering that involves producing an estimate of the noise by filtering the reference input and then subtracting this noise estimate from the primary input containing both signal and noise. It makes use of an auxiliary or reference input which contains a correlated estimate of the noise to be cancelled. The reference can be obtained by placing one or more sensors in the noise field where the signal is absent or its strength is weak enough. Subtracting noise from a received signal involves the risk of distorting the signal and if done improperly, it may lead to an increase in the noise level[5].

C. Wavelet

Wavelet theory provides a unified framework for a number of techniques which had been developed independently for various signal processing applications. For example, multi resolution signal processing, used in computer vision; subband coding, developed for speech and image compression; and wavelet series expansions, developed in applied mathematics, have been recently recognized as different views of a single theory. In fact, wavelet theory covers quite a large area. It treats both the continuous and the discrete-time cases. It provides very general techniques that can be applied to many tasks in signal processing, and therefore has numerous potential applications.[6]. A wavelet is a waveform of effectively limited duration that has an average value of zero and nonzero norm.Sinusoidal waves are smooth and predictable, while wavelets tend to be irregular and asymmetric.Wavelet method is a basic method that is used for noise filtering, compression and analysis of non-stationary signals. It is an appropriate method for semi-stationary signals which provides a good resolution in both time and frequency domain. The wavelet transform produces better results than traditional methods in improving speech[7,8].

D. Signal To Noise Ratio

- SNR is the ratio of signal power to the noise power.
- In terms of signals it indicates, how the original signal is affected by the added noise.
- SNR is given by the following formula:

SNR= Average Signal Power/ Average Noise Power

E. Peak Signal to Noise Ratio

Peak signal to noise ratio (PSNR) is usually expressed in terms of the logarithmic decibel scale,
where Max is the maximum value attained by the signal.

$$PSNR = 10.\log_{10}\frac{MAX_I^2}{MSE} = 20.\log_{10}\frac{MAX_I}{\sqrt{MSE}}$$
$$= 20.\log_{10}(MAX_I) - 10.\log_{10}(MSE)$$

F. Mean Absolute Error (MAE)

The MAE measures the average magnitude of the errors in a set of forecasts, without considering their direction. It measures accuracy for continuous variables. The MAE is a linear score which means that all the individual differences are weighted equally in the average.

$$\text{MAE}= \left(\sum_{i=1}^{n} |P(i)-Q(i)| \right) / n$$

G. Mean Squared Error (MSE)

The MSE is a quadratic scoring rule. The difference between forecast and corresponding observed values are each squared and then averaged over the sample.
This means the MSE is most useful when large errors are particularly undesirable.

$$\text{MSE}= \left(\sum_{i=1}^{n} [P(i)-Q(i)]^2 \right) / n$$

III. ALGORITHM FOR THE ANALYSIS OF AWGN

1. Take the input from the end user and store it as a wav file.
2. Add white Gaussian Noise in the original Signal with given value of SNR.
3. Express the acquired signal in the form of wavelet tree of multiple levels.
4. Denoise the noisy signal wavelet tree with the help of wavelet filtration process.
5. Reconstruct the denoised signal to produce the noise free output from the wavelet tree after filtration process.

6. Calculate Peak Signal to Noise Ratio (PSNR), Mean Absolute Error (MAE) and Mean Squared Error for Denoising process.

Flow Chat for the study of White Gaussian Noise in speech signal is shown in Figure 1.

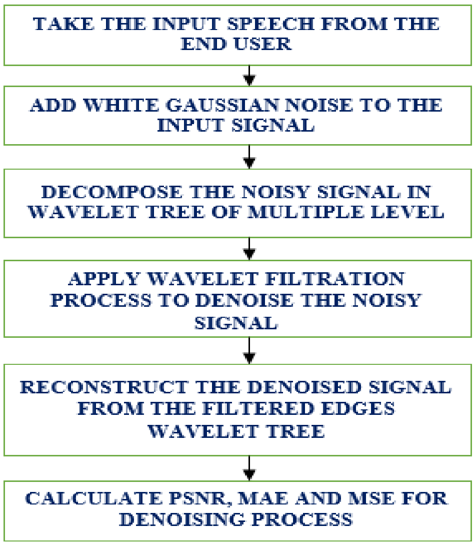

Figure 1: Flow Chart to study additive white Gaussian Noise in speech signal

IV. RESULTS AND DISCUSSION

In the present study, initially we have taken the input from a user, which has been stored in the given file, then nature of white Gaussian noise is studied for 50 different values of Signal To Ratio ranging 1 to 50 magnitude by decomposing the signal with the help of wavelet tree decomposition method. The signal is reconstructed and noise is studied in every case. We have calculated Mean Absolute Error, Mean Squared Error and Peak Signal to Noise Ratio for each case. Figure 2 shows the Mean Absolute Error with increasing SNR, which clearly shows that as soon as SNR increases, MAE is decreased towards zero.

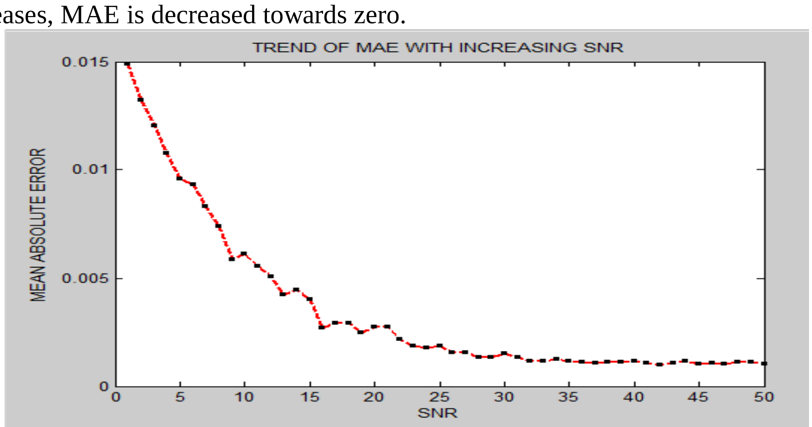

Figure 2: Mean Absolute Error with increasing Signal to Noise Ratio

Figure 3 shows the Mean Squared Error with increasing SNR, which clearly shows that as soon as SNR increases, MSE is converged towards zero. Hence higher values of SNR are the cause of least error in the system.

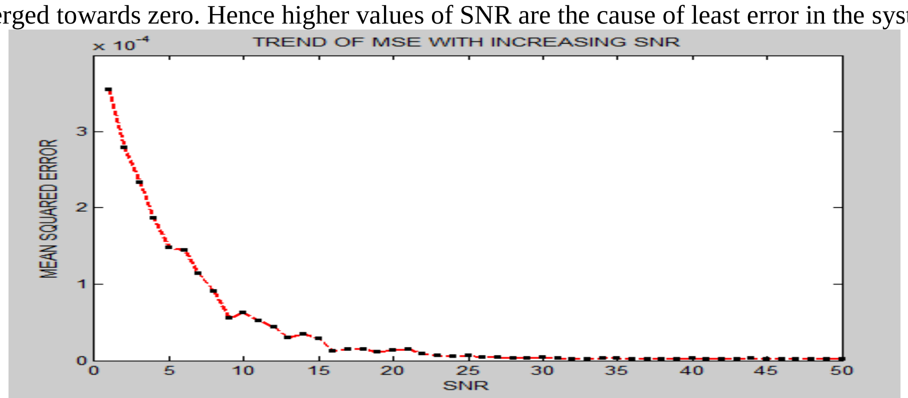

Figure 3: Mean Squared Error With Increasing Signal to noise Ratio

Figure 4 shows the trend of Peak Signal to Noise Ratio with Varying SNR value for any speech signal. Form the figure it is clear that PSNR is also increased as soon as SNR increases.

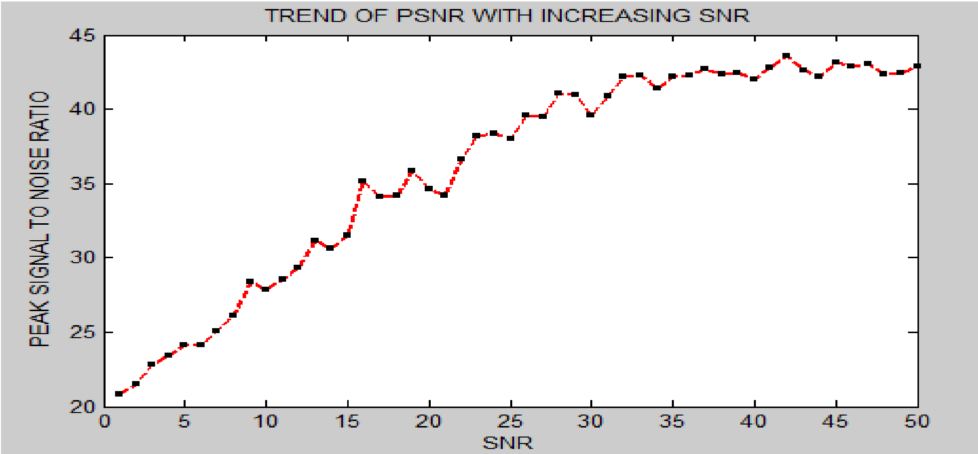

Figure 4 :Peak Signal to Noise Ratio with Increasing Signal to Noise Ratio

Table 1 shows the experimental results for 50 different values of SNR and accordingly changes in MAE, MSE and PSNR.

Table 1 : Comparative values of PSNR, MAE and MSE for given SNR

SNR(awgn)	PSNR	MAE	MSE
1.000000	19.06753346	0.01597779	0.00041488
2.000000	22.62021913	0.01225435	0.00023910
3.000000	22.42535967	0.01167325	0.00021817
4.000000	23.99842489	0.00960622	0.00014712
5.000000	23.53053201	0.00976086	0.00015467
6.000000	23.44631596	0.00954710	0.00015208
7.000000	25.55228243	0.00783434	0.00009979
8.000000	27.03922011	0.00690158	0.00007754
9.000000	26.98740603	0.00668165	0.00007467
10.000000	27.52186398	0.00622715	0.00006574
11.000000	28.30574314	0.00577301	0.00005741
12.000000	30.85792490	0.00434487	0.00003101
13.000000	30.79477671	0.00440112	0.00003253
14.000000	31.55151822	0.00399151	0.00002691
15.000000	32.16039231	0.00365411	0.00002275
16.000000	35.34790496	0.00261536	0.00001120
17.000000	33.18826307	0.00320348	0.00001791
18.000000	33.65565001	0.00307668	0.00001691
19.000000	34.63259343	0.00275584	0.00001353
20.000000	35.67881905	0.00248127	0.00001082
21.000000	35.25597733	0.00250707	0.00001155
22.000000	37.06215864	0.00207250	0.00000757
23.000000	38.79438076	0.00171366	0.00000504
24.000000	39.66200189	0.00157095	0.00000422
25.000000	38.27960105	0.00178987	0.00000578
26.000000	41.21004136	0.00131571	0.00000295
27.000000	39.63630198	0.00153284	0.00000419
28.000000	41.17553092	0.00130864	0.00000300
29.000000	41.50751615	0.00125645	0.00000278
30.000000	40.94585359	0.00132622	0.00000315
31.000000	40.18588513	0.00142083	0.00000370
32.000000	42.39886310	0.00112741	0.00000223
33.000000	41.03367267	0.00129908	0.00000307
34.000000	41.50979569	0.00123379	0.00000275
35.000000	41.13501197	0.00127669	0.00000298
36.000000	43.34306198	0.00101069	0.00000180
37.000000	42.46086724	0.00110636	0.00000219
38.000000	42.49977479	0.00110097	0.00000218
39.000000	42.84330863	0.00106001	0.00000200
40.000000	42.22521510	0.00113420	0.00000232
41.000000	42.75817095	0.00106867	0.00000205
42.000000	42.74095326	0.00107130	0.00000206
43.000000	43.28583736	0.00101095	0.00000182
44.000000	42.69765451	0.00107557	0.00000208
45.000000	41.52557801	0.00121682	0.00000272
46.000000	42.59825381	0.00108677	0.00000213
47.000000	42.01851229	0.00115599	0.00000244
48.000000	41.88736919	0.00117205	0.00000251
49.000000	42.69847551	0.00107443	0.00000208
50.000000	42.11083582	0.00114355	0.00000238

V. CONCLUSION

In the present study, we have analyzed the behavior of Additive White Gaussian Noise with varying signal to noise ratio. We have drawn the conclusion that *as soon as the SNR increases the mean absolute error and mean squared error are reducing and tending towards Zero for higher values of SNR and the Peak Signal to Noise Ratio (PSNR) is increased as SNR is increases.* This work is very significant for determining the behavioral trend of Gaussian noise in speech signals, which may be helpful for different noise reduction and noise cancellation algorithms.

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