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Speech Enhancement In Dual Microphones In Mobile Communication Using Adaptive Filters

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Abstract—The speech enhancement is one of the imperative proceduresuse to enhance the nature of a speech signal i.e. debased by noise. In this paper, a twofold recipient speech change computation for the phones is proposed. The adopted technique exploitsthe intelligence work calculation and the Kalman channel. This estimation has a clear execution that does not require a forecast of intruding signs estimations. Besides, this count can be used as a piece of little devices with so almost expel between the two recipients Moreover, the usage of such figuring grants reducing diverse commotion sources at various azimuths positions. Finally, the new computation shows its presentations suggesting the perceptual evaluation of talk quality and the time zone waveforms. However the perceptual attributes of the discourse flag relies on the perceptual attributes of human ear. To additionally enhance the execution of discourse upgrade framework, Kalman channel is replaced with wiener channel, which depends on covering attributes of human sound-related framework.

Index Terms—Coherence function, Speech Enhancement, Kalman filter, wiener filter, Peak to signal ratio and Perceptual evaluation of speech quality measurements.

1 INTRODUCTION

SPEECH is one of the main form of communication which is spoken by human. Vocal cord plays one of the major role in supporting the speech communication. The spoken words are created with the help of phonetic combinations. The phonetics consists of limited set of vowels and consonants. Different speech sounds units are found according to the human languages based on the syntax structure and the vocabularies that are used. About 1000 different wordsvocabularies are available. With the help of the vocal ability speech enables them to sing and even enable to produce different sounds.

They are different type of human communications in which speech language and for deaf a gesture form of language is available which is said to be a sign language. Speech is used in different formats, such as in mental processes to enhance and organise the meaning of the interior monologue. Speech also is termed as a basis written language for some cultures.

Speech is examined as far as the Speech creation and speech view of the sounds utilized as a part of vocal dialect. Other research subjects concern speech repetition, the capacity to outline talked words into the vocalizations expected to reproduce them, which assumes a key part in vocabulary development in youngsters and speech blunders. A few scholarly controls concentrate these, including acoustics, brain science, phonetics, subjective science, correspondence studies, and otolaryngology and software engineering. Another territory of research is the means by which the human mind in its diverse ranges, for example, the Broca's region and Wernicke's region underlies speech.

It is disputable how far human discourse is one of a kind; in that creatures likewise speak with vocalizations. While none in the wild have perfectly huge vocabularies, inquire about upon the nonverbal capacities of dialect prepared primates, for example, Washoe and Kanzi raises the likelihood that they may have these abilities. The birthplaces of discourse are obscure and subject to much level headed discussion and theory.

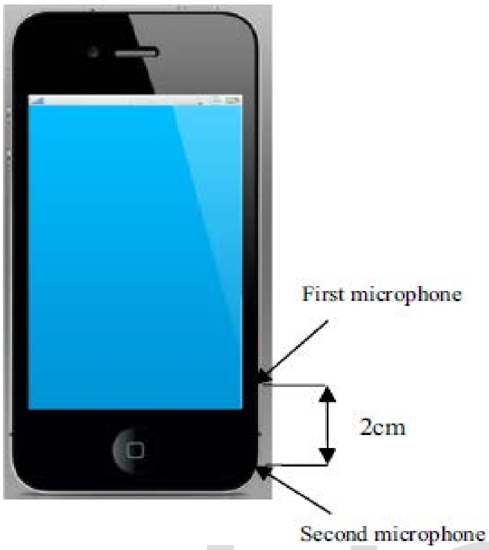

Fig.1.Illustration of mobile phones with closely spaced microphones.

2LITERATURE REVIEW

Billions of individuals around the globe utilizing cell phones have issues in phone discussion when the discourse is debased by the clamor. Consequently, unique discourse improvement systems were included to lessen the meddling signs in cell phones. Especially, singlemouthpiece commotion diminishment strategies demonstrate its adequacy amid fifty years in many fields as they are anything but difficult to apply [1,2]. Be that as it may, the execution of single channel discourse upgrade calculations is constrained on the non-stationaryclamor case because of the issues identified with the estimation of this commotion utilizing such calculations [2]. Accordingly, discourse quality can be enhanced utilizing multi-channel discourse upgrade calculations [3]. For that, progressively the quantity of mouthpieces is expanded increasingly the nature of discourse will be moved forward. Conversely, an extraordinary number of receivers is hard to actualize in mobile phones and requires more computational many-sided quality. The utilization of double amplifier discourse upgrade calculations can take care of these two issues refered to beforehand. Additionally, these calculations are indicated by its great exhibitions in term of discourse quality and coherence. The writing is advanced by many works which treat a few double channel discourse improvement techniques. Among them, we can refer to the work given by Youssefian et al. [4] which delineated a double channel discourse upgrade calculation utilizing power level distinction for close field. This calculation abuses the distinction of the power motions in the two receivers as a model for clamor decrease. In [5], Youssefian and Lazio exhibited a double amplifier discourse improvement calculation in light of the rationality work. The proposed methodology treats the rationality between the objective and commotion motions as a measure for clamor decrease and can be utilized for barely divided amplifiers. Koldovsky [6] proposed a commotion decrease double receiver in cell phones utilizing a bank of pre-measured target-cancelation channels. This technique depends on an objective cancelation channels misused to assess the commotion, which is then subtracted from the uproarious discourse utilizing Wiener channel or power level contrast calculation. A double receiver discourse improvement strategy in cell phones is proposed in [7]. This technique depends on the between receiver Posteriori SNR Difference (PSNRD) for Speech Presence Probability (SPP) estimation and a MVDR Filter for commotion diminishment. At last, the given work by Prajna et al. [8] exhibited another calculation in light of gravitational hunt calculation (GSA). This approach utilized heuristic calculation for commotion diminishment. Contrasted with this works.

In this paper a dual-channel speech enhancement algorithm dedicated to mobile phone applications using the coherence function and the wiener filter. The proposed method can be applied to all mobile phones as shown in figure , which has high level of computational complexity. This approach utilized heuristic algorithm for reduction of noise. Contrasted with this works, we introduce in this paper a double channel speech improvement algorithm committed to cell phone applications utilizing the coherence function and the wiener filter.

3 PROPOSED SYSTEM

3.1 Determining Coherence function

Coherence function is presented for the noisy signals. It is one of the parameter which is used to predict the noisy signals. In our proposed work we assume two microphone which are been placed in a environment which is effected with noise. The signals received by the micro phones are given as:

$$L_1(x) = s_1(x) + n_1(x) \quad (1)$$

$$L_2(x) = s_2(x) + n_2(x) \quad (2)$$

where

L1(x) and L2(x) - Noisy signals of the two microphones

'x' is the simple index,

s1(x) and s2(x) - signals obtained at each microphone

n1(x) and n2(x) - the noise signal of the two microphones.

The Fourier transform of the two noisy signals can be described as follows:

$$L_1(y,f) = S_1(y,f) + N_1(y,f) \quad (3)$$

$$L_2(y,f) = S_2(y,f) + N_2(y,f) \quad (4)$$

where,

' y' represents the frame index

' f' is frequency bin.

The coherence function algorithm is a basic dual microphone speech enhancement method proposed in [9,10]. This technique is based on a correlation between speech signals in the two microphones instead of the noise signals are uncorrelated. The coherence function between the signals x1 and x2 received by the two microphones is presented as follows:

$$F_{coh}(y,f) = \frac{P_{x1x2}(y,f)}{\sqrt{P_{X1}(y,f)P_{x2}(y,f)}} \quad (5)$$

where,

Px1x2(y,f) - cross power spectral density (CPSD) of the two noisy signals x1 and x2,

Px1 (y,f) and Px2 (y,f) - power spectral density (PSD) of x1 and x2.

The goal of the intelligence procedure is to decide whether the speech signal at a particular recurrence container is available or absent though it is relative to the size of the coherence function. The speech signal is administering when the greatness is near one and the noise signal is administering when the extent is near zero. The last theory is genuine when the noise signals at the two receivers are not all that quite connected. Usually, noise and distance are inversely proportional to each other, hence the correlation of the noise signals at the two microphones increases when the distance between them decreases [14].

Thus, the two microphones are too much coherent in our work because they are so closely spaced. The design of the two microphones with target sound sources and the approximate coherence function adopted by this work are detailed in [5]. The two microphones are placed on a duplicate head with a 2-cm distance between them. The clean speech source is placed at 0_ azimuth and the noise source is at b_ azimuth. The distance between the two sources and the microphones are 1.2 m. Based on the previous assumptions, the near coherence function is computed as:

$$F_{x1x2} = [cos(\omega\tau) + j\,sin(\omega\tau)]\frac{SNR}{1+SNR} + [cos(\omega\tau\,cos\,\beta) + j\,sin(\omega\tau\,cos\,\beta)]\frac{1}{1+SNR} \quad (6)$$

3.2 Speech Enhancement Algorithm

The proposed double channel Speech upgrade algorithm comprises of two stages. The initial step is the double amplifier speech improvement algorithm in view of the coherence function proposed in [5]. The second step is the Kalman channel which displays a decent answer for the direct MMSEtroubles for the stochastic framework by getting an ideal forecast of the perfect discourse [11]. Fig.2. presents the piece graph of the existing speech upgrade calculation.

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It demonstrates the time postpone remuneration connected by numerous double microphone speech improvement strategies. In addition, the figure depicts the inexact coherence function characterized by (6).

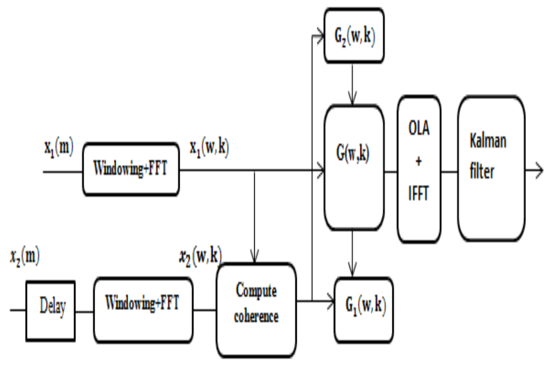

Fig.2.Existing Algorithm

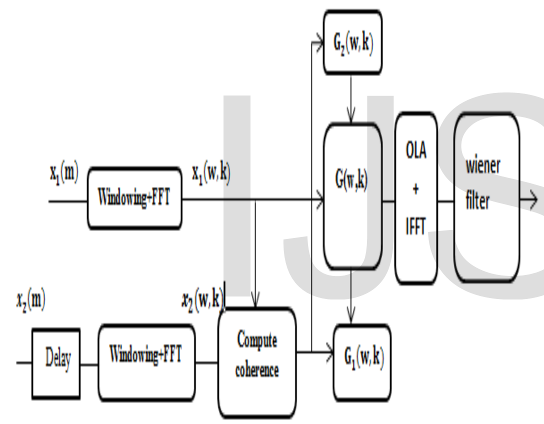

Fig.3.Proposed Algorithm

3.3 wiener filter

Typical deterministic filters are designed for a desired frequency response. However, the design of the 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 signas possible. Wiener filters are characterized by the following:

3.3.1 Assumption

Signal and (additive) noise are stationary linear stochastic processes with known spectral characteristics (or) known autocorrelation and cross-correlation.

3.3.2 Requirement

The filter must be physically realizable/ causal (this requirement can be dropped, resulting in a non-causal solution) performance criterion: minimum mean square error (MMSE).

3 RESULTS

4.1 Kalman Results

Using the Kalman Filter for the existing algorithm for the good speech quality.

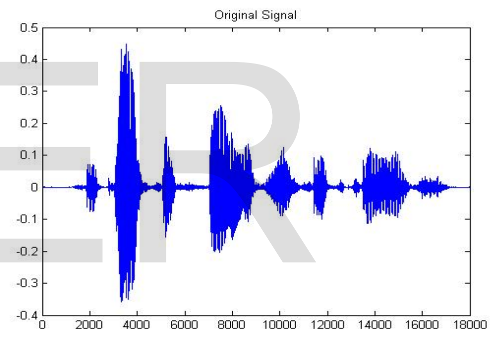

Fig.4.Clean speech signal

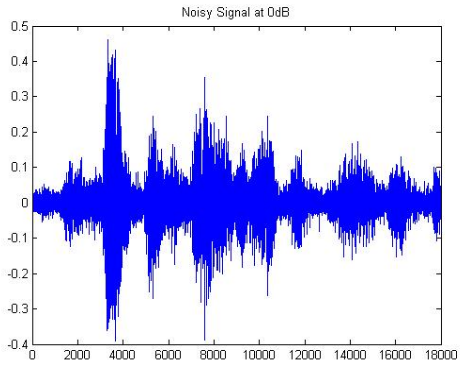

Fig.5.Speech signal which is recorded near to microphone

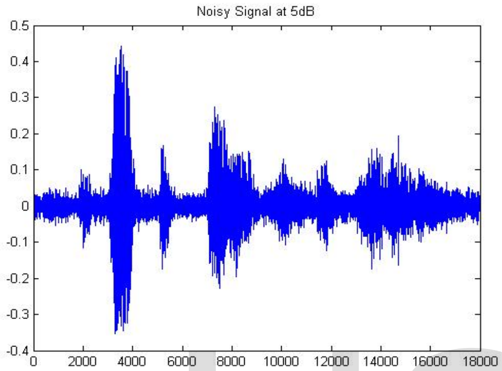

Fig.6.Speech signal which is recorded far from microphone

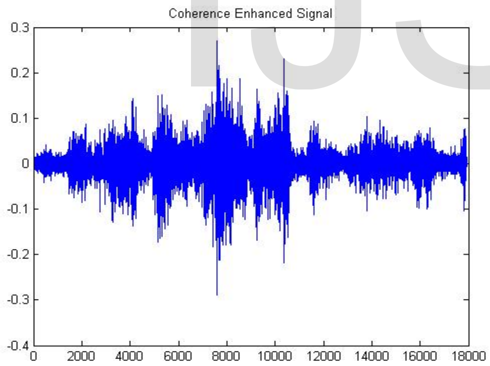

Fig.7.Coherence enhanced signal

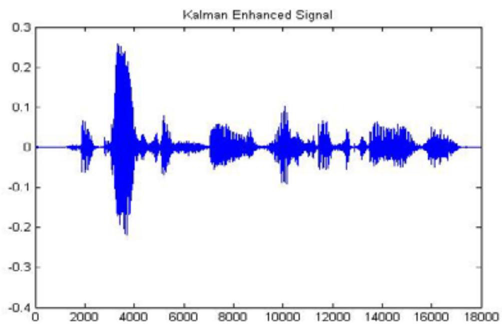

Fig.8.Enhanced signal using Kalman Filter

4.2 PSNR and PESQ values for Kalman filter

Peak signal to noise ratio and the Perceptual evaluation of speech quality measures for the existing algorithm is

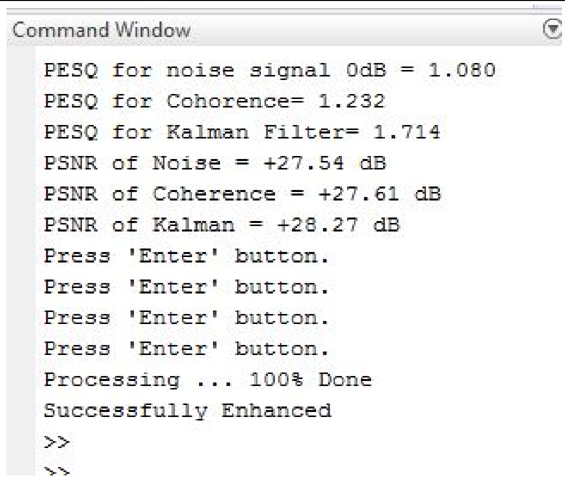

Fig.9.Psnr and Pesq measurements for original signal, coherence and Kalman Filter

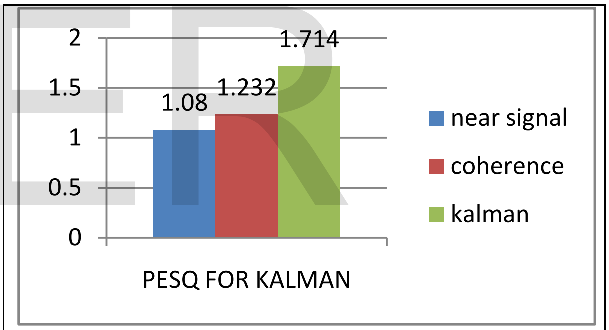

Fig.10. PESQ values of Kalman Filter

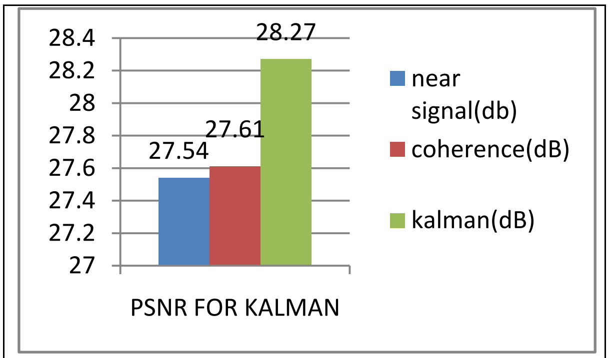

Fig.11. PSNR values of Kalman Filter

4.3 Wiener Results

Work extended by using wiener filter for

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Obtained better speech signal compared to Kalman filter.

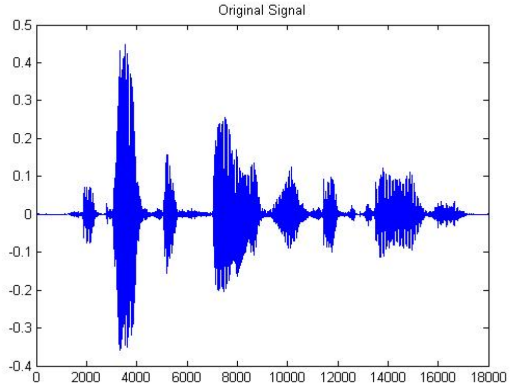

Fig.12.Clean speech signal

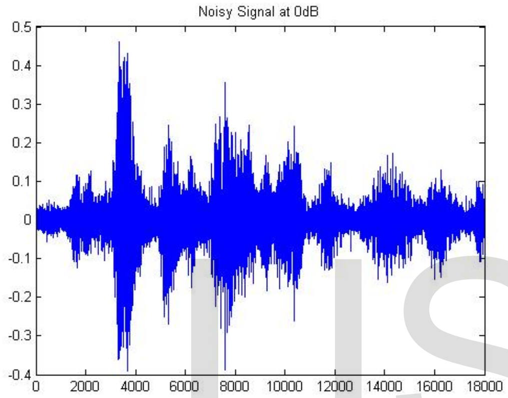

Fig.13.Speech signal which is recorded near to microphone

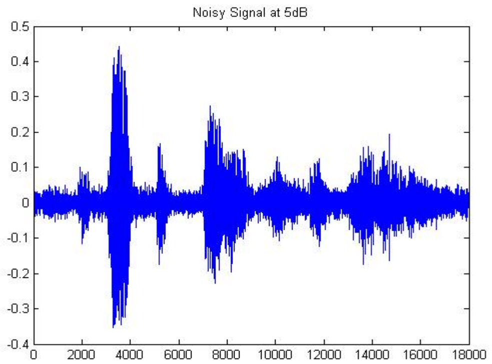

Fig.14.Speech signal which is recorded far from microphone

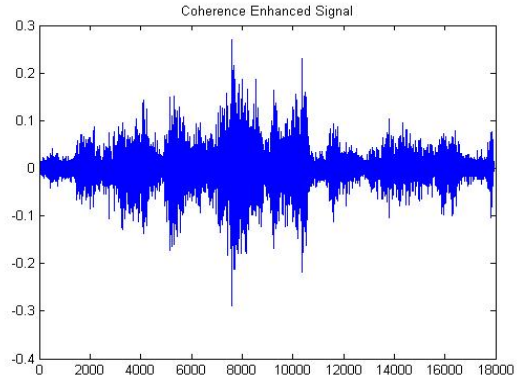

Fig.15.Coherence enhanced signal Microphone

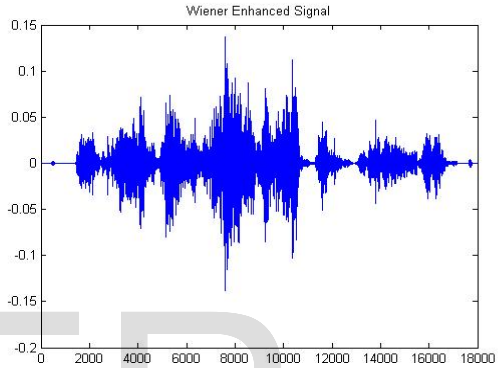

Fig.16.Enhanced signal using wiener filter

4.4 PSNR and PESQ values for Wiener Filter

The below one is output obtained for wiener filter by psnr and pesq values which are run in matab2014.a.

Command Window

```
PESQ for noise signal 0dB = 1.080
PESQ for Cohorence= 1.232
PESQ for Wiener= 3.044
PSNR of Noise = +27.54 dB
PSNR of Coherence  = +27.61 dB
PSNR of Wiener = +30.35 dB
Press 'Enter' button.
Press 'Enter' button.
Press 'Enter' button.
Press 'Enter' button.
Processing ... 100% Done
Successfully Enhanced
>>
>>
```

Fig.17.PSNR and PESQ measurements for original signal, coherence and wiener filter.

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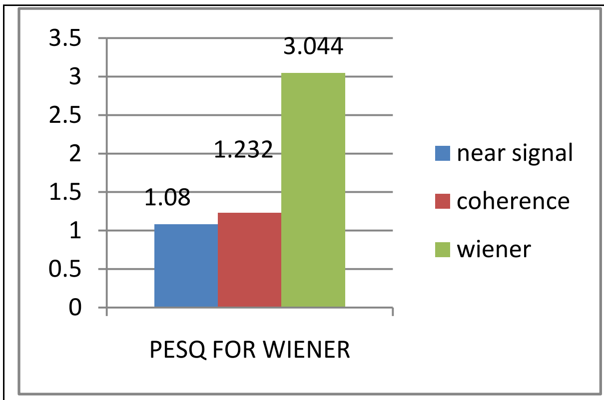

Fig.18.PESQ values for Wiener Filter

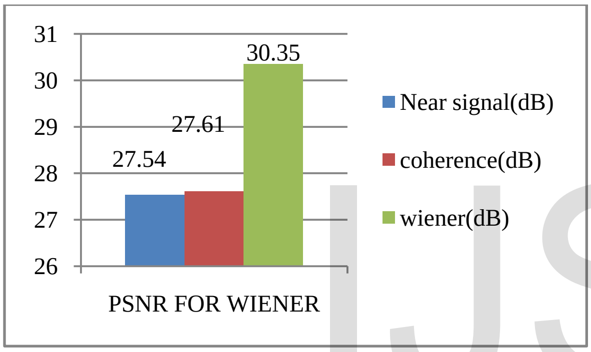

Fig.19.PSNR values for Wiener Filter

The comparison between the Kalman and wiener filter by using the Perceptual Evaluation of Speech Quality and peak-signal to noise ratio values for 30 individual signals are taken restaurant noise at (odb,5db,clear signal).

TABLE 1
Psnr and Pesq measures for the restaurant noise at 0dB, 5dB signal using Kalman and Wiener filter

Name of the signal	Kalman filter		Weiner filter	
	PSNR	PESQ	PSNR	PESQ
1. sp01 2. sp01_ restaurant_sn0 3. sp01_restaurant_sn5	+29.27 dB	1.482	+29.40 dB	2.522
4. sp02 5. sp02_ restaurant_sn0 6. sp02_ restaurant_sn5	+28.79 dB	1.118	+29.87dB	2.004
7. sp03 8. sp03_ restaurant_sn0 9. sp03_ restaurant_sn5	+29.03 dB	1871	+28.89dB	2.343
10. sp04 11. sp04_ restaurant_sn0 12. sp04_ restaurant_sn5	+28.04 dB	1.871	+29.55dB	2.542
13. sp05 14. sp05 _restaurant_sn0 15. sp05_ restaurant_sn5	+28.72 dB	1.434	+27.56dB	2.205
16. sp06 17. sp06_ restaurant_sn0 18. sp06_ restaurant_sn5	+28.67 dB	1.354	+28.67 dB	2.474
19. sp07 20. sp07_ restaurant_sn0 21. sp07_ restaurant_sn5	+27.49 dB	1.420	+27.49 dB	2.598
22. sp08 23. sp08_ restaurant_sn0 24. sp08_ restaurant_sn5	+27.91 dB	1.074	+27.91 dB	2.229
25. sp09 26. sp09_ restaurant_sn0 27. sp09_ restaurant_sn5	+28.40 dB	1.398	+28.40 dB	2.838
28. sp10 29. sp10_ restaurant_sn0 30. sp10_ restaurant_sn5	+28.40 dB	1.329	+28.40 dB	2.911

4 COMPARISON GRAPHS

Comparison graphs between the methods ofIdentification rate comparison of different

ISSN 2229-5518

processing method of peak signal to noise ratio and mean square error comparison by Spectral subtraction, Normalised least mean squares filter, Recursive least squares, affine projection algorithm, General Kalman Filter for (Speaker Identification) [1] and the speech enhancement using adaptive filters of Kalman and wiener filter by Psnr and Pesq measures.

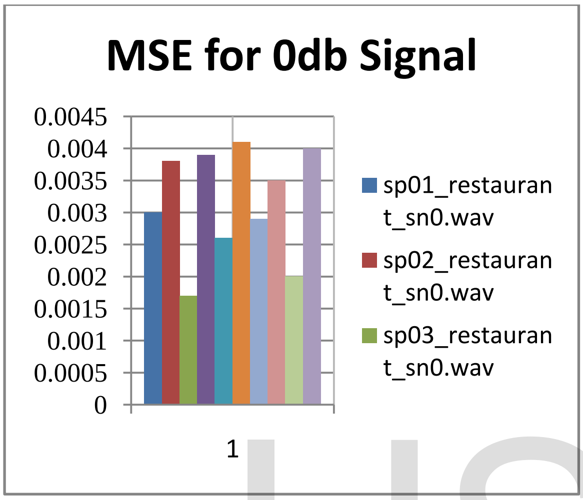

Fig.20.Mean Square error for the restaurant noise at 0dB

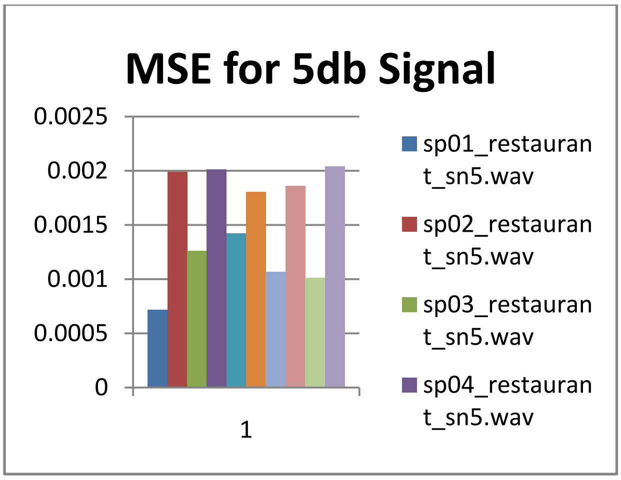

Fig.21 Mean square error for the restaurant noise at 5dB

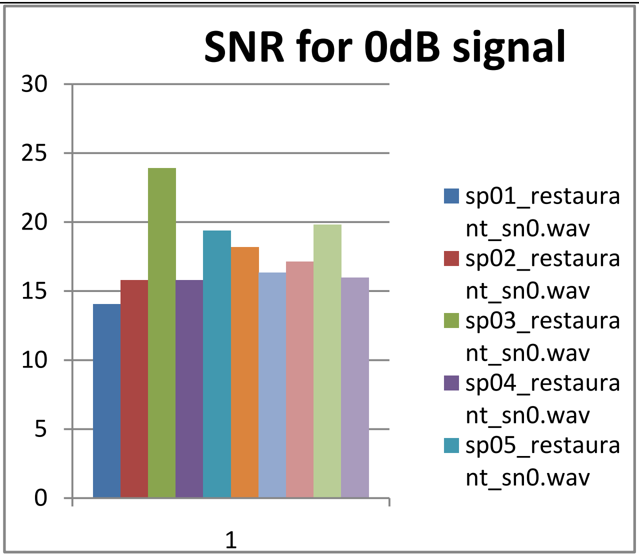

Fig.22.Signal to noise ratio for the restaurant noise at 0dB

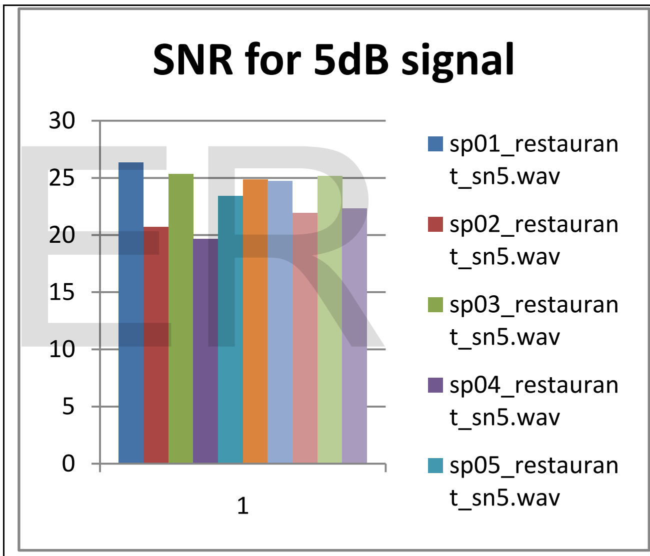

Fig.23.Signal to noise ratio for the restaurant noise at 5dB

6 CONCLUSION

ISSN 2229-5518

The project gives us an improved dual-microphone speech enhancement algorithm for mobile communications. In this system the distance between the two microphones are very much placed closely. So the proposed technique can be used in small mobile phones. First, we depicted the approximate coherence function used in this algorithm. Then, we explained the determined method replacing the Kalman filter with wiener filter. The results obtained with the help of wienerfilter are good compared to the Kalman filter results. Finally, the ability of implementation andintelligibility advantage make this method a good way for future commercial mobile phones.

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