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Speech Enhancement by a Microphone Array Using Various Modelling Approach

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ABSTRACT

Speech enhancement is one of the most important topics in speech signal processing. The main difficulty towards enhancement is different environmental conditions as well as speech enhancement is processed in real time basis. In this paper a new approach towards speech enhancement is proposed using EM algorithm and Hidden markov model (HMM) model. This unique approach gives information about statistical structure of speech signal using speech model. In this paper, we present a new technique to improve the peak signal-to-noise ratio in the enhanced speech signal by using an adaptive implementation of the Wiener filter. For this paper, a comparative performance analysis of lower values of noise signal and higher values of noise signal has been estimated. Four performance criteria are utilised in this study, peak signal to noise ratio, MMSE, delay in filtering, convergence rate. Here combination of EM algorithm and HMM model are used for good quality of clean signal. There is a strong reason why Hidden Markov Model is used, the model is very rich in mathematical structure and it is based on probabilistic approach, hence can form the theoretical basis for use in a wide range of applications. By using this model results also shows that for lower values of noise signal PSNR is in the range of 70 db and for higher values of noise signal PSNR value decreases linearly. Other performance parameter varies with respect to each other.

Keywords: AWF (adaptive wiener filter) EM algorithm, HMM Model, PSNR, MMSE.

INTRODUCTION

The main purpose of Speech enhancement is to improve speech quality by using different algorithms or methods. The quality of signal comprises clarity and intelligibility, pleasantness, compatibility with some other speech processing techniques. The basic methods for enhancing speech are the removal of background noise, echo suppression and artificially bringing certain range of frequencies into the speech signal. In this paper our main focus is to remove the background noise and to get clean audible signal as output. Whenever suppression of background noise is done, it is very important that original speech signal not badly harm. During active noise suppression method the delay of processing should be very small to avoid producing extra noise instead of cancelling existing noise[5]. To avoid delay most of the methods for active noise suppression are completely analog. Analog to digital and digital to analog conversions produces some amount of delay during operation, which is harmful for speech enhancement process .Most of the speech enhancement techniques are based on removal of background noise, if background noise is stationary then it is very easy to estimate value of noise signal. The degradation of the speech signal because of background noise is a severe problem in speech related systems and therefore it should be eliminated through speech enhancement algorithms. microphone array is a series of microphone or more than two microphone placed at same time to capture the sound for speech enhancement. The benefit of using series of microphone in a room is that it allows the algorithm to determine the position of sound in the room. The position of sound is find out by comparing the arrival times of sound for each microphone[3]. Our main goal behind speech enhancement process is Enhance quality and intelligibility of original speech signal.Speech enhancement aims to improve speech quality by using various methods like spectral subtraction, noise cancellation, array processing and others[11][7]. The objective of enhancement is improvement in intelligibility and overall perceptual quality of degraded speech signal using audio signal processing techniques. In this particular paper our focus is on the removal of background noise by using EM

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algorithm and HMM model. By using this unique approach, the main objective is speech enhancement of original spoken signal using HMM model, speech enhancement by removing the noise signal by using EM algorithm, Performance evaluation of speech enhancement with the help of microphone array. More focus is to improve PSNR value, ideally it should be infinite. The speech enhancement is strongly dependent on real time conditions because when speaker is close to microphone, the noise levels as well as reverberation effects are very small. As a result different signal processing technique gives better performance. In another case if the distance between speaker and microphone increases, the distortion of speech signal increases which results large amount of noise & reverberation effects are more which comes to PSNR value is less.

The traditional methods are used earlier for speech enhancement such as spectral subtraction, noise cancellation, array processing etc. These methods typically exploits just second order statistics and ignoring higher order statistics. They implicitly make a Gaussian assumption on a speech signal that are highly non Gaussian. The major issue is that above methods typically disregard information on the statistical structure of speech signal. Apart from above methods, some of these methods suffer from lack of principled framework. These methods sometimes used in ad-hoc solutions also[1][7]. To remove all the drawbacks of above traditional methods , a new approach of speech enhancement has been used using EM algorithm & HMM model. In this paper EM algorithm follow from taking a probabilistic modelling approach to problem. The mixture model commonly known as hidden markov model (HMM) is trained offline on a large dataset of clean speech. The main operation of this paper is that input is given to combination of HMM-EM model, consider that model as model 1.Another input is given to combination of EM-HMM model, so called as model 2.T he output of both the model is filter through the adaptive wiener filter, so we get the resultant value of speech signal in the form of PSNR. The resultant speech signal is known as Enhanced speech or clean speech. The resulting model is known as hidden variable model, in this original speech signal and speech states are hidden, which is unobserved to user. So in this hidden variable model, a suitable EM algorithm estimates from data the noise parameter and original speech signal, so that noiseless clear speech signal is obtain as output[2]. This idea is very much useful in everyday life where the speech signal is distorted by noise & reverberation effects. This is an efficient approximation scheme that reduces computational complexity for noise case only. In this, EM algorithm is known as EXPECTED MAXIMIZATION. EM algorithm is used where E step updates the sufficient statistics, which gives speech signal estimator and M step updates the parameter of noise signal & reverberation filters[3]. Hidden markov model (HMM's) are a formal foundation for making probabilistic model of linear sequence labeling problems. They provide conceptual toolkit for building complex models just by drawing a statistical picture. HMM are based on computational sequence analysis. The operation of HMM is that it is generating a sequence and every sequence visit a state, then emit a residue from the states emission probability distribution method ,the HMM model choose next state to visit next according to states transition probability distribution. So it is completely based on computational sequence of states[10][8].

In this paper, we proposed a new technique to improve the peak value signal to noise ratio in the enhanced speech signal by using an adaptive implementation of the Wiener filter. The implementation of wiener filter is perform in time domain, because of varying nature of the speech signal. And comparative analysis between the lower values of noise level and higher values of noise level has been verified with respect to other performance parameter.

LITERATURE SURVEY

From last three decades, many researchers have been worked on speech enhancement, but problem remain unsolved. Many good techniques used application in various field like communication, speech recognition, biomedical application and user interfaces. The major difficulty of enhancement task is depends on environmental conditions and real time scenario. Already traditional methods are used for speech enhancement, such as a spectral subtraction, noise cancellation & array processing[4] .In previous techniques, Active noise suppression method is used to remove extra surrounding noise. The idea behind is to produce anti-noise signal into the listener's ear to cancel the noise which is somewhat impractical. The computation time commonly known as delay, should have very small value to avoid producing more noise instead of cancelling the existing noise. Because of Delay factor, most of the methods for active noise suppression are based fully analog .Analog to digital and Digital to analog transforms inevitably produce some amount of delay during speech processing. All the above methods have satisfactory results. But these are not used for advanced implementation in speech enhancement technology. These traditional methods have many drawbacks, like some of these method used just second order statistics for sequence modelling And Noise estimating parameter ignores higher order statistics during operation. Some of these methods are suffer from lack of principled framework. Many times these methods are used in ad-hoc solutions, which is not permanent for long time. This is used to remove noise with algorithm that handle reverberation problem in a systematic manner[8]

Historically, the general EM-algorithm is credited to Scientist Dempster, Laird and Rubin, who proved in their 1977 paper, among other things, that the algorithm leads to a sequence of parameters with monotonely increasing likelihood values of there sequence. They also coined the term "EM-algorithm". Interestingly, the EM-algorithm for HMM model was described already in 1970 by Scientist Baum and Welch. And is also often referred to as the *Baum-Welch* algorithm in the previous HMM literature[9]. EM is an algorithm for estimation of unknown parameters, like noise and signal parameters. In HMM model, Viterbi is the algorithm for computing the most probable sequence of hidden states. You would, indeed, use EM for HMMs for parameter estimation. Hidden Markov Models (HMMs) and the Baum–Welch algorithm was first described in a series of articles by scientist Leonard E. Baum .It is developed in the Institute for purpose of Defence Analysis in the late 1960s. At that time HMM was mostly used for speech enhancement purpose only.

Basic Model of Speech Enhancement System

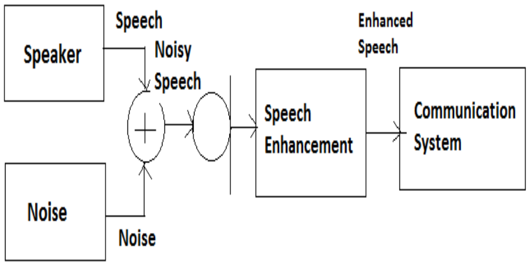


Fig1. *Basic steps to follow in speech enhancement*

OBJECTIVE OF PROJECT

The main objective of this paper is

1) Speech enhancement of original spoken signal using HMM model.

2) Speech enhancement by removing the noise signal by using EM algorithm and adaptive wiener filter approach.

3) Performance evaluation of speech enhancement with the help of microphone array. The main aim is to improve speech quality by removing background noise & echo suppression. More focus is on getting highest PSNR value and analysis of performance parameters like convergence rate, filter length, MMSE, Delay in operation.

Proposed Model for Speech Enhancement

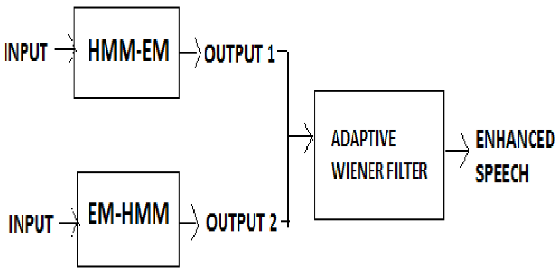


Fig2. *Basic idea behind this speech enhancement*

Basic operation of this model is that we are taking two signals as an input, firstly one is a speech signal which is free from noise and another one is noise signal which is very small in magnitude as compare to original speech signal. So consider as a INPUT and given to combination block of HMM-EM. The original speech signal and noise signal are hidden. When we correlate HMM model and EM algorithm together, then by using EM algorithm separate out original speech signal and noise parameter. The unknown parameters are carried out by E-step and M-step respectively. The resulting output is OUTPUT 1, same input signal is pass through another combination block of EM-HMM so resulting output is OUTPUT 2.Both the OUTPUT 1 and OUTPUT 2 is given to adaptive wiener filter block, so filtration of original speech signal is process at this block and resultant value of speech signal is taken out as a output, so we can call it as enhance speech signal which is our final output. In this particular technique, peak signal to noise ratio is higher, so higher value of PSNR is better for clear reception of original spoken signal.

EM ALGORITHM

This paper presents an EM algorithm for speech enhancement. That estimates the filter parameters H_m^i [K]and the noise spectra $B^i[k]$ from the data Y. The main purpose of this algorithm is to computes the sufficient statistics (SS) which is require for speech enhancement and the speech signal estimator used here as $X_m[k]$. The main function of the E-step is to find out the conditional distribution over the unobserved variables X,and S is given the observed ones Y , which is denoted by p (X ; S |Y).

This distribution of sequence is termed as a posterior distribution, and it can be obtained from the complete data distribution of speech signal using Bayes rule[2][12]. It is nothing but the posterior distribution form which derives the sufficient statistics (SS) which is useful for removal of noise. The main difficulty having to sum over the C^M configurations of component labels which called as S=(S_0------S_{M-1}), where C is number of speech model component and M is the number of frames. For at least 100 component speech model leads to gives good performance. And for short filters, the relative window length M=1,2 so exact summation is possible. If we consider in a realistic scenario the value of M≥5, which require summation of at least 10^{10} configuration.

In this EM algorithm, the function of E-step is to update the sufficient statistics which include speech signal estimator and M-step updates parameter of noise signal . The EM algorithm is nothing but EXPECTED MAXIMIZATION (EM).Now illustrates each step individually.[1]

M-Step

The function of M step is to update the parameter of noise signal and reverberation filter.

$$\sum H_n^i[K]\lambda_{m-n}[K] = n_m^i[K] \tag{1}$$

For $H_n^i\ [k]$, this can be done using sub band FFT as follows. For each sub band k, define M-point FFT as $H_m^i\ [k]$by

$$H^i[K,l] = \sum_{m=0}^{M-1} e^{-jwtm} H_m^i\ [K] \tag{2}$$

Where ωt=2Π l/M is the frequencies, where l = (0:M-1)

The sub band FFT are λ [K,l] and η[K,l] are defined in same manner .After this we get

$$H^i[K,l] = \frac{n^i[K,l]}{\lambda\ [K,l]} \tag{3}$$

E –Step

The function of E step is to update the parameter of sufficient statistics, which contain speech signal estimator.[2]

$$\sum B^i[K] H_{n-m}^i k]^* (Y_n^i[K] - \sum_{r \neq m} H_{n-r}^i[K] X_r) = v_{sm}[K]\, \rho_{sm}[k], \tag{4}$$

Where the variance are given by

$$v_{sm}[K] = \sum B^i[K]\, |H_{n-m}^i[k]|^2 + A_s[K]. \tag{5}$$

Finally update the parameter for probabilities and expressed γ_{sm} in logarithmic term express as

Log γ_{sm} = $\sum$ (v_{sm}[K] ρ_{sm}[K]² +log$\frac{A_s[K]}{v_{sm}[K]}$) + logΠ_s . (6)

The E-step equations are solved iteratively, so that value of $\rho_{sm}[K]$ and γ_{sm} are calculated.

PROBABILISTIC SIGNAL MODEL

In this particular method, HMM model uses probabilistic approach. In this model consider a complex variable Z, we define a Gaussian distribution with mean μ and precision is defined as the inverse variance ν by

$$\rho(z) = N(z|u,v) = \frac{v}{\pi} \exp(-v|z-\mu|^2) \quad (7)$$

As it is viewed as joint distribution over Real z & Imaginary z, ρ(z) integrates to one, which satisfies the equation.

$$E(z)=\mu, E(|z|^2) = |\mu|^2+\frac{1}{v} \quad (8)$$

here operator E denotes averaging.

When building a statistical model of sub band signal, the model have to ignore real valued sub bands like K=0, N/2 and give focus on complex variables. Here model define a complex variable (N/2-1) dimension vectorX_m containing all sub bands of frame m.[1]

X_m= (X_m[1],------X_m[N/2-1])

Here for K>$\frac{N}{2}$,X_m[k]=X_m[N-K]* . Here we also let X[k] denote sub band k of all frames and let X denotes all sub bands of all the frames.

X[k]={X_m[k],m=(0: M -1)},

X={X_m[k],k=(0: N -1),m=(0 : M-1)} (9)

So by using above probabilistic method HMM model works for enhancement of speech signal[4].

HIDDEN MARKOV MODEL

Hidden markov model (HMM's) are a formal foundation for making probabilistic model of linear sequence 'labeling' problems. It is based on predefined toolkit for constructing a complex model just by drawing an imaginary picture which gives step by step sequence generation. They are use for a diverse range of program including Gene finding in biomedical field, Profile searches, multiple sequence alignments during generation, Regulatory site identification.[8]

HMM are the Lego's of computational sequence analysis Hidden Markov models having close connection with mixture model. An HMM is generating a sequence. The first step is to visit a state, then emit a residue from the states emission probability distribution theory. We choose next state to visit next according to states transition probability distribution.

The model thus generating two strings of information.

1) The underlying state path, as we transition from one state to another.

2) The observed sequence where each residue being emitted from one state path to another state path.

The state path is consider as a markov chain, in this generation of next step is completely depend upon previous state. The output of this operation is an observed sequence, and the resulting underlying state path is hidden over here. The state path is hidden markov chain[10]. An HMM is a full probabilistic model..Model parameters & the overall sequence 'scores' are all probabilities. For that it is use Bayesian probability theory to manipulate these numbers in standard and powerful ways. It includes optimizing parameter of noise and speech signal.

To Find Out the Best State Path

In this paper observing a sequence, & between this sequence to find out the hidden state path. There are potentially many state paths that can generate same sequence, but It is necessary to find out the one with highest probability. In HMM model, the efficient viterbi algorithm is used, and it is guaranteed to find the most probable state path given a sequence .

An HMM means specifying four things.

The symbol alphabet, k different symbol. (e.g. a c g t,k=4).

The no. of states in model M.

Emission probabilities ei(x)=1

Transition probability ti(j)=1.

Markov Generation Model

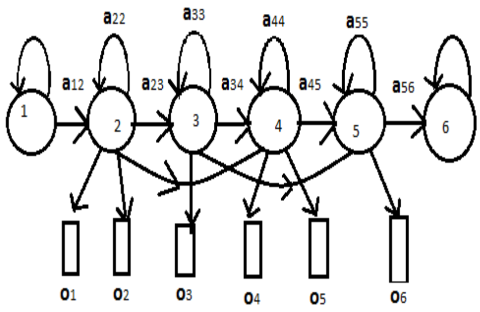


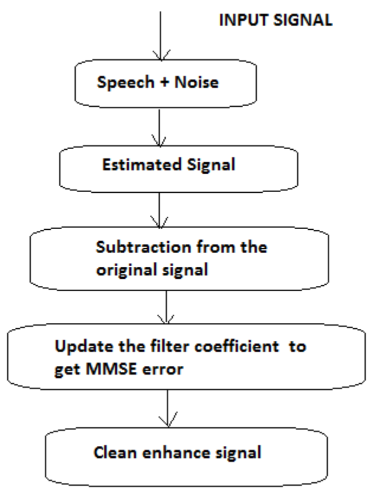
Fig3. *Hiddem markov model generating a sequence*

In this particular model the sequence of observed speech vectors, which is nothing but word spoken by user is generated by a Markov model [9]. The working principle of Markov model is just like a finite state machine which changes its state once every time unit. Consider at each time t that a state j is entering, a speech vector denoted by o (t) is generated from the probability density bj o (t). The transitions are also probabilistic and are governed by ai j.

ADAPTIVE WIENER FILTER APPROACH

Filter is a vital component or a device that process the signal in order to extract information about a prescribed quantity of interest. During speech enhancement of original spoken signal, filter works as a finishing agent. In this particular method we are using adaptive wiener filter, because of his superiority over another filters. When useful signal and noise signal occupy two separate frequency bands then ordinary FIR and IIR filter are suitable for lower values. Background noise is mostly variation in nature and frequency content of noise signal varies with time[4]. Adaptive filter can help in those situations where useful signal and noise signal overlap on each other and variation of noise signal with respect to time. An adaptive wiener filter is a digital filter with self adjusting characteristics. It adopts automatically with changes to the input signal. In most of the time, adaptive wiener filter works as a noise canceller for speech enhancement process.[6]

Flowchart for Adaptive Wiener Filter



Working Model of Adaptive Wiener Filter

Speech enhancement by using Adaptive wiener filter is one of the popular techniques. The basic principle of adaptive wiener filter is to get clean signal which is corrupted by background noise. It is necessary to estimate an optimal filter for noisy input speech by minimizing the mean square error(MMSE) between the desired signal and estimated signal respectively[5][6].Two input signals y_k and x_k are applied simultaneously to the adaptive filter. The signal y_k is the contaminated signal containing both the desired signal,s_k and the noise signal n_k, assume uncorrelated with each other. Signal x_k is a measure of the contaminating signal which is correlated in some way with n_k.signal x_k is processed by the digital filter to produce an estimate of n_k which is nothing but $\hat{n}_k$ An estimate of desired signal is then obtained by subtracting the digital filter output $\hat{n}_k$ from the contaminated signal y_k .and the equation is given by

$$\hat{s}_k = y_k - n_k = s_k + n_k - \hat{n}_k \tag{10}$$

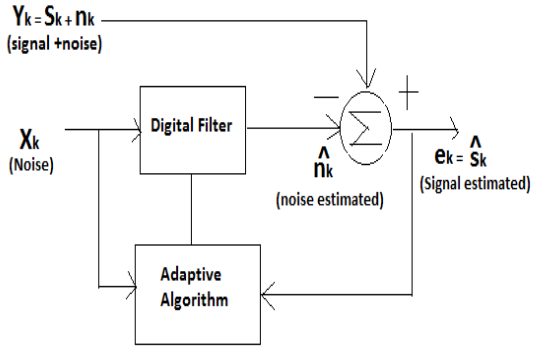


Fig4. *Block diagram of adaptive wiener filter.*

The main objective of adaptive wiener filter is to produce an optimum estimate of the noise in the contaminated signals and hence an optimum estimates of the signal. This is achieved by using $\hat{s}_k$ in a feedback arrangement to adjust the digital filter coefficients, using a suitable adaptive algorithm to minimize the noise signal. The output signal serves two purposes, first it is an estimate of desired signal and secondly As an error signal which is used to adjust the filter coefficients. Adaptive algorithms are used to adjust the coefficients of the digital filter such that the error signal e_k is minimized according to some criterion. Mostly the algorithms use are least mean square(LMS),recursive least square(RLS),kalman algorithm etc. In terms of computation and storage requirements, the LMS algorithm is the most efficient[6]. Further it does not suffer from numerical instability problem inherent in other two algorithms.LMS algorithm is the first choice in many applications.

In the above diagram the error signal e_k between the wiener filter output and the primary signal is given by

$$ek = yk - \hat{n}k = yk - = y_k - W^T X_k = y_k - \sum W(i)\, x_{k-i} \tag{11}$$

Where X_k and W are the input signal vector and weight vector respectively.

The square of the error is given by

$$e^2_k = y^2_k - 2y_x\ X^T_K W + W^T X_K\ X^T_K\ W \tag{12}$$

The mean square error (MSE) J is obtained by taking the expectations of the square of the error. Assuming that the input vector x_k and the signal y_k are jointly stationary J is given by equation,

$$J = E\,[e^2_k] = E[y^2_k] - 2E[y_k\ X^T_K W] + E[W^T X_k\ X^T_K W] \tag{13}$$

$$J = \sigma^2 + 2P^T W + W^T R W \tag{14}$$

where E symbolizes expectation

σ^2 is the variance of, P is the N length cross correlation vector. R is the N*N auto correlation matrix.

Each set of coefficients *w (i) (i=0,1,….,N-1)*, corresponds to a point of the surface, the gradient is zero and the filter weight vector has its optimum value Wopt when $W_{OPT} = R^{-1}P$ This is known as Wiener-Hopf equation. Hence the task in adaptive filtering is to adjust the filter weights W(0),W(1),…….., using a suitable algorithm, to find the optimum point on the performance surface.[14]

OTHER PERFORMANCE PARAMETERS

Convergence Rate

One of the important parameter for this model is the convergence parameter. Convergence rate is related to stability of the model for different noise level inputs. In this it determines the rate at which the adaptive filter converges its property to its resultant state. For such kind of model convergence rate should be faster. There will be a trade off in other performance criteria. For some cases, the system getting an improved convergence rate and at the same time other performance parameter will get decrease and vice versa .So convergence rate is inter dependable to others. Sometimes it may happens that if the convergence rate is increase, the stability of particular model will get decrease. So during system development make sure that every performance parameter is in proper range [15]. The model is based on real time basis so different performance parameter should be justice properly with their respective value. In this paper we are getting average value of convergence rate is 2.96, which is one of good range.

Minimum Mean Square Error (MMSE)

Another important parameter for this model is minimum mean square error (MMSE). In signal processing, a minimum mean square error (MMSE) estimator is an estimation method which minimizes the mean square error (MSE) of the fitted values of a dependent variable, which is a common measure of estimator quality. Minimum mean square error is indicating the system adaptability to words any value of input .So it in our hand to control the value of MMSE. And its value should be minimum .Because of low value of MMSE; the errors during operation are getting reduced at certain level. For accurate modelling of system MMSE value is always less.If any user getting high value of MMSE so it is an indication that system having some complex error, and it should be remove by developing accurate adaptive model[16]. For finding MMSE value a Bayesian estimation approach is used and Bayesian estimation approach also deals with where sequence of observation is independent.

Filter Length

In signal processing many of the performance parameter is directly dependable to others. The filter length of the adaptive system is one of important factor. The length of the filter indicates that efficient accuracy of a design model. So for better result, in this paper we maintain filter length unity. In most of the cases the filter length affects the convergence rate, by variation in computation time, also affect stability of system., At certain levels, it affects the minimum MSE also. For any system ,if the filter length is increase , the number of computations during operation also increase, which results into increase complexity and vice versa. For particular system, if there are too many poles and zeros then there may be increase computational complexity. So filter length is very sensitive for other parameter, it should be select very carefully.

Peak Signal to Noise Ratio (PSNR)

In this paper the most important parameter is peak signal to noise ratio, commonly called as PSNR. In Signal to noise ratio it compares the level of a desired spoken signal to the level of background noise or surrounding noise. Peak Signal to noise ratio is defined as the power ratio between a signal and the background noise[13] .PSNR is most important performance parameter in speech enhancement Field .It is measured in dB and it is represented as PSNR. Ideally PSNR should be infinite. The Signal to Noise ratio can be represented by formula 20log10(S/N) db. In this paper we are getting average value of PSNR in the range of 70 db respectively. From the graph And observation table it is clearly mention that as noise level increases, PSNR value decreases and vice versa. noise level and PSNR ratio is reciprocal to each other.

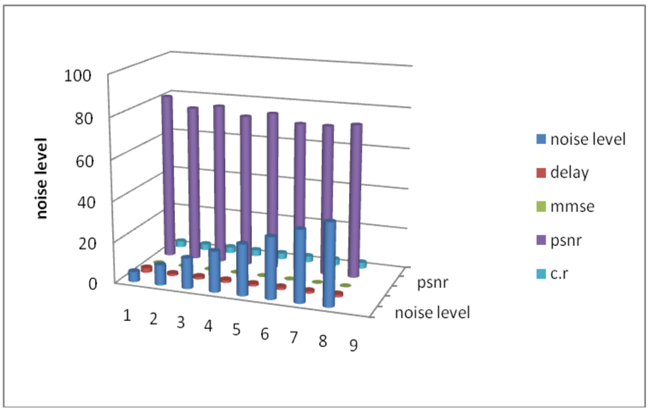
Experimental results

The success of EM algorithm and HMM model is determine by the success of individual test. Here we are comparing lower values of noise level to the peak signal to noise ratio. At the same time comparison of higher noise level to the peak signal to noise ratio. After adding different noise with original speech signal we are getting different values of parameter like Delay in filtering, Convergence rate, minimum mean square error(MMSE),Filter length, PSNR. In this paper our main focus is to get improve value of PSNR . Ideally PSNR value should be infinite, but because of some background noise and real time scenario, we are getting in the range of 70db.From below table, it is notice that for lower value of noise level we are getting higher value of PSNR and as noise increases PSNR value decreases. Noise level and speech signal are reciprocal to each other.

Table1. *Different values of lower noise level to the peak signal to noise ratio.*

Noise level (db)	Delay in filtering	PSNR(db)	MMSE	Convergence Rate
5	2.29	81.60	0.00045	2.98
10	1.15	76.74	0.00137	2.98
15	1.31	78.66	0.00088	2.97
20	1.32	74.68	0.0022	2.96
25	1.16	77.10	0.00126	2.96
30	1.28	73.14	0.00315	2.96
35	1.15	73.21	0.00310	2.96
40	1.29	73.08	0.00213	2.96

Graphical Representation

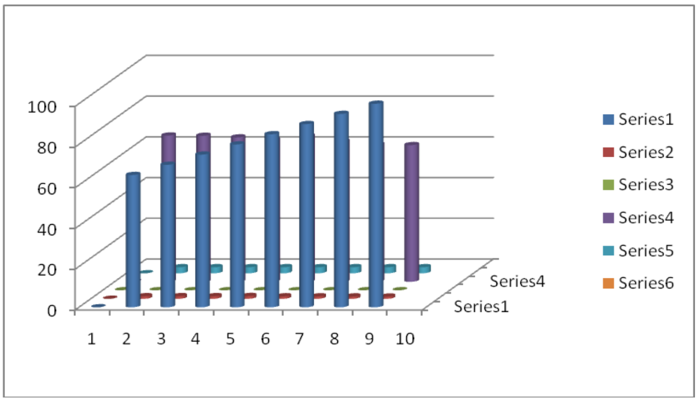


Graph1. *all four parameters corresponding to lower noise level.*

Table2. *Different values of higher noise level to peak signal to noise ratio.*

Noise level (db)	Delay in filtering	PSNR(db)	MMSE	Convergence Rate
65	1.28	71.81	0.00428	2.98
70	1.30	71.70	0.00438	2.98
75	1.28	70.92	0.00526	2.97
80	1.36	69.70	0.00696	2.96
85	1.22	71.44	0.00467	2.97
90	1.21	68.97	0.00824	2.96
95	1.15	67.32	0.01204	2.96
100	1.09	67.13	0.00737	2.94

Graphical Representation



Graph2. *all four parameters corresponding to Higher noise level.*

CONCLUSION

In this project results are getting to enhance speech which is free from noise. The main aim of this paper is to get the improved value of SNR which we are getting in the range of 70 db. Other performance parameters are also observed like delay in processing, convergence rate, filter length,MMSE etc .so we are getting better result. From above tables it is notice that for lower values of noise signal, PSNR value is higher .As well as during processing delay in operation is less, which reduces surrounding noise and gives better value of PSNR. Filter length maintains to unity to get higher level of enhance signal and convergence rate is also small. From above graph it is clear that noise signal and original speech signal is inversely proportional to each other. Now a day's enhance speech is having lots of applications like source coding for mobile phone, video conferencing, IP phone etc. It is also used in biomedical application, automatic speech recognition, speaker identification, Hearing aids, Hands free human machine interface and Cockpits and noisy manufacturing. So all the above applications are of day to day life. Speech enhancement is one of the important fields, and aim is user get speech signal which is Noise free and easily audible. The hidden Markov model has been studied thoroughly together with the EM algorithm for signal processing of speech signals. By using adaptive wiener filter in time domain the results are good. In recent years, there are many improvements in speech processing field, because of wide application in day to day life of human being. The future scope of this model and EM algorithm is online version of this is use for non stationary noisy signal.

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