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Acoustic Echo Cancellation of Speech signal using Average Recursive Least Square Adaptive Algorithm

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Abstract: In this high-tech environment, the use of digital communication systems like hands-free systems for delivering speech in the classroom or in an auditorium are becoming more popular. In such situations, the loudspeaker and a high-gain microphone are commonly found. But, the presence of a large acoustic coupling between loudspeaker and the microphone would produce a loud echo and the background noise that reduces the quality of speech and results in inconvenience to the listeners. The solution to these problems is the elimination of the echo using an echo cancellation algorithm. The basic component of an echo canceller is an adaptive filter which creates a replica of the echo and subtracts it from the combination of the actual echo and the near-end signal. In this paper, the speech signal in an auditorium environment has been enhanced by eliminating the echo and noise signal by adopting an adaptive filter algorithm called Average RLS and using Band pass filter in pre-processing step. Finally for evaluating the performance of the algorithm, the parameters such as Mean Square Error (MSE) and ERLE (Error rate loss enhancement) are measured.

Keywords: Echo-Cancellation algorithm, Adaptive filters algorithm, Average RLS, Mean Square Error (MSE) and ERLE (Error rate loss enhancement).

I. INTRODUCTION

In this global communication arena, wireless phones and Hands-free communication devices are referred as an essential communication tools and it is utilized for people's day-to-day personal works and for business communications purpose. Use of such electronic communication device undoubtedly experiences an echo. This is nothing but hearing the voice coming back through the device, sometimes with delay caused due to various couplings and/or interactions between two physical phenomena all along the speech transmission chain. This problem can lead to simply annoying or to absolutely unbearable. Thus controlling or eliminating echo becomes a major consideration in telecom applications.

The phenomenon of echo can be defined as delay and distorted version of the speech signal. Echoes of speech are received as they are reflected from the floor, walls and other surrounding objects. After reflecting if the signal arrives within a short time then it is specified as spectral distortion or reverberation similarly if the reflected signal arrives after tens of milliseconds then it is referred as echo.

End-to-End delayand Round trip delay are two important factors used in echo. End-to-End delay is the time between the generation of the sound at one end and its receiver at the other end. Round trip delay, which is the time taken to geta reflected copy of the signal (echo), is approximately twice the end-to-end delay.

Echoes become disturbing when the round trip delay exceeds 30 milliseconds. Such an echo will be clear. When round trip delay exceeds 30 milliseconds and echo strength exceeds 30 dB, echoes become steadily more disruptive.

There are two types of echoes in communication network. Hybrid echo is originated within the telecommunications network. This echo is the result of impedance mismatch in the analog local loop. And the development of hands-free communications gave rise to another kind of echo known as an acoustic echo. The acoustic echo is due to the coupling between the loudspeaker and microphones.

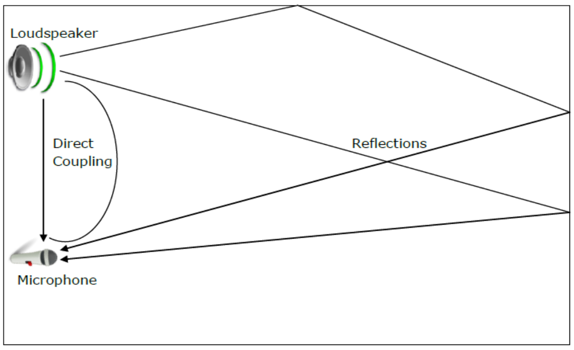

Figure 1: Sources of acoustic echo in a room

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In this paper, our main objective is acoustic echo cancellation. The sources which participate in acoustic echo are depicted in Figure 1 such as Microphone, Loudspeaker, Wall, Roof etc., for reflections. Cancellation of acoustic echo is a challenging problem for the following main reasons:

- The impulse response of the acoustic echo path is several times longer, between 100 to 500 msec.
- Due to opening and closing of a door or movement of people inside the room, the characteristics of the acoustic echo path are non-stationary.
- The acoustic echo path has a mixture of linear and nonlinear characteristics.
- The reflection of acoustic signals inside a room is almost linearly distorted.
- However, the loudspeaker does introduce nonlinearity. The main causes of this nonlinearity are the suspension nonlinearity that affects distortion at low frequency and homogeneity of flux density that produces nonlinear distortion at large input signal levels.

From the early 1960s majority of echo cancellation methods were developed by AT&T Bell Labs and later by COMSAT Tele-Systems. In echo cancellation, algorithmic procedures are computed where it involves generating the sum from reflected echoes of the actual speech, then subtracting it from any other signals picked by the microphone.

Figure 2 shows the basic operation of an echo canceller. When the far end signal or received signal is passed through loudspeaker or any hands-free telephony application, the microphone used can also pick up the near-end signal along with the echo signal of the output and reflections created by room acoustics. The goal is to remove the echo. To achieve this different adaptive algorithm is adopted in this paper.

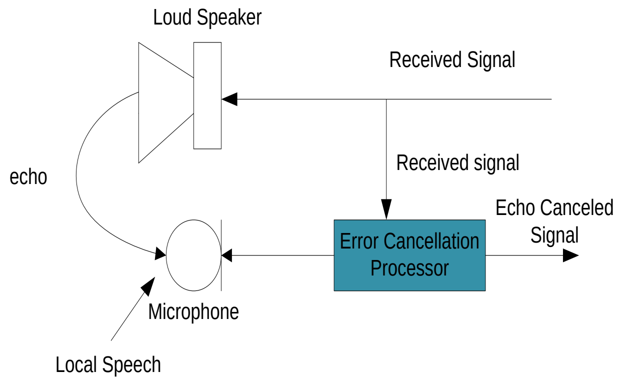

Figure 2: Basic operation of an echo cancellation

Each adaptive algorithm is made up of filters known as adaptive filters, which is one of the core technologies in digital signal processing and finds many applications such as echo cancellation, channel equalization and adaptive noise cancellation. An adaptive filter is made up of digital filter and adaptive algorithm. The task of an adaptive filter is to operate suitably in an unknown environment and track time variation of input statistics. The adaptive filter evaluates the echo signal and the evaluated signal is removed from the original signal thus generating an error signal. This error signal calculated is passed back into the adaptive filter and its coefficients are changed algorithmically in order to minimize the cost function. In case of echo cancellation, the optimal output of the adaptive filter is equal to the unwanted echoed signal. There are many recursive algorithms for adopting in adaptation of adaptive filtering. The choice of algorithm is determined by the performances like rate of convergence, miss-adjustment, tracking, robustness, computational complexity, structure. Adaptive algorithms are Least MeanSquare (LMS) and Recursive Least mean Square (RLS) and Frequency Domain Adaptive Filter (FDAF) etc.
Related works has been carried out by many authors using different adaptive filter techniques and different approaches for echo reduction and also noise removal from input speech signal were designed by the author Upal Mahbub et.al [1].They used a recorded echo in the form of corrupted data and proposed a practical approach for single channel acoustic echo cancellation. Their method was developed using gradient based adaptive LMS algorithm.It showed that the proposed offline AEC algorithm has the ability to provide a satisfactory performance on cancelling echo for recorded echo corrupted speech in terms of average ERLE (dB) and SDR improvement (dB) under various acoustic room environments. Mugdha. M. et.al [2], worked on reducing the additive noise from the speech signal by using an adaptive NLMS algorithm but step size, filter lengthand a ratio of energy spectral density (ESD) of speech is kept constant and algorithm used a noise signal as a reference for updating weights of adaptive filter. Developed algorithm resulted for improved performance by measuring performance analysis factors and can be concluded that proposed algorithm outperforms LMS and NLMS algorithms in terms of SNR, MSE and convergence time.Sabri et.al [3] has designed a new framework for AEC system which has the ability to handle the mismatch between the sampling rate for the input signals and appropriate generation of a balanced sampling rate output.
A novel method by Subhash C et.al [4] has developed for both acoustic echo and noise cancellation in ageneralized sidelobe canceller framework. The primary contribution of this work is the development of multichannel adaptive Kalman filter (MCAKF) in a modified generalized sidelobe canceller (MGSC) framework. Additionally, in this work both the near end speech signal and noise is assumed to be unknown. In the proposed method speech acquired by a microphone array is subject to adaptive beamforming using MVDR method. On the other hand a blocking matrix filter is used to attenuate the near end speech signal while passing both the noise and residual echo. A MCAKF is developed in this context that also estimates the noise andresidual echo. Hence, a difference of MCAKF output and the adaptive beamformer (ABF)

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output gives an estimate of the near end speech signal. The performance of proposed method is evaluated using subjective and objective measures on the ARCTIC database. Distant speech recognition experiments are also conducted on the ARCTIC database. The proposed method gives reasonable improvements both in terms of perceptual evaluation and distant speech recognition. By referring to the existing methods described by different authors on acoustic echo cancellation and by studying different concepts on room acoustics, the nature of echo, background noise and characteristics of speech signal, in this paper we are designing a framework for echo cancellation and noise reduction method from the speaker speech signal in an auditorium environment by incorporating adoptive filter and algorithm.

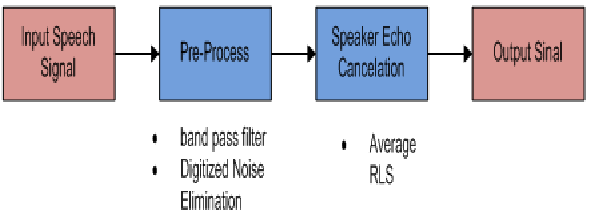

Figure 3: Block-Diagram for Speaker Echo cancellation

II. METHODOLOGY

This section briefly explains our proposed method and Figure 3, shows the block diagram of our proposed speaker side echo cancellation. The proposed system mainly made up of two modules namely: pre-processing and echo cancellation module. As the initial step, A Band pass filter limits the frequencies and eliminates the signal which passes out of range that can be treated as noises i.e., frequencies that are out of range of human voice. To remove the noise within the range of speech signal, Gaussian filter method is adopted. The second step is echo cancellation using adaptive algorithm. The Average Recursive least square (RLS) algorithm is implemented and the resultant output signal is made free from echo and noise. This section briefly explains each module of our proposed system.

A. Band Pass Filter

Band pass filter is a device with a combination of two types of filters low-pass filter and high-pass filter. Band pass filter filters the input signal to fall within the specified range and rejects the signal frequencies fall outside the range. The main task of a band-pass filter is to reduce the noises that are the frequencies which lie outside of range with respect to human voice.A band pass filter is a circuit or an electronic device, made up of low-pass and high-pass filters and commonly found in the wireless transmitter and receiver. The main function of a band-pass filter is to limit the frequency of signal to the minimum bandwidth necessary to be heard by the humans i.e., which lie in between 20Hz and 20 kHz. And preventing the signals which are out of range, that are considered as noise can be removed.

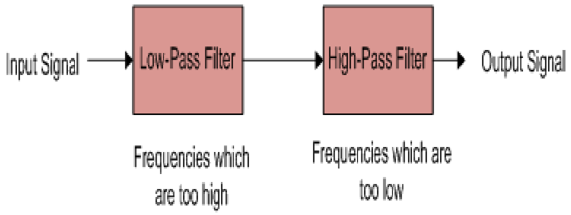

Figure 4: Block-Diagram of Band-pass filter

After applying the band pass filter, the frequencies which lie outside the hearing frequency is attenuated, but the noise frequencies falling within the frequency range of 20Hz and 20 kHz still exists. This need to be removed by using another noise elimination technique called Gaussian method.

B. Gaussian Noise Elimination Technique

Gaussian Filter method named after famous scientist called Carl Gauss, one of the popularly used filtering approaches in signal processing and image processing to achieve smoothing and de-noising. Gaussian Filter is a type of windowed filter and convolution-based filter that uses a Gaussian matrix as its underlying kernel. By its nature it is weighted mean filter. Following steps are used to calculate:

Given window size 2N+1 calculate support points

$$x_n = \frac{3n}{N}, n = -N, -N+1, \dots\dots, N;$$

Step 1 Calculate values G(n)

$$G(n) = \begin{cases} \frac{1}{\sqrt{2\pi\sigma}} e^{-(\pi^2/2\sigma^2)}, & |n| \leq N_0 \\ 0, & |n| > N_0 \end{cases} \quad (1)$$

Step 2 Scale factor is calculated using

$$k = \sum G(n) \quad (2)$$

Step 3 Calculate window weights

$$G'(n) = \frac{G(n)}{k} \quad (3)$$

Step 4 For each signal element:

a. Initially a window is placed on it.
b. Elements are picked up;
c. Window weights of each element is multiplied;
d. Sum up products will be the new filtered value.

C. Adaptive Filters

In other words, a filter is a device that maps its input signal into another output signal by extracting only the desired information contained in the input signal. The aim of an adaptive filter is to calculate the difference between the desired signal and the adaptive filter output, e (n).The error signal is fed back into the adaptive filter and its coefficients are changed algorithmically in order to minimize a function of this difference, which is known as

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the cost function. In the case of acoustic echo cancellation, the optimal output of the adaptive filter is equal in value to the unwanted echoed signal. When the adaptive filter output is equal to desired signal the error signal goes to zero. In the situation the echoed signal would be completely cancelled and the far user wouldnt hear any of their original speech returned to them.

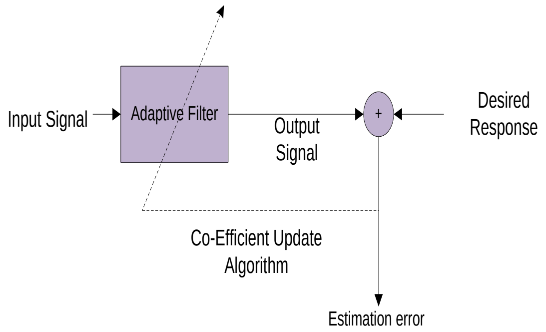

Figure 5: Block-Diagram of Adaptive filters

D. Recursive Least Squares Filter

RLS filter is a simple type of adaptive filters and it is similar to wiener filter with time update version. The Recursive Least Squares (RLS) algorithm is implemented based on least squares method. It is a mathematical procedure finds the best fitting curve; this is achieved by reducing the sum of the squares of the offsets of the points from the curve.

The RLS algorithm recursively solves the least squares problem. In the following equations few parameters and terminologies are used such as λ and δ represents the forgetting factor and regularization. Where the forgetting factor is a constant and less than unity and the value of regularization are determined by the signal-to-noise ratio (SNR) of the signals.

The weight vector $w^{\hat{}}$ represents the adaptive filter's weight vector and the M-by-M matrix P is referred to as the inverse correlation matrix. In this algorithm w (n) updates continuously with each set of new data without solving matrix inversion. The two main parameters of the algorithm are: first, matrix inversion calculation is only required for derivation and not used for implementation, therefore reduces the complexity of the algorithm. Second, required variables are each time updated within the loop of iteration and use the value of previous iterations. Figure 5 shows the flowchart of the steps involved in the computation of RLS.

Step 1 The output of filter is measured using the filter tap weights from the previous iteration and the current input vector.

$$\bar{y}_{n-1}(n) = \bar{w}^{t}(n-1)x(n) \quad (4)$$

Step 2 The intermediate gain vector is calculated using eq. (2).

$$u(n) = \bar{\Psi}_{\lambda}^{-1}(n-1)X(n) \qquad (5)$$

$$k(n) = \frac{1}{\lambda + X^{T(n)u(n)}}u(n) \quad (6)$$

Step 3 The estimation error value is calculated using the following equation:

$$\bar{e}_{n-1}(n) = d(n) - \bar{y}_{n-1}(n) \qquad (7)$$

Step 4 The filter tap weight vector is updated using eq. (8) and the gain vector is calculated in eq. (2).

$$w(n) = \bar{w}^{T}(n-1) + k(n)\bar{e}_{n-1} \qquad (8)$$

Step 5 The inverse matrix is calculated using eq. (8).

$$\psi_{\lambda}^{-1}(n) = \lambda^{-1}(\psi_{\lambda}^{-1}(n-1) - k(n)[X^{T}(n)\psi_{\lambda}^{-1}(n-1)]) \qquad (9)$$

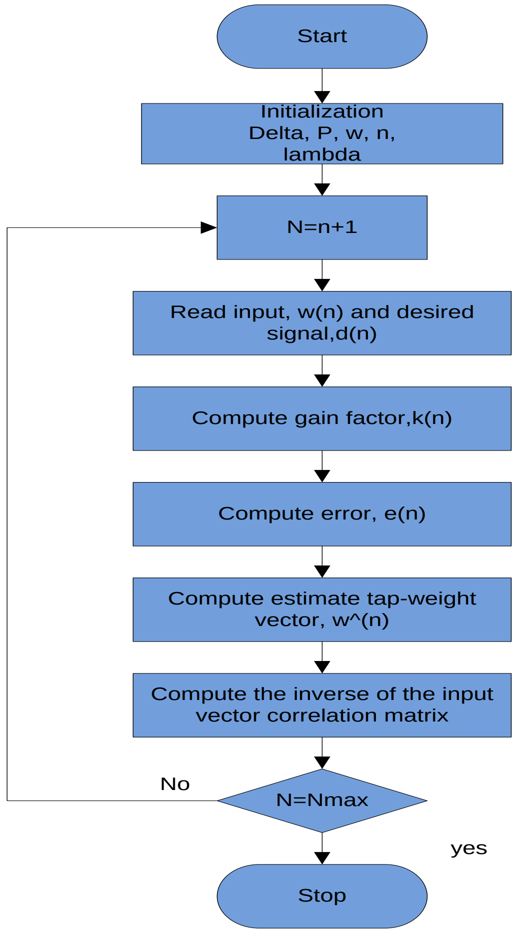

Figure 6: The flowchart of the RLS algorithm

E. Average Recursive Least Squares Filter

The block-diagram of average RLS algorithm is as shown in Figure 7, where adjustable filter (i) and adaptive filter (j) denotes minimum and maximum value of forgetting factor values. The adaptive filters process the input signal mixed with the echo and reverberation and produces the estimated echo. The algorithm selects the combined estimated echo of the both adaptive filer (i) and adaptive filter (j) which, leads to reduce the error value. Using these two systems in parallel, based on forgetting factor by applying (10), (11) & (12) equations.

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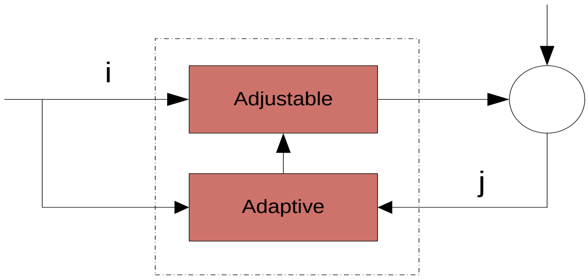

Figure 7: Block Diagram of Average RLS algorithm

$$\psi_{\lambda_1}^{-1}(n) = \lambda_1^{-1}\big(\psi_{\lambda_1}^{-1}(n-1) - k(n)[X^T(n)\psi_1^{-1}(n-1)]\big) \quad (10)$$

$$\psi_{\lambda_2}^{-1}(n) = \lambda_1^{-1}\big(\psi_{\lambda_2}^{-1}(n-1) - k(n)[X^T(n)\psi_2^{-1}(n-1)]\big) \quad (11)$$

Calculate Average filter value for RLS filter:

$$\psi_{\lambda_{avg}}^{-1}(n) = Avg(\psi_{\lambda_1}^{-1}(n), \psi_{\lambda_2}^{-1}(n)) \quad (12)$$

III. RESULTS AND DISCUSSION

Simulation results for echo cancellation using average RLS algorithm for input signal speech is explained in this section. For input signal as a pre-processing step the Band pass filter is applied and the noise outside the range of hearing threshold is removed. Figure 8(a) shows the input speech signal of our proposed system, which need to be process and improve the quality of speech. Figure 8(b) echo signal added for input speech. After applying the Average RLS algorithm Figure 8(d) shows the estimated error calculated. By subtracting the error curve with the input signal the desired output is as shown in Figure 8(c).

The various performance parameters are used for evaluating the performance of AEC methods are as follow:

A. Mean Square Error(MSE)

MSE is a measure of the sequence of mean squared error. The expected value of square of error will be the result of MSE. At each time instant of an adapter, MSE predicts themean-square error. The lower the value of MSE value is favourable. It is calculated using the following equation:

$$MSE(n) = \frac{1}{n}\sum_{i=1}^{n}(y(n) - e(n))^2 \quad (13)$$

B. Error Rate Loss Enhancement (ERLE)

ERLE measures the attenuation of the echo signals in an acoustic echo cancellation system. It is expressed using dB. Higher ERLE corresponds to higher reduction in echo. It is calculated using the following equation:

$$ERLE(n) = 10\log_{10}\frac{y^2(n)}{e^2(n)}\ dB \quad (14)$$

Using the values from the existing methods we are comparing with proposed average RLS algorithm as shown in the table below.

Table 1: COMPARISION TABLE

Methods	Performance Measures	
	MSE (dB)	ERLE (dB)
LMS	0.038	2
NLMS	0.014	12
AverageRLS	**0.0129**	**18.44**

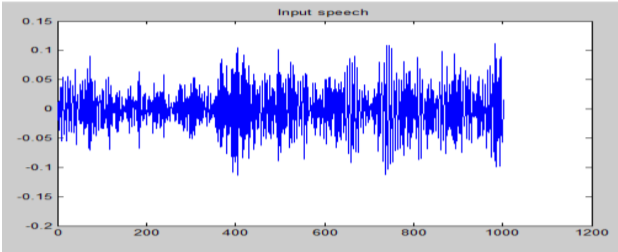

Figure 8 (a): Input Speech signal

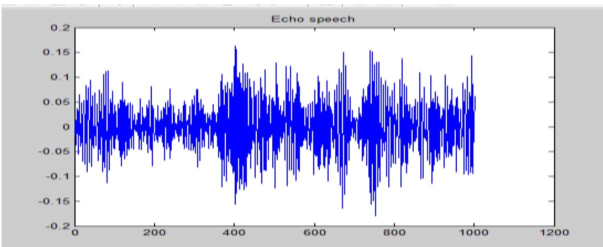

Figure 8 (b): Echo Signal

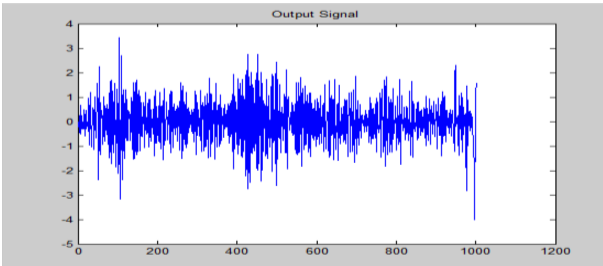

Figure 8(c): Output Signal

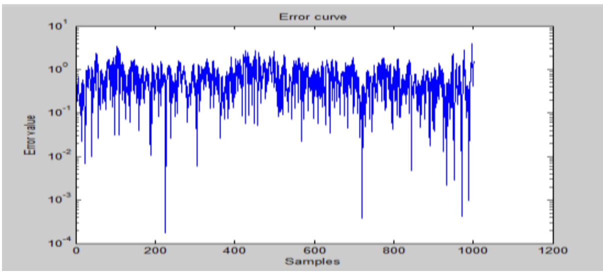

Figure 8(d): Estimated error curve

IV. CONCLUSION

In hands-free communication devices, echo occurs due to the open-air path or coupling between the loudspeaker and microphone which results in degradation of quality of speech. In order to enhance the speech signal, a novel scheme is designed where for the input speech signal significant amount of echo and noise signals are added and using the band pass filternoise in the input speech has

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been reduced and by adopting the adaptive filters called average RLS algorithm acoustic echo is minimized. Finally, the system parameters are analyzed and the gain of the signal is calculated and the error performance is also calculated using Echo Return Loss Enhancement (ERLE) and Mean Squared Error (MSE), which represents the amount of echo signal removed and also the residual error signal for overall cancellation of acoustic echo from the input speech. The resultant value of our proposed algorithm shows the higher value for ERLE results with higher rate of convergence and lower complexity and MSE shows the higher rate which depicts less amount of error in the input signal.

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BIOGRAPHIES

Mr. Praveen N. was born in Hosanagar, Shimoga district, India. He graduated in Electrical & Electronics Engineering from UBDT College of Engineering, Davanagere, affiliated to Kuvempu University, Karnataka, India, in the year 1996 and completed his M.Tech (Power & Energy Systems) from Karnataka Regional Engineering College (NITk), Surathkal, affiliated to Mangalore University, Karnataka, India in the year 1999 and since then he is actively involved in teaching undergraduate students. He is having 13 years of experience in teaching and has an additional M.Tech (Digital Electronics) to his credit. He is currently working towards Ph.D. degree in signal processing at Bangalore Institute of Technology, Bangalore-04. His research interests include analysis and modelling of acoustic signals, speech enhancement,noise estimation, adaptive filtering, Image processing and Power system.

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