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Short-term uncleaned signal to noise threshold ratio based end-to-end time domain speech enhancement in digital hearing aids

Padmaja Nimmagadda[1], Kondru Ayyappa Swamy[2,3], Samuda Prathima[4], Sushma Chintha[5], Zachariah Callottu Alex[6]
[1]Center for Communications and Signal Processing, Department of Electronics and Communication Engineering, Sree Vidyanikethan Engineering College, Tirupati, India
[2]Department of Electronics and Communication Engineering, Aditya Engineering College, Surampalem, India
[3]Department of Electronics and Communication Engineering, Jawaharlal Nehru Technological University Kakinada, Kakinada, India
[4]Department of Electronics and Communication Engineering, RMK College of Engineering and Technology, Thiruvallur, India
[5]Department of Electronics and Communication Engineering, Jawaharlal Nehru Technological University Anantapur, Ananthapuramu, India
[6]School of Electronics Engineering, Vellore Institute of Technology, Vellore, India

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ABSTRACT

This paper presents the improvements in the combined solution for the noise estimation and the speech enhancement in digital hearing aids in time domain. This study focuses on the single channel statistical temporal speech enhancement using adaptive Wiener filtering. In this technique, the noise is updated based on the short-term uncleaned signal to noise threshold ratio (ST-USNTR) of the frame. It works best if and only if the background noise level is low compared to that of speech of interest. We considered the time domain algorithms in order to consider the time varying nature of speech signal. The performance of the proposed algorithm is evaluated for speech signal with seven types of noises and three signal to noise ratios (SNR) levels in each type of noise. From the results, it is clear that the basic level of adaptive speech enhancement is obtained using statistical parameters of noisy speech without the need for reference input.

Corresponding Author:

Samuda Prathima
Department of Electronics and Communication Engineering,
RMK College of Engineering and Technology
Thiruvallur, Tamil Nadu, India
Email: samudaprathimaece@rmkcet.ac.in

1. INTRODUCTION

Performance of most of the systems with speech applications depends on the quality of the input signal. Applications like hearing aids [1], [2], specially designed for the people with sensorineural hearing loss needs high SNR than that of people with normal hearing to have same level of intelligibility. However, looking into real world scenarios consisting of background noises interferes the intended signal and also degrades the device performance. Therefore, to have better performance of the device, there is a need for enhancement of the signal of interest. To perform the signal enhancement there, exist a vast variety of algorithms and methodologies. Now, the problem arises with the selection of an algorithm and the methodology that best performs for that particular application. The below are the few criteria's those influence the selection of speech enhancement algorithm for particular application.

Type of application is the first criterion: coding or speech to speech applications. Coding applications of speech, requires complete noise elimination and won't bother of speech distortions, i.e., they need high SNR with tolerate intelligibility. Maximum of speech-to-speech applications, need high quality speech, i.e., higher SNR along with the good quality. Second criterion is based on position of the application: device is fixed or portable. If the device is fixed in a known place for example room, car etc, it can be easy to estimate the noise, which simplifies the enhancement of speech. Third one is based on number of microphones used in the device. Using array of directional microphones, it is easy to eliminate noise if the direction of intended speech or the direction of unknown noise. The solution becomes critical with single microphone device where there is no reference input for noise or desired signal. In this type of devices, we must estimate the noise from the given noisy input and then calculate the appropriate gain to enhance the noisy speech. Along with the above criteria noise spectrum also influences the selection of algorithm [3], [4]. If noise is broadband or a high frequency noise, then we can easily suppress it by passing it through a filter. Here the problem will arise when the noise spectrum exists within the spectrum of intended speech. Finally, it becomes a challenge to enhance speech signal with low-frequency noise using single channel.

Throughout the process of speech enhancement there are two parameters to balance, the amount of noise reduction and the amount of speech distortion i.e., the SNR and the intelligibility of the enhanced speech signal. So, it is noted that having higher SNR is important without loss in intelligibility in enhanced speech. Chen *et al.* [5] explained it in three ways of the algorithm with a priori knowledge of the signal, or with an array of microphones or by proper changes in the Wiener filtering. In this research, the third option Wiener filtering have been followed for better management of speech distortion.

In literature, most of the voice activity detection (VAD) techniques for speech enhancement have been implemented in the spectral domain [6]–[10]. The accuracy of VAD is an important factor in this and it is based on the decision rule employed in it. Many approaches have been proposed in spectral domain to increase the accuracy of detection. Few of the techniques based on the long-term spectral features like, long term spectral flatness measure (LTSF) [6], long term spectral divergence (LTSD) [7], and long term signal variability (LTSV) [8] between speech and noise.

The rest of the techniques presented in literature are based on short term features both in the time domain or in the frequency domain. Sohn *et al.* [9] proposed VAD that employs the decision-directed parameter estimation method for the likelihood ratio test. Jo *et al.* [10] proposed the VAD that employs support vector machine (SVM) for decision function using the LRs, where as in the conventional techniques perform VAD by comparing the geometric mean of the LRs with a given threshold value. Shin *et al.* [11] presented a VAD based on conditional maximum a posteriori probability (MAP) criterion that exploits the voice activity decision of the previous frame along with that of current frame in the estimation of probability of voice presence in the current frame. It outperforms the conventional VAD by using two separate thresholds for the likelihood ratio test (LRT), which are resulted from temporal correlations between current and previous frames. Upadhyay *et al.* [12] proposes the recursive noise estimation algorithm, in this the noise power is updated based on present and previous values of it with the help of a smoothing parameter. It depends on the filter transfer function from sample to sample based on the speech signal statistics; the local mean and the local variance and it is implemented in frequency domain. The proposed algorithm is similar to the work done in [12], but the major difference exists in the domain of computations, the above existed work done in spectral domain and the proposed work completely done in time domain.

Xiao *et al.* [13] developed time domain speech enhancement using generative adversarial network (GAN) to improve the performance of the generator and also compared difference GANs available for speech enhancement. Tan *et al.* [14] proposed an end-to-end multi task model for VAD which increases the robustness of VAD system for low SNR conditions. Tejaswini*et al.* [15] compared the different approaches available both in frequency domain and time domain. They have discussed the steps involved in speech enhancement using MATLAB. They have also explained the mathematical operations involved in Fourier Transform, windowing, averaging, finding variance and minimum mean square error. Zhao *et al.* [16] projected the noisy speech into speech dominated subspace and noise dominated subspace and fed to encoders to detect the speech and noise separately.

In this present research, the noise estimation and the corresponding gain calculations are performed in time domain. Both the time domain and frequency domain techniques have their own advantages and disadvantages [17]. Since it is complicated to implement transformation techniques in digital hearing aids, a simple and effective algorithm is developed for noise estimation and speech enhancement in time domain for low-cost applications in real time. Majorly the proposed time domain technique has three advantages: i) accommodates the time varying nature of the speech signal [18], ii) reduces the number of computations [19], and iii) avoids unpleasant signal distortion which exist in spectral domain techniques because of invalid short-term fourier transform (STFT) [20].

This paper is organized as follows. In first section, we presented the proposed algorithm. In second section, we have explained the proposed block diagram. In third section, results are discussed. The conclusions are given in the last section.

2. PROPOSED VAD BASED ON ST-USNTR

In speech processing applications, the conventional VAD is used to separate incoming signal into voiced, un-voiced and silence frames. The procedure of VAD slightly changes in speech enhancement applications, since the input signal is corrupted with the background noise. So, it is important to choose feature that separates the incoming speech frames well in background noise. To have the accuracy of operation, the proposed VAD separates the incoming noisy speech into 3 different group of frames based on the ST-USNTR, which is defined as the ratio between the short-term temporal energy (STTE) of incoming noisy signal and the noise threshold (NT). The update of NT for noise frames with STTE ≤NT i.e., silence frames, speech frames with dominated noise STTE≤1.5×NT and speech dominated frames is given in (1).

$$NT = \{NT - avg, USNTR \leq 0dBNT - smooth, 0dB < USNTR \leq 0.176dBNT - previous, 0.176dB < USNTR \tag{1}$$

This type of division is important to update the noise even if the VAD detects noise dominated speech frames and to completely suppress noise without distortion in speech. The flow chart of proposed algorithm is shown in the Figure 1. First, the initial 3 frames were averaged to have the reference for noise threshold. The frames with STTE less than or equals to this threshold are considered as noisy frames. Therefore, frames with the negative USNTR values are applied with a zero gain. Also, the average estimation of the noise threshold is done for this type of frames. Next, the frames with STTE greater than the threshold are considered as high USNTR frames. For this category of frames, the Wiener gain (G) from [21], [22] is applied with the help of the mean, the variance of the noisy speech signal. The relation between Wiener gain (G) and threshold USNTR is given in (2).

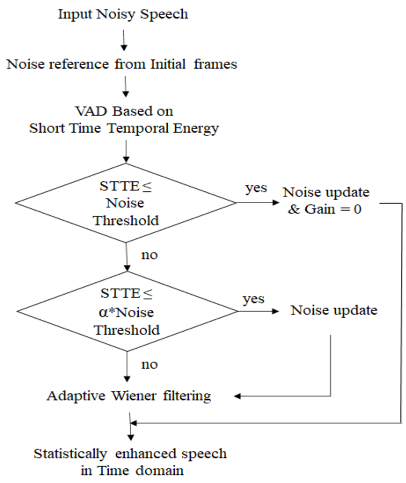

Figure 1. VAD based on ST-USNTR

$$Gi = \{0, USNTR \leq 0dBG, dB < USNTR \tag{2}$$

In this case also noise is estimated but the difference is here we used smoothed estimation of noise instead of average estimation. At the last, the frames with higher USNTR are applied with high gain values calculated by using same Wiener filtering. Here there is no option for estimation of noise threshold, therefore the previous estimation of it is used in gain calculations as in (3).

$$\begin{aligned} NT\ Current &= 0.5 \times NT\ Estimated + 0.5 \times NT\ Previous, USNTR \leq 0dB \\ NT\ Current &= 0.3 \times NT\ Estimated + 0.7 \times NT\ Previous, USNTR \leq 0.176dB \\ NT\ Current &= NT\ Previous, 0.176dB < USNTR \end{aligned} \tag{3}$$

2.1. Block diagram of proposed work

The block diagram of proposed end-to-end time domain single channel speech enhancement is shown in Figure 2. The time domain analysis and synthesis of speech into frames is done using overlap buffering and addition. In this work, the STTE value of each frame compared with NT in the time domain. The frames with ST-USNR less than or equals to 0.176 dB are assumed as noise dominated frames, applied with a zero gain. Next, for remaining frames the Wiener filter based on first order statistics is used to calculate the gain to obtain linear estimation [10] of original clean speech and also as it minimizes the mean squared error between the clean speech and enhanced speech.

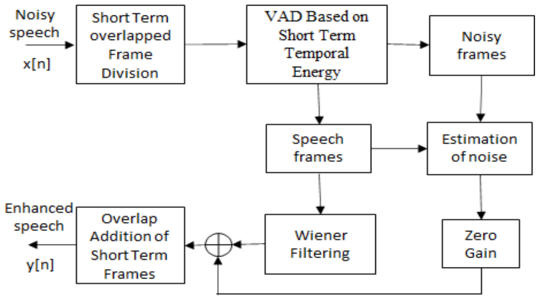

Figure 2. Block diagram of time domain single channel adaptive Wiener filtering

3. RESULTS AND DISCUSSION

The approach for proposed speech enhancement algorithm is completely different from existing algorithms. It is completely based on the noise statistics and the noisy input in time domain. The simulations are done in MATLAB software using the noisy speech signals taken from noisy speech corpus (NOIZEUS) database [23]. Experiments were conducted for one clean speech utterance: “I am the small, salt and tasty”, added at three different SNRs 0dB, 5dB and 10dB with seven different types of noises one at a time. Among them six are different real-world noises are taken from AURORA database namely, train, station, restaurant, car, airport and babble noises and the other are the most common AWGN noise. The input noisy speech from the database is divided into 10 msec frames i.e., 80 samples per frame using Hanning window with 50% of overlapping. First three incoming noisy frames were averaged to have the reference for noise threshold (NT). Then the incoming noisy input frames are grouped into three different categories based on the adaptive noise threshold and each group is applied with a corresponding gain as explained in the above sections. The Figure 3 shows that the STTE of estimated noise adaptively changes with that of incoming noisy frames and the noisy (silence) frames are suppressed completely.

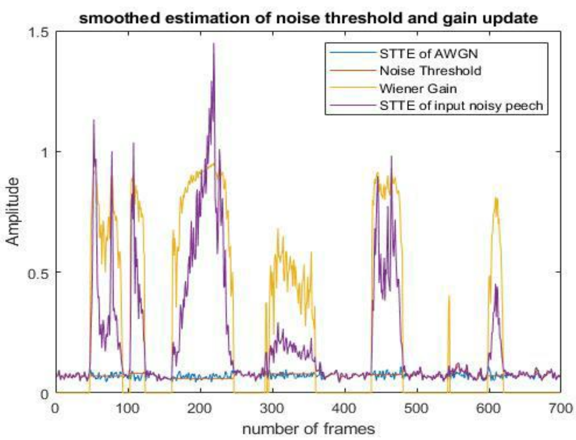

Figure 3. STTE of incoming noisy frames vs adaptive noise threshold and gain

Figures 4 and 5 shows the temporal and the spectral plots of 4(a) and 5(a) original speech, 4(b) and 5(b) incoming noisy speech corrupted by AWGN noise and the enhanced speech using the 4(c) and 5(c) direct subtraction method and the 4(d) and 5(d) proposed method respectively, plotted using MATLAB with the parameters defined in this paper. It can be observed that proposed method suppresses noise effectively with a smaller number of computations compared to the work done previously in spectral domain [24].

The short time signal-to-noise ratio (ST-SNR) [25] can be used to evaluate the speech enhancement algorithms either in time or frequency domain. Perhaps, the time domain evaluation of ST-SNR is one of the simplest objective measurements used to evaluate speech enhancement applications. Kolbæk *et al.* [26] combinedly presents six different types of loss functions to evaluate the performance of the end-to-end time domain speech enhancement techniques using neural networks. Among them the short time mean square error (ST-MSE) [26] and the short time objective intelligibility (STOI) [27] are used along with the ST-SNR for objective evaluation of proposed speech enhancement in time domain for seven types of noises at three different SNRs and the results are summarised in the Table 1. From the simulations of proposed method, it is clear that it has given better results, if the signal intended is corrupted with the AWGN only. The frames with negative ST-SNR values are omitted from the calculation of ST-SNR.

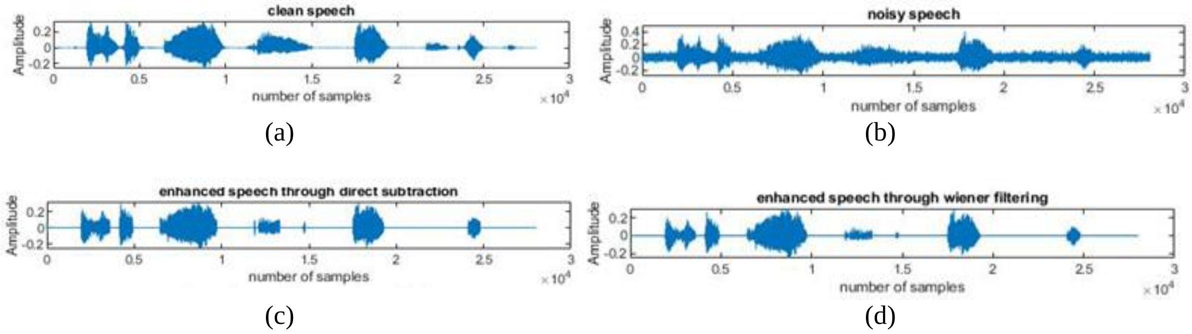

Figure 4. Time domain plots of: (a) pure speech, (b) AWGN noisy speech, (c) speech enhanced through direct subtraction, and (d) speech enhanced through proposed method

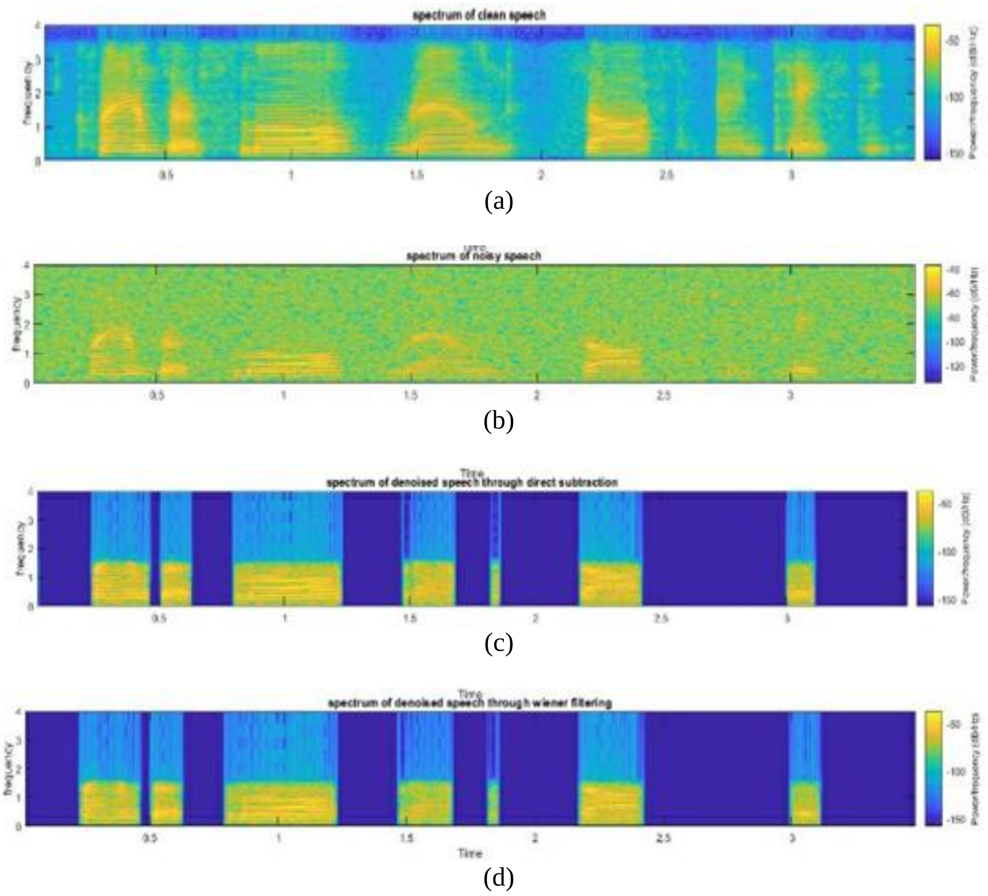

Figure 5. Spectrograms of (a) pure speech, (b) AWGN noisy speech, (c) speech enhanced through direct subtraction, and (d) speech enhanced through proposed method

Table 1. Objective evaluation of proposed method

Noise Type	0dB			5dB			10dB		
	ST-SNR	STOI	ST-MSE	ST-SNR	STOI	ST-MSE	ST-SNR	STOI	ST-MSE
AWGN	6.68	0.74	0.0255	9.26	0.84	0.0099	12.52	0.88	0.0037
Train	4.01	0.611	0.095	11.09	0.80	0.0224	12.48	0.88	0.0085
Station	6.55	0.649	0.057	10.97	0.83	0.04	14.04	0.9	0.013
Restaurant	4.37	0.712	0.11	6.011	0.84	0.03	13.26	0.887	0.0096
Car	3.98	0.66	0.055	8.81	0.78	0.0027	11.91	0.86	0.0085
Airport	3.73	0.62	0.07	10.47	0.83	0.04	12.36	0.91	0.01
Babble	5.85	0.65	0.066	9.43	0.79	0.032	11.25	0.888	0.0095

For better comparison, the frames with very low ST-SNR values are omitted from the calculation of ST-SNR. From the data given in the Table 2, it is clear that the proposed method meets the performance of the previous methods in all the aspects with minimum number of computations since it does not involve in transformation of the input. The direct subtraction of estimated noise is also done in time domain like Boll's subtraction in spectral domain. From the objective results, it clear that it performs well in improving ST-SNR but intelligibility of enhanced speech is reduced.

Table 2. Comparison of proposed method with existing spectral domain methods

AWGN		0dB			5dB			10dB		
		ST-SNR	STOI	ST-MSE	ST-SNR	STOI	ST-MSE	ST-SNR	STOI	ST-MSE
Spectral domain	SSBoll	12.46	0.70	0.0276	13.56	0.77	0.0134	14.8	0.8	0.0072
	Wiener Scalart	12.5	0.79	0.0116	13.13	0.81	0.0073	14	0.83	0.0057
Time domain	Direct subtraction	12.7	0.68	0.0289	14.7	0.78	0.0172	15.93	0.84	0.013
	Proposed method	12	0.74	0.0255	13.9	0.84	0.0099	15.8	0.884	0.0037

4. CONCLUSION

The end-to-end time domain speech enhancement algorithm was implemented in this present research. The proposed VAD based on ST-USNTR is used to divide incoming noisy speech into 3 different types of frames. Since it is complicated to implement transformation techniques in digital hearing aids, a simple and effective algorithm is developed for noise estimation and speech enhancement in time domain for low-cost applications in real time. In this work, the time domain Wiener filter based on first order statistics was used. There is slight change in the gain calculation in order to apply it equally well to the different real-world noises other than an AWGN. The separation of incoming frames into three different types and the smoothed update of the noise threshold improved the performance of single channel speech enhancement in hearing aids.

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BIOGRAPHIES OF AUTHORS

Dr. Padmaja Nimmagadda received B E (ECE) from University of Mumbai, India in 1998 and M Tech from S V University College of Engineering, Tirupati in 2003 and Ph.D. from S V University in the area of Atmospheric Radar Signal Processing in 2012. Currently she is working as Professor, SreeVidyanikethan Engineering College (Autonomous), Tirupati, India. She has 20 years of teaching experience. She has published 45 technical papers in various reputed International Journals & conferences. Her areas of interests include Signal and Image Processing, Communication Systems. She is Life Member of ISTE, IETE, IAENG, ISCA and IACSIT. She is Member, Review Panel of various International Journals. She served as Technical Session Chairs, Technical Program Committee member and delivered keynote address in various National and International Conferences. She is working for two projects funded by ISRO and DST and also serving as Mentor for DST Project under WOS scheme worth 90 Lakhs. She served as Executive Council member, and presently serving as Treasurer, IETE, Tirupati center. She received Best Teacher Award at C R Engineering College and Best Citizens of India Award from International Publishing House, New Delhi. She conducted several Workshops, conferences and Faculty Development Programs for the benefit of Faculty. She is presently serving as President, Institute Innovation Council, SVEC under MHRD Innovation Cell. She can be contacted at email: padmaja202@gmail.com

Kondru Ayyappa Swamy was born in India in 1988. He received the B.Tech. degree in Electronics and Communication Engineering from Aditya Engineering College, Surampalem, India and the M.Tech. degree in Digital Systems and Signal Processing from GITAM University, Vishakhapatnam, India, in 2009 and 2012, respectively, and pursuing Ph.D. degree in VIT University, Vellore, India. He is currently a Senior Assistant Professor with the Department of Electronics and Communication Engineering, Aditya Engineering College, Surampalem, India. His current research interests include signal processing for hearing aid, image and video compression. He can be contacted at email: ayyappa.kondru@gmail.com.

Samuda Prathima received B Tech (ECE) from Gokula Krishna College of Engineering, Sullurpet, India in 2010 and M Tech from SreeVidyanikethan Engineering College, Tirupati, India in 2013. Currently she is working as an Assistant Professor in RMK College of Engineering and Technology (Autonomous), Chennai, India. She has three years of teaching experience and three years of research experience. Her areas of interests include speech signal processing and deep llearning. She can be contacted at email: samudaprathimaece@rmkcet.ac.in.

Sushma Chintha obtained her bachelor's degree in Electronics and Communication Engineering in 2015 from Sree Venkateswara College of Engineering, Tirupati, Andhra Pradesh. She completed her Post graduation in Digital Electronics and Communication Systems (DECS) at SreeVidyanikethan Engineering College, Tirupati, India in 2018. Currently she is pursuing her doctoral studies at JNTUA University, Anantapur. She is also working as Junior Research Fellow in DST-TIDE project at SreeVidyanikethan Engineering College (Autonomous), Tirupati, Andhra Pradesh. Her areas of interests include Signal and Image Processing. She can be contacted at email: sushma040694@gmail.com.

Zachariah Callottu Alex was born in India in 1964. He received the B.Sc. degree in physics and the M.Sc. degree in electronics from Kerala University, Palayam, India, in 1984 and 1986, respectively, and the Ph.D. degree from Jamia Millia Islamia Central University, New Delhi, India, in 1996. He is currently a Senior Professor with the Department of Sensor and Biomedical Technology and Director (Sponsored Research), VIT University, Vellore, India. His current research interests include antenna design for UWB applications, RF switching and fabrication of fiber-optic gas sensors. He can be contacted at email: zachariahcalex@vit.ac.in.