ISSN (Print) : 2319-5940
ISSN (Online) : 2278-1021

International Journal of Advanced Research in Computer and Communication Engineering
Vol. 2, Issue 1, January 2013

ESTIMATION OF REVERBERATION TIME USING LPC FILTER AND MAXIMUM LIKELIHOOD ESTIMATOR AND DRR

JAYASHREE.R[1], MANJU KRISHNA[2], M.VANITHA LAKSHMI[3]

PG Scholar, Dept of PG studies in Engineering, S.A Engineering College, Chennai, India[1]

PG Scholar, Dept of PG studies in Engineering, S.A Engineering College, Chennai, India[2]

Assistant professor, Dept of PG studies in Engineering, S.A Engineering College, Chennai, India[3]

ABSTRACT: ***This paper presents a speech-model using the Linear Predictive (LP) residual signal and Maximum Likelihood Estimator (MLE). With this model an accuracy of the reverberation time estimation can be improved. During past decade, the reverberation time estimation was performed using only maximum likelihood detector, which resulted in excess time of estimation. For the purpose of estimating room acoustics, it is very essential to calculate the reverberation time more accurately. The study shows that Direct-to-Reverberation Ratio can also be calculated in accordance to the estimation of Reverberation Time. These two parameters can be made useful in the application of hearing aid.***

Keywords*:* **LP, Reverberation Time (RT), MLE, Direct-to-Reverberation Ratio(DRR), Hearing aid.**

I. INTRODUCTION

SPEECH signals captured in an enclosed environment contain reverberation due to reflections from surrounding objects. This deteriorates the quality and intelligibility of speech which can degrade the performance of many applications such as hands-free teleconferencing and automatic speech recognition (ASR).The reverberation time(RT) is important quantity for the characterization of enclosed auditory spaces. The reverberation time is typically represented by the parameter $RT_{60,}$ which is defined as the time at which the remaining RIR power is 60 dB lower than the total RIR power .The measurement of RT out of a sound decay can be done by the interrupted noise method[1].Schroeder has proposed a method to obtain the RT directly from a measured Room Impulse Response(RIR) instead of calculating it from the ensemble average of different sound decays[2].

Traditionally, reverberation time can be determined analytically using the room geometry and wall absorption properties. Semi-blind methods have been developed where the room characteristics are obtained using statistical learning theory [3].At present , blind methods have been developed which perform RT_{60} estimation from the received speech signal[4]. The method presented in this paper is a partially blind method based on locating segments in the input signal which are suitable for the estimation of the reverberation time. The latter group includes maximum likelihood estimation based methods [5] [6] [7] and blind deconvolution, which gives the impulse response as a byproduct. However, blind deconvolution only works when the impulse response is minimum phase, a condition that is not fulfilled in most real acoustic spaces [5].

Recently, Wen *et al.*[8] proposed a blind method based on a time–frequency room decay model which is related to Polack’s statistical reverberation model. This method requires a speech signal obtained from the start of a pause, and also needs speech with a long duration to estimate $RT_{60.}$ The diffuse tail of the reverberation is modeled as exponentially damped white Gaussian noise in [9] and ML estimate of the time constant of the decay is used to obtain $RT_{60.}$ A fast online implementation of this method is proposed in [10]. An order-statistics filter is used to extract the true estimates [9], [10]. As

ISSN (Print) : 2319-5940
ISSN (Online) : 2278-1021

International Journal of Advanced Research in Computer and Communication Engineering
Vol. 2, Issue 1, January 2013

a result ,it is sensitive to the analysis window length, as too large a window may cause the estimator to track the sound decay, reflecting the properties of the ongoing sound envelope rather than the reverberation curve.

Reverberation time estimation method that can satisfactorily estimate from the entire input speech signal, even in noisy environments, is a very challenging and practical research problem. In this paper, an ML-based method is proposed using the linear predictive (LP) residual of the speech signal to blindly perform reverberation time estimation using all of the input speech signal .The autocorrelation function can be used as an input to the ML estimator as they have the same properties as that of the room impulse response. Additionally, DRR is proposed to be estimated in accordance to the estimated RT.

As a result, the proposed method has the following advantages over other reverberation time estimation methods:

- RT to be estimated faster .
- Accuracy is achieved considering the effect of noise with the help of LPC filter. Sensitivity to Window length is overcome.
- The new RT estimation possesses significantly lower computational complexity.
- It is shown that this method can successfully estimate reverberation time with less speech data than that required by conventional ML-based methods.

The remainder of this paper is organized as follows. Section II describes models and characterization of Room Reverberation .In Section III, the proposed method is presented. More specifically, provides the notation and a description of the system model, followed by the proposed method. The experimental results are presented in Section IV. Finally, conclusions are given in Section V.

II.ROOM REVERBERATION

In this section, models of room reverberation are presented. The parameters commonly used to characterize reverberation are presented, as well as methods to generate reverberant speech.

A. Models of Room Reverberation

Conventionally, the propagation from source to microphone in a reverberant enclosure is modeled as a linear filtering process. The reverberant signal s(n) is modeled as a convolution of the anechoic Source speech signal x(n) ,with the room IR r(n) as

$$s(n) = x(n) * r(n). \qquad (1)$$

If additive background noise N(n) is present, then (1) becomes

$$s(n) = x(n) * r(n) + N(n). \qquad (2)$$

It is known that under the diffuse sound field assumption, the ensemble average of the squared room IR exponentially decays with time as

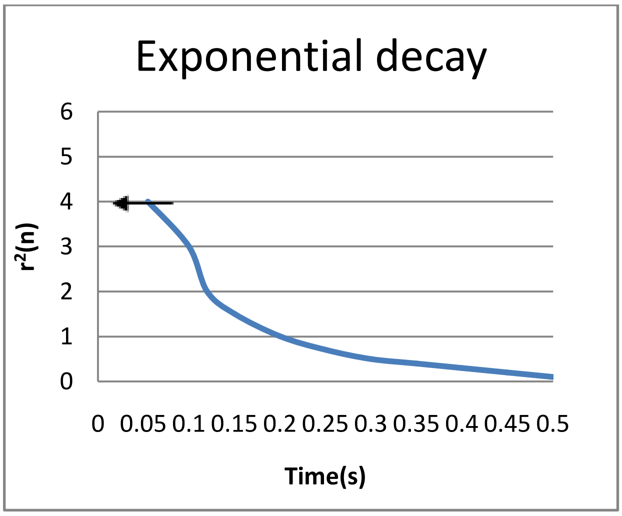
$$\langle r^2(n) \rangle = A \exp(-kn) \qquad (3)$$

The angled brackets $\langle . \rangle$denote the ensemble average, A is a gain term, and k is the damping factor given by

$$k = \log 10^6 / (Fs \times T_{60}). \qquad (4)$$

where Fs is the sampling frequency, and T60 is the so-called reverberation time.

The plot in Fig. 1 illustrates the exponential decay of a room IR generated via with *T*60 = 0.5 s and *Fs* = 8 kHz. A simulation method for small rooms based on an approximate image expansion for rectangular non rigid-wall enclosures has been discussed. The method is simple, easy to implement and efficient for computer simulation.



→ Early Reverberation

ISSN (Print) : 2319-5940
ISSN (Online) : 2278-1021

International Journal of Advanced Research in Computer and Communication Engineering
Vol. 2, Issue 1, January 2013

Fig.1. Exponential decay of the late reflections of a room with *T*60 = 0.5 s.

B .Characterization of Room Reverberation

Reverberation time (*T*60) is the parameter most widely used to characterize room acoustics. By definition, it is the time required for the sound energy to decay by 60 dB after the sound source has been turned off . Commonly, the so-called Schroeder integral is used to calculate *T*60 from the room IR. Other parameters that characterize room acoustics and are obtained from the room IR include the early decay timethe speech clarity index (energy ratio between the 50-ms early reflections and the remaining late reflections), and the direct-to-reverberant energy ratio (DRR). The DRR, which is

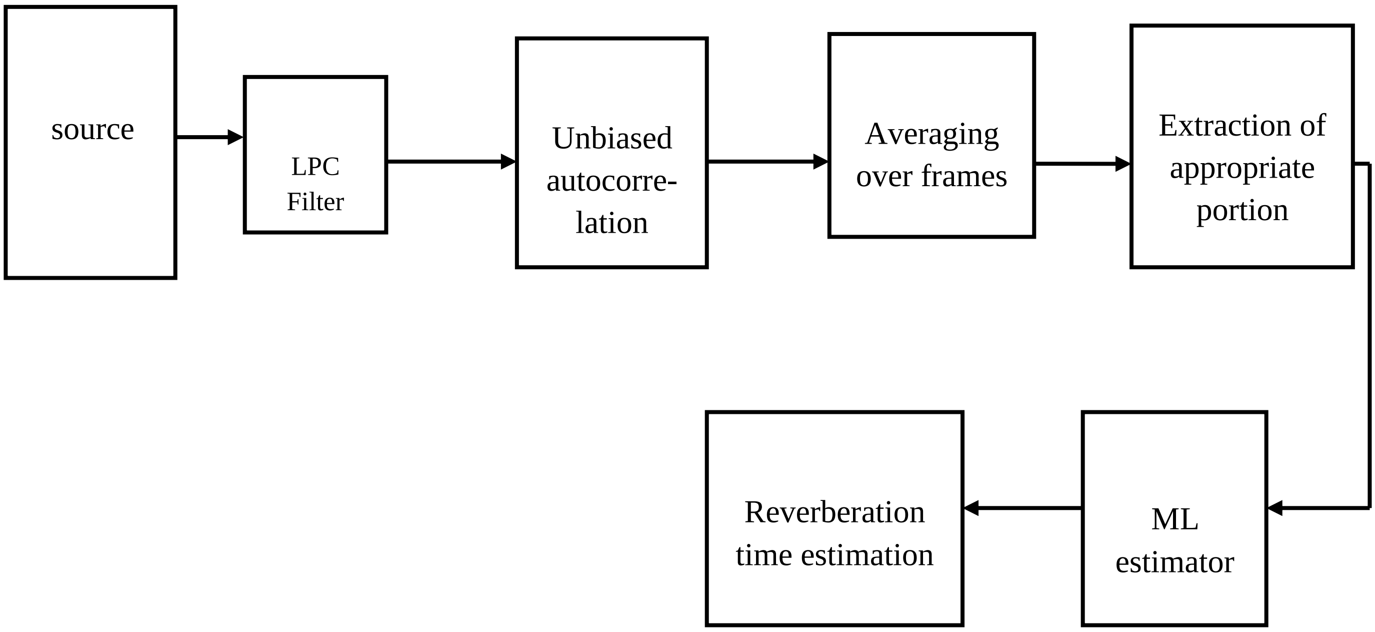


Fig.2. Block diagram to estimate Reverberation Time

expressed in decibels, is the energy ratio between the direct sound and the room reverberation and is given by,

$$\mathrm{DRR} = 10\log_{10}\left[\frac{\sum_{n=0}^{n_d} r^2(n)}{\sum_{n=n_d+1}^{\infty} r^2(n)}\right] \qquad (5)$$

Where, $n_d F_s$ is the direct sound arrival time.

III. PROPOSED SPEECH-MODEL-BASED METHOD

The idea of the proposed method is to extract the required information for estimation from all available data . The speech model is developed, the vocal tract effects of the speech signal are first removed by a linear predictive coding (LPC) filter. Then, the autocorrelation function of the LPC filter output (LP residual), has the same required statistical properties for ML estimation as the RIR. From the output of the ML estimation,RT is estimated more accurately. With this estimated RT , Direct-To-Reverberation energy ratio is estimated.

METHOD DESCRIPTION

A speech signal is given to LPC filter to get LP residual signal as,

$$x[n] = s[n] * h[n] \qquad (6)$$

In most enclosed environments of interest, the RIR can be expressed by Polack's model as one realization of a non-stationary stochastic process,

$$h[n] = w[n]\, a^n \qquad (7)$$

where,n> 0

Also,

ISSN (Print) : 2319-5940
ISSN (Online) : 2278-1021

IJARCCE

International Journal of Advanced Research in Computer and Communication Engineering
Vol. 2, Issue 1, January 2013

$$a = e^{-\delta} \tag{8}$$

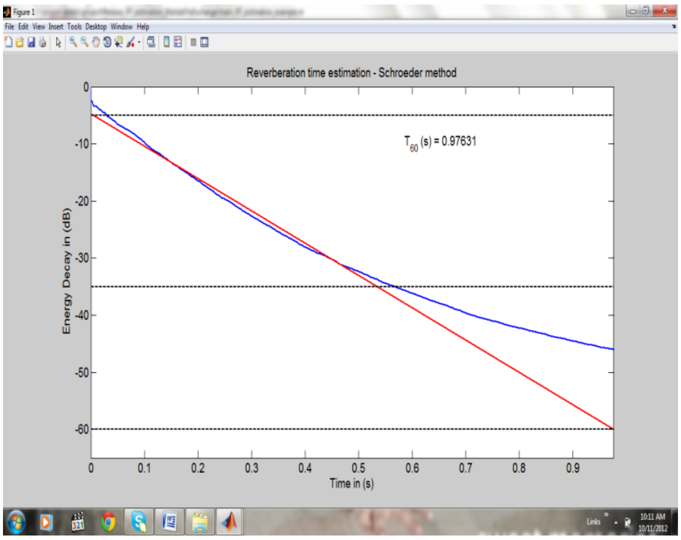
δ is inversely proportional to the reverberation time as given from[13],

$$\delta = \frac{3\ln(10)}{RT_{60} f_s} \tag{9}$$

Where , f_s is the sampling frequency.

IV . SIMULATION AND RESULTS

In this section, simulation results are provided to illustrate the performance of the proposed method. In this section, simulation results are provided to illustrate the performance of the proposed method. The RIRs have reverberation times of 0.1s to 1 s. The Reverberation time is estimated to be more accurate than the previous method given by Schroeder as shown in fig 3.



The fig 3 shows the plot of time versus energy decay. The decay is less for the proposed method as shFig.2. Plot of reverberation time versus energy decay.

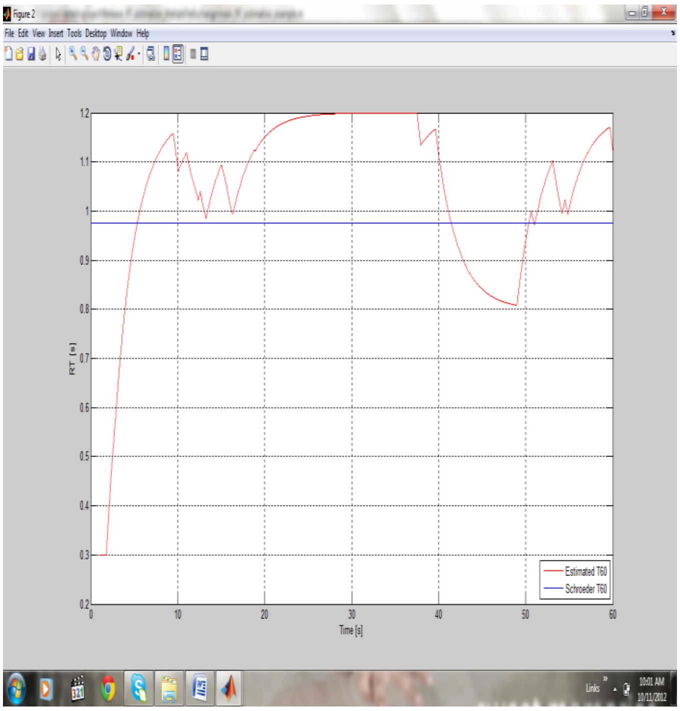
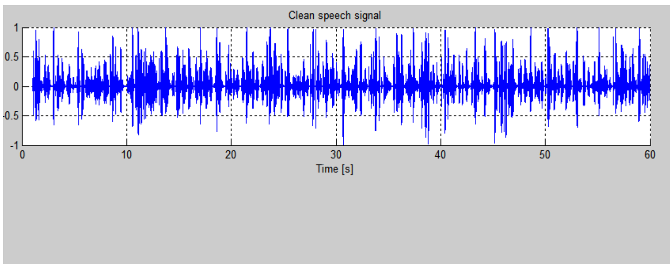


Fig.4. Plot of normal speech time versus Reverberation time

V.CONCLUSION

The 1min 48 seconds of clean speech from the TSP database sampled at 16 kHz were convolved withRIRs constructed by the image method to obtain the reverberant signals as shown: The results shows that the Elapsed time is 2.112663 seconds .Schroeder T60: 0.98s Estimated T60: 1.08s.Thus the estimated output shows no much deviation from the original speech time. The future work is led to the estimation of direct-to-reverberation energy ratio. The aim of the work is to make the DRR to fall within the range of -10 dB to 10dB for the speech signal



ISSN (Print) : 2319-5940
ISSN (Online) : 2278-1021

International Journal of Advanced Research in Computer and Communication Engineering
Vol. 2, Issue 1, January 2013

REFERENCES

[1]IS0-3382,"Acoustics-Measurement of the Reverberation Time of Rooms with Reference to Other Acoustical parameters", Geneve,Switzerland,1997.

[2]M.R Schroeder,"New Method of measuring Reverberation Time",Journal of the acoustical Society of America,vol.37,pp.409-412,1965.

[3] T. H. Falk and W.-Y. Chan, "Temporal dynamics for blind measurement of room acoustical parameters," IEEE Trans. Instrum. Meas., vol. 59, no. 4, pp. 978–989, Apr. 2010.

[4] S.Vesa and A. Harma, "Automatic estimation of reverberation timefrom binaural signals," in Proc. IEEE Int. Conf. Acoust., Speech, Signal Process., Mar. 2005, pp. 281–284.

[5] R. Ratnam, D. L. Jones, B. C. Wheeler, W. D. O'Brien Jr.,C. R. Lansing, and A. S. Feng, "Blind Estimation of ReverberationTime," Journal of The Acoustical Society of America,vol. 114, no. 5, pp. 2877–2892

[6] R. Ratnam, D. L. Jones, and W. D. O'Brien Jr., "Fast Algorithms for Blind Estimation of Reverberation Time," IEEE Signal Processing Letters, vol. 11, no. 6, pp. 537–540, 2004.

[7] L. Couvreur and C. Couvreur, "Robust Automatic Speech Recognition in Reverberant Environments by Model Selection,"in Proceedings of the International Workshop on Hands-Free Speech Communication (HSC-2001), Kyoto,Japan, April 2001, pp. 147–150.

[8] J. Y. C. Wen, E. A. P. Habets, and P. A. Naylor, "Blind estimationof reverberation time based on the distribution of signal decay rates,"in Proc. IEEE Int. Conf. Acoust., Speech, Signal Process., Mar.–Apr.2008, pp. 329–332

[9] R. Ratnam, D. L. Jones, B. C. Wheeler, W. D. O'Brien Jr., C. R.Lansing, and A. S. Feng, "Blind estimation of reverberation time," J. Acoust. Soc. Amer., vol. 114, no. 5, pp. 2877

[10] R. Ratnam, D. L. Jones, and W. D. O'Brien Jr., "Fast algorithms for blind estimation of reverberation time," IEEE Signal Process. Lett., vol. 11, pp. 537–540, Jun. 2004.

[11] H. Kuttruff, Room Acoustics, 4th ed. New York: Elsevier, 2000.

[12] E. A. P. Habets, "Single- and multi-microphone speech dereverberation using spectral enhancement," Ph.D. dissertation.