MULTIPATH DELAY ESTIMATION FOR ACOUSTIC ECHO CHANNEL

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ABSTRACT

A new algorithm for multipath delay estimation based upon Autocorrelation Estimator(AE) has been proposed. The existing Generalized Autocorrelation Estimator (GAE) has been modified by passing it through a sliding window of suitable shape and size resulting in improvement in performance. The performance improvement obtained because of the window has been determined for the case when the input is a filtered version of a white Gaussian signal. Accuracy Percentage (AP) has been plotted against SNR. Simulation results show that the proposed scheme performs better delay estimation even at relatively low SNR.

1. INTRODUCTION

Multipath is observed when the transmitted signal is received at the receiver through more than one path. In acoustic echo source sound reaches the microphone not only directly, but also via reflections from neighbouring objects. Therefore the received signal is a sum of delayed, attenuated and filtered versions of the source signal. However most of the existing methods for estimating time delay in multipath assume that the channel is ideal [1],[2],[3],[4],[5],[7],[8]. In the present work we relax this assumption and do not assume that the channel is ideal.

One simple way to estimate the time delays is to autocorrelate the received signal. The resulting correlation function has peaks to various delay differences; the locations of these peaks can be used as estimate of the time delay, in case these are resolvable. These multipath peaks in correlogram are resolvable as long as the time delay differences are at least greater than the duration of source signal correlation plus length of impulse response of channel filter. In general the signal from the source encounters many scatterers, leading to the difficulty of nonresolvability since peaks merge among each other [6].

In this paper the existing Generalized Autocorrelation Estimator [1] has been modified by running a window on the squared correlogram before applying the detection procedure. The window will smoothen the correlogram and minimize the effect of noise. The peaks of the resulting correlogram are detected to obtain estimates of delays and their differences.

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2. PROBLEM FORMULATION

The observed signal is modeled as a sum of delayed, attenuated and filtered version of the input signal. The simplified model for two reflections apart from direct path is shown in fig.1 [9]. The

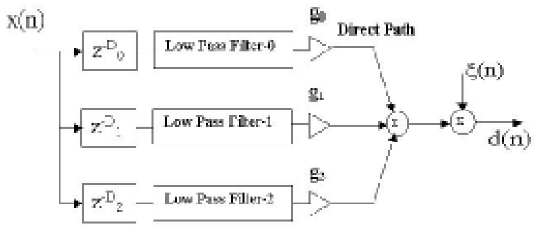

Fig. 1. Multipath Reflection Model

observation is a linear function of filter coefficients but nonlinear function of delay.

$$d(n) = \sum_{i=0}^{M} H_i(q)\, x(n - D_i) + \xi(n); \qquad (1)$$
$$n = 0, 1, ..., N-1$$

where
$D_0 = 0$: Direct path is assumed to have zero delay.
$d(n)$:Output due to input $x(n)$ and uncorrelated noise $\xi(n)$.
$H_i(q)$: ith path Low Pass filter of order L_i with

$$H_i(q) = g_i \times \left(1 + \sum_{j=1}^{L_i} h_{i,j} q^{-j}\right)$$

D_i, g_i: Delay and attenuation associated with ith reflection respectively.
h_{ij}: Impulse response (real) of ith path
q: Unit delay
M: Number of multipaths.

Autocorrelation of the output signal is:

$$\begin{aligned} R_{dd}(D) &= R_{\xi\xi}(D) + R_{xx}(D) \\ &\times \left(\sum_{i=0}^{M} H_i(q) H_i(q^{-1})\right) \\ &+ \sum_{i=1}^{M} H_0(q) H_i(q^{-1})\, R_{xx}(D + D_i) \\ &+ \sum_{i=1}^{M} H_i(q) H_0(q^{-1})\, R_{xx}(D - D_i) \\ &+ \sum_{i=1 i\neq j}^{M} \sum_{j=1}^{M} H_i(q) H_j(q^{-1}) \\ &\times R_{xx}(D - (D_i - D_j)) \quad (2) \end{aligned}$$

where $R_{xx}(D)$ & $R_{\xi\xi}(D)$ are autocorrelation function of $x(n)$ & $\xi(n)$ respectively. Although the peaks of $R_{dd}(D)$ are expected to be exactly at $0, \pm D_i$ & $\pm |D_i - D_j|$, the superposition of various correlograms results in bias and hence the peaks are obtained at say $0, \pm D_i'$ & $\pm |D_i' - D_j'|$.

The requirement is to estimate these delays from the available data. A suitable algorithm is to be designed to detect these delays in the presence of noise and resolve them with low computational cost.

3. PROPOSED SCHEME

The following assumptions are made:

3.1. Assumptions:

1. Signal and noise are white Gaussian processes having variances σ_x^2 and σ_ξ^2 respectively.
2. Number of multipaths M are known and associated attenuations g_i are real.
3. The duration of the observed signal is greater than the highest time delay ($N > D_M$).
4. The time delays are positive and distinct and satisfy $D_1 < D_2 <, ..., < D_M$.
5. The time delays satisfy

$$D_i \neq D_j,\ D_i \neq D_j - D_r,\ D_i - D_j \neq D_r - D_s; \quad i \neq j \neq r \neq s \in [1, 2, ..., M]$$

6. Time delay differences are greater than the autocorrelation of input signal $x(n)$ plus length of channel impulse response $h_{i,j}$.

3.2. Algorithm

The location of peak near D_i are estimated from the sample correlation according to the following steps:

1. Step1: Obtain

$$\begin{aligned} \hat{R}_{dd}(D) &= \frac{1}{N} \sum_{n=0}^{N-D-1} d(n)d(n+D); \\ &= \hat{R}_{dd}(-D) \\ & D = 0, 1, 2, ..., (N-1) \end{aligned}$$

2. Step2: Obtain $\hat{R}_{dd}^2(l) = \left[\hat{R}_{dd}(l)\right]^2$
3. Step3: Zero padding $\hat{R}_{dd}^2(l)$ then apply sliding window of length W

$$Z(D) = \sum_{\tau=D-W}^{D+W} \bar{R}_{dd}(\tau); \quad D = 0, 1, 2, ..., (N-1)$$

Where $\bar{R}_{dd}(D) = \sum_{l=D-W}^{D+W} \hat{R}_{dd}^2(l)$

4. Step4: *do* $t = 1, N$

$$\begin{aligned} & if\ \ Z(t+1) > Z(t) \ \ \&\ \ Z(t+1) > Z(t+2) \\ & \quad return\ \ Z(t+1) \\ & else\ \ \ Z(t+1) = 0 \end{aligned}$$

5. Step5: Number each peak of modified $Z(D)$ with index $n_0, n_1, n_2, ..., n_{M(M+1)/2}$ where $n_0 = 0$ and $n_{M(M+1)/2} = D_M$ the Mth delay.
6. Step6: *do* $i = 1, M(M+1)/2$

$$\begin{aligned} & if\ \ Z\left(n_{M(M+1)/2} - n_i\right) = 0; \\ & \quad return\ \ Z(n_i) = 0 \\ & \quad otherwise\ \ Z(n_i) \end{aligned}$$

7. Step7: Renumber all the peaks, only $2M-1$ peaks be present apart from n_0 and $n_{2M-1} = D_M$
8. Step8: do $i = 1, (2M-1)$

$$\begin{aligned} & if\ \ \hat{R}_{dd}^2(n_{2M-1}) * \hat{R}_{dd}^2(n_i) = \hat{R}_{dd}^2(n_{2M-1} - n_i) \\ & \quad retain\ \hat{R}_{dd}^2(n_i) \\ & \quad otherwise\ \ \hat{R}_{dd}^2(n_i) = 0 \end{aligned}$$

9. Step9: Save the identified indexes of M peaks which are the estimate of true delays.

4. SIMULATION RESULTS

Received signal is obtained by passing white Gaussian noise through LPF (FIR) of order 12. The coefficients of the filter are scaled by the attenuation factors g_is . We consider two multipaths apart from direct path having $g_1 = 0.9$, $g_2 = 0.8$ and delays 20 and 35 respectively. The triangular window selected for smoothing the waveform is 7 samples long. The simulation results are averaged for 400 independent runs and observation length N=4096. Accuracy Percentage (AP) as defined in [10] is the number of times the delay is correctly estimated divided by the total number of trials. In all the plots D_i and WD_i is estimate of delay D_i without and with window respectively. Fig.2 presents a plot of AP as a function of SNR. We observe that the performance with window is better. AP is plotted with difference between delays as shown in Fig.3. There is increase in AP with the increase in difference between delays. In Fig.4 AP is plotted with observation length and shows that there is improvement with the increase in observation length.

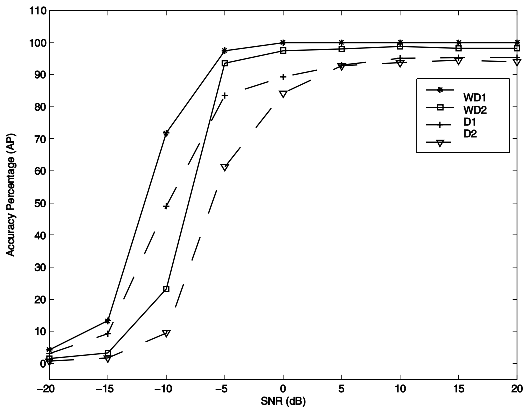

Fig. 2. Performance improvement due to running window

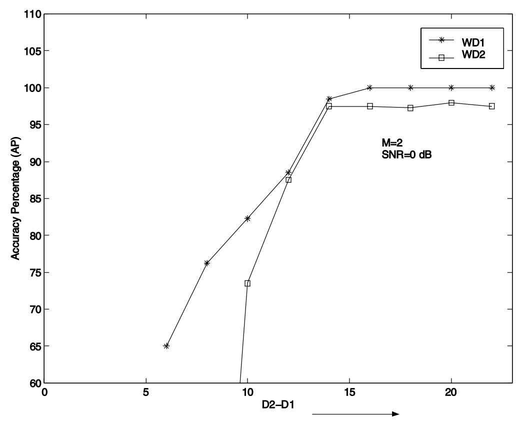

Fig. 3. Accuracy percentage (AP) vs Delay difference

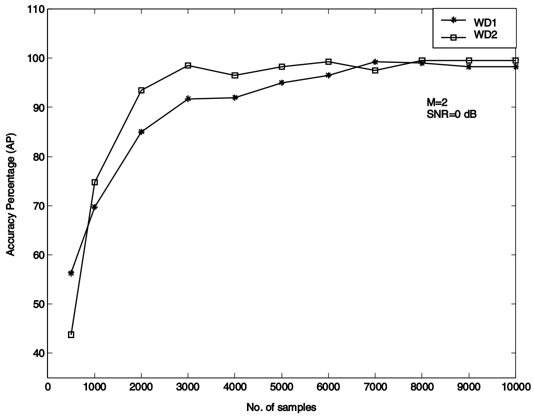

Fig. 4. Accuracy percentage (AP)for different Observation lengths

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