Sound Source Localization Based on Time Delay Estimation Using Variable Step Size in LMS Adaptive Filter

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***Abstract*—Time delay estimation is one of the key methods in sound source localization and LMS adaptive filter is one of the popular time delay estimation methods. However the time delay estimation based on LMS adaptive filter has to sacrifice the Real-time performance to improve the positioning accuracy in sound source localization. Compared with the LMS adaptive filter, a variable step size LMS adaptive filter applied on a time delay estimation system is presented in this paper for sound source localization. The results show that time delay estimation based on variable step size LMS adaptive filter has better real-time performance and position accuracy in sound source localization.**

***Index Terms*—least mean square, sound source localization, time delay estimation, variable step size**

I. Introduction

Sound source localization is an indispensable technology in the field of voice communications such as video conferencing, hand-free mobile telephony and so on [1]. Time delay estimation (TDE) is one of the key technologies in sound source localization. The aim of TDE is to estimate the time delays caused by the distance difference between the sound source and different microphones. The sound Source can be estimated by time delays according to the model of microphones.

LMS adaptive filter is one of the popular TDE methods. However, TDE based on LMS adaptive filter has to sacrifice the real-time performance to improve the positioning accuracy.

Instead of that this paper present a TDE method based on a variable step size (VSS) LMS algorithm applied on a three-dimensional microphone model. Simulation results illustrate the new method has better real-time performance and positioning accuracy in sound source localization.

II. Survey of Related Works

The initial widely used TDE methods are generalized cross-correlation (GCC) and LMS adaptive filter. Literature [2] and [3] present the subspace based eigenvalue decomposition (EVD) and acoustic transfer function ratio (ATF-s ratio) methods respectively. Literature [4-5] described the features of different methods by comparison of them. LMS adaptive filter is still popular because it does not have to know the statistical characteristics of the input signals.

III. Problem Statement and Main Contribution

Literature [6] presented a VSS algorithm applied in LMS adaptive filter, but it hasn't been used for TDE in sound source localization. The objective of this paper is to find out whether the new method consisting of the VSS improves the real-time and positioning accuracy in the sound source localization.

The TDE system based on VSS adaptive filter and a three-dimensional microphone array system are modeled and implemented by MATLAB. Finally simulation results verify the real-time performance and positioning accuracy respectively.

IV. Problem Solution

A. Modeling

1) Three-dimensional microphone array system: The model of three-dimensional microphone array is shown in Fig 1.

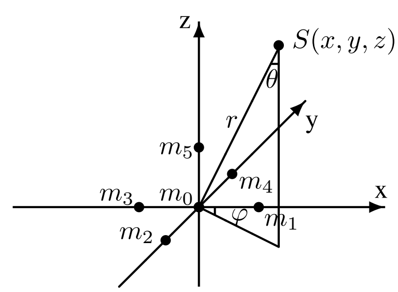

Fig. 1. Hexahydric microphone array model, where $m_0\_m_5$ denote six microphones, r is the distance between sound source $S(x,y,z)$ and reference microphone m_0, θ and φ represent pitch angle and azimuth angle respectively.

Sound source can be estimated by time delays according to the system which is easy to model mathematically.

2) Time delay estimation system: Time delays t_{i0} can be obtained by VSS-LMS adaptive filter as the Fig.2 shown.

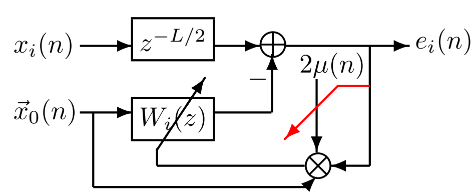

Fig. 2. Time delay estimation based on VSS-LMS adaptive filter, where L is the filter order, $\vec{x}_0(n)$ is the input signal from reference microphone m_0 and $x_i(n), i=1,2,...,5$ are desired signals from remaining microphones.

$$\begin{cases} \vec{w}_i(n+1) = \vec{w}_i(n) - 2\mu(n)e_i(n)\vec{x}_0(n) \\ \mu(n) = \beta\{1 - exp[-\alpha|e_i(n)|^2]\} \end{cases} \tag{1}$$

where $\vec{w}_i(n)$ is filter weight for the desired signal $x_i(n)$, α and β represent two coefficients for updating the step size $\mu(n)$.

Since the derivation of the phase function $\phi_i(w)$ of the filter represent the delay of samples, the time delays can be estimated as (2) shown.

$$t_{i0} = \frac{1}{f_s}(L/2 + \frac{d\phi_i(w)}{dw}) \tag{2}$$

where f_s is the sampling frequency and t_{i0} denotes time delay between $x_i(n)$ and $x_0(n)$.

B. Implementation

The implementation is done by MATLAB script as below.

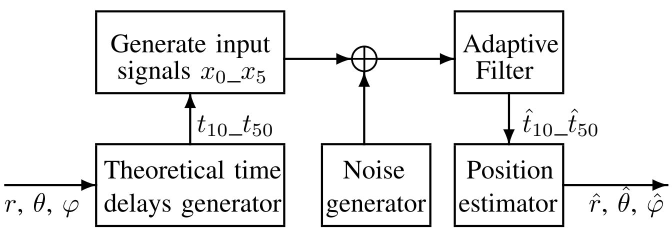

Fig. 3. Block diagram of the whole system

It can be divided into four steps as the following procedure.
1) Input the parameters of actual position r,θ,φ and generate theoretical time delays $t_{10}\_t_{50}$.
2) Generate theoretical received signals $x_0\_x_5$ using theoretical time delays $t_{10}\_t_{50}$.
3) Apply the received signals into adaptive filter using both LMS algorithm and VSS-LMS algorithm after adding noise to get the estimated time delays $\hat{t}_{10}\_\hat{t}_{50}$.
4) Using estimated time delays $\hat{t}_{10}\_\hat{t}_{50}$ to obtain the estimated sound source position $\hat{r}$, $\hat{\theta}$, $\hat{\varphi}$.

V. Simulation Results

A. Real Time Performance

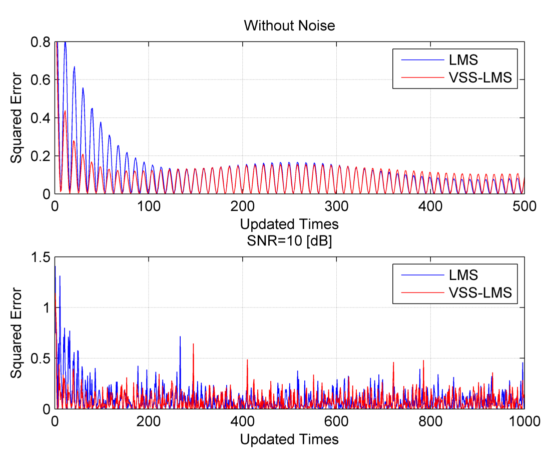

Fig. 4. Convergence speeds of both methods, where fixed step size for LMS $\mu = 0.0002$, variable step size coefficients $\alpha = 0.75$ and $\beta = 0.0025$, input position $P(r, \theta, \varphi) = (50m, 60°, 120°)$, filter length $L = 320$ and sampling frequency $f_s = 10000Hz$.

The real-time performance is observed by the convergence speed of both methods as shown in Fig 4.

B. Positioning Accuracy

Input three positions $P1$, $P2$ and $P3$, the estimation of positions by both methods is shown in Fig 5.

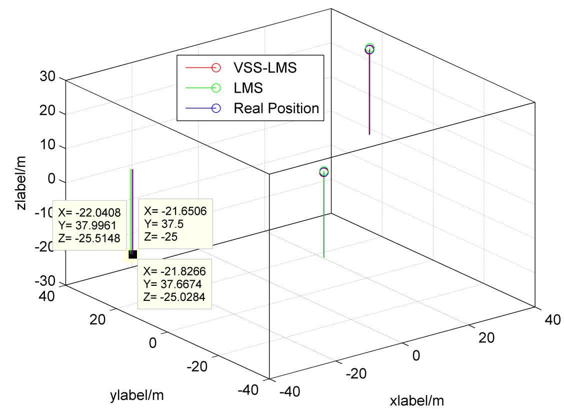

Fig. 5. Position estimation of both methods, where $P1 = (50m, 60°, 30°)$, $P2 = (50m, 60°, -120°)$, $P3 = (50m, 120°, 120°)$, SNR=10 dB and other parameters are the same as Fig 4

VI. Conclusion

As Fig 4 shown, TDE based on VSS-LMS algorithm converges faster than that based on LMS algorithm with and without noise. That means the new method has better real-time performance in sound source localization. On the other hand, positions estimated by the new method consisting of VSS-LMS algorithm has better accuracy comparing to LMS algorithm according to Fig 5.

However, the new method applied on sound source localization hasn't been experimented on an actual device. Furthermore, this paper doesn't consider aliasing noise and the influence of the microphone array model. These problems discussed above can be researched in the future.

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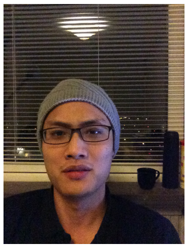
Biographies



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Appendix

A. The Mathematical Model of Microphone array system

$$\begin{cases} x^2+y^2+z^2=r^2 \\ (x-l)^2+y^2+z^2=(r+t_{10}c)^2 \\ x^2+(y-l)^2+z^2=(r+t_{20}c)^2 \\ (x+l)^2+y^2+z^2=(r+t_{30}c)^2 \\ x^2+(y+l)^2+z^2=(r+t_{40}c)^2 \\ x^2+y^2+(z-l)^2=(r+t_{50}c)^2 \end{cases} \qquad \text{(A-1)}$$

where r is the distance between sound source $P(x, y, z)$ and the reference microphone m_0, and l is the distance between other microphones m_i and sersor m_0, t_{i0} represents the time delay between m_i and m_0, and r is the sound speed.

B. The Table of Positioning Estimation

TABLE A-I

POSITIONING ESTIMATION

$P(r, \theta, \varphi)$	Estimated using LMS	Estimated using VSS-LMS
$(50, 60, 30)$	$(50.40, 59.81, 30.13)$	$(49.86, 60.11, 30.11)$
$(50, 60, -120°)$	$(50.62, 59.78, -120.09)$	$(49.81, 60.14, -120.09)$
$(50, 120, 120)$	$(50.22, 120.16$, $120.07)$	$(50.35, 119.97, 120.06$