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# Application of array microphone measurement for the enhancement of sound quality by use of adaptive time reversal method

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ABSTRACT

The purpose of this research is to develop an algorithm which can be used in the system of speech recognition. In the previous research, a target signal can be reconstructed by a signal separation technique which is based on time-reversal method (TRM) via self-focusing property. In this research, a multiple signal separation technique which is based on adaptive time-reversal method (ATRM) is developed. The advantage of ATRM is to reconstruct target signal and to cancel noise signals simultaneously if impulse response functions of environment between target source, noise sources, and array microphone are known. In the simulation, correlation coefficient and signal to noise ratio (SNR) are implemented as indicators to evaluate the performance of signal separation for TRM and ATRM in a reverberation room, and to investigate the differences between two methods. The comparison indicates that number of array microphone is significantly reduced from 14-microphones via TRM to 6-microphones via ATRM. Furthermore, the experiments of signal separation are conducted to a model of smart TV with multiple noise sources in the environment. Finally, a real time system of signal separation was developed. All the results indicate that the target of source was clearly reconstructed with enhancing of correlation coefficient and SNR by using ATRM.



## 1. Introduction

In recent years, the quality of audio is an important factor in our daily life. For the system of speech recognition, it is very important to have clarity and quality of voices. Time reversal method (TRM) has advantages of compensating distortion due to path effect in propagation and focusing signals at the original source location. Also it has been widely applied in ultrasonic, optics, underwater communication and nondestructive test [1]. The concept of time-reversal in time domain is equivalent to phase conjugation in frequency domain [2]. In 2000, Dungan, Dowling and Sabra researched the properties of TRM in shallow water [3–5]. These properties include the effects of the quantity, the spacing and the measuring distance of array hydrophones on TRM focusing. TRM reduced the errors of communication by self-focusing. Sabra used the self-focusing of TRM to overcome the reverberation environment and applied TRM for target detection in ocean [6]. Wu et al. [7] also compared the results of beamforming and TRM in audio separation problem. He pointed out TRM gives better results than beamforming does in a reverberation environment.

Suppose there is only one source, it is simple to reconstruct the source by using TRM due to the self-focusing property. However, TRM cannot completely reduce all the non-target sources as there are multiple sources in the environment. In an attempt to reduce the interference of the non-target sources, adaptive time-reversal method (ATRM) can be used to reconstruct the target source and cancel the non-target sources. The performance of ATRM is to provide multiple constraint conditions for all the sources. The fundamental theory of ATRM was based on adaptive beamforming which is proposed by Cox et al. [8]. In their study, the problem of multiple optimizations was defined by using Lagrange multiplier method. Also the adaptive beamforming approaches are used to process the array signal, such as minimum variance distortionless response (MVDR) beamforming, linearly constrained minimum variance (LCMV) beamforming and generalized sidelobe canceller (GSC) beamforming. Their objective is to focus the transmission signal at original source location and defocus the transmission signal at other appointed locations. The purpose is to focus the transmission signal at multiple source locations simultaneously.

In the present study, the technology of multiple signal separation (ATRM) which is based on TRM and MVDR is developed. The purpose is to separate the target signal from other noise signals. In Section 2, the theories of TRM and ATRM are discussed in details. In Section 3, the differences and performances of signal separation

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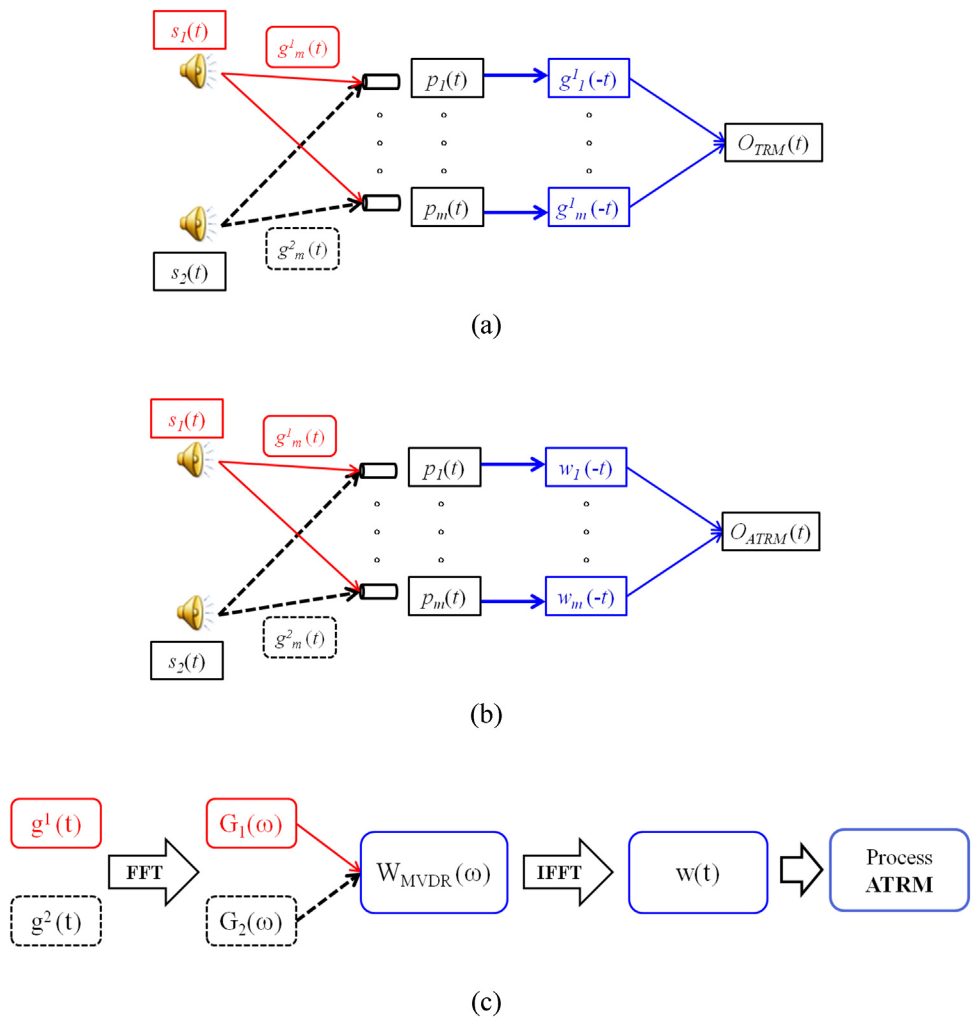


**Fig. 1.** (a) The diagram of TRM process. (b) The diagram of ATRM process. (c)The flow chart to obtain solution for adaptive weighting function $w(t)$.

between TRM and ATRM are analyzed. In Section 4, multiple sound signals are separated by using array microphone in the experiments. The results are shown to verify this approach in the present study. Finally, a real time test of ATRM system is performed and the results are discussed.

## 2. Theory

### 2.1. Time-reversal method (TRM)

In the time domain, suppose that $s_1(t)$ and $s_2(t)$ are the signal of target source and noise source, $p_m(t)$ is a received signal by the $m$th receiver. $g_m^1$ is an impulse response function of environment between the target source and the $m$th receiver, $g_m^2$ is an impulse response function of environment between the noise source and the $m$th receiver (as Fig. 1a). The output of TRM $O_{\mathrm{TRM}}(t)$ can be expressed as the following:

$$
\begin{aligned}
O_{\mathrm{TRM}}(t) &= \sum_{m=1}^{M} g_m^1(-t) \otimes p_m(t) \\
&= \sum_{m=1}^{M} g_m^1(-t) \otimes g_m^1(t) \otimes s_1(t) + \sum_{m=1}^{M} g_m^1(-t) \otimes g_m^2(-t) \otimes s_2(t),
\end{aligned} \tag{1}
$$

where $\otimes$ denotes the convolution integration. The self-focusing property of TRM can reconstruct the target signal. Suppose the reconstructed target signal is described in the first term of right hand side of Eq. (1) shown as following:

$$
\sum_{m=1}^{M} g_m^1(-t) \otimes g_m^1(t) \otimes s_1(t). \tag{2}
$$

However, interference of noise signals can't be completely cancelled in the output signal of TRM. It is assumed that noise signal is described as interference. The second term of right hand side of Eq. (1) is shown as following:

$$
\sum_{m=1}^{M} g_m^1(-t) \otimes g_m^2(t) \otimes s_2(t). \tag{3}
$$

The noise interference can be reduced by the summation effect of receiver array. However, the performance of summation effect is limited by the quantity of microphones.

### 2.2. Adaptive time-reversal method (ATRM)

In contrast to use impulse response function $g^1(-t)$ in time-reversal process, ATRM uses an adaptive weighting function $w(t)$