Applied Acoustics 137 (2018) 69–81

Contents lists available at ScienceDirect

Applied Acoustics

journal homepage: www.elsevier.com/locate/apacoust

Acoustic noise reduction by new two-channel proportionate forward symmetric adaptive decorrelating algorithms in sparse systems

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ARTICLE INFO

Keywords:
Acoustic noise
Sparse system
Proportionate algorithm
Decorrelation
SegMSE criterion
Convergence rate property

ABSTRACT

In several previous works, the two-channel adaptive filtering algorithms have been proposed and combined with forward-and-backward blind source separation structures for acoustic noise reduction and speech enhancement when the impulse responses are dispersive. In this paper, firstly we present all mathematical formulations of forward symmetric adaptive decorrelation algorithm based on normalized step-sizes control. Secondly, we propose three new proportionate algorithms that improve the convergence rate of cross-adaptive filters when the impulse responses of convolutive mixing system are sparse. To validate the good performance of proposed algorithms in term of noise reduction and speech enhancement properties, we do intensive experiments based on several criteria. To evaluate exactly the convergence speed property, we use the system mismatch and segmental mean square error criteria. We use also the cepstral distance and segmental signal-to-noise ratio to evaluate the quality of enhanced speech output. The obtained results have shown good performances of the proposed proportionate algorithms in comparison with the two-channel normalized decorrelation and forward NLMS algorithms with sparse systems.

1. Introduction

Acoustic noise reduction (ANR) and speech quality enhancement (SQE) are often used in many applications in telecommunication systems such as hand-free telephony, teleconferencing systems and hearing aids. In order to improve the robustness of ANR and SQE systems in such noisy environments, we can use different approaches [1,2]. Several one-, two- and multi-channel sensors techniques are proposed to resolve this problem [3–5]. Recently, the blind source separation (BSS) technique has been used to separate the speech and acoustic noise signal in different mixing system, i.e. instantaneous or convolutive.

Adaptive filtering algorithms are frequently employed in signal processing, telecommunications and many other applications because of its simplicity and robustness [6,7]. Recently, a very important amount of papers have investigated in ANR and SQE by using different adaptive filtering algorithms combined with the two-channel BSS structures [8–11]. These approaches are proposed to improve the behavior of ANR and SQE systems in terms of speed convergence, steady state, misadjustment values and complexity.

In two-channel BSS algorithms, we can use forward-and-backward structures which are simples and efficient [8,11]. We note that, the two-channel forward BSS is important structure used to enhance the speech signal but with a distortion. However, the second two-channel BSS structure (i.e. backward) represents the solution to resolve the distortion problem observed in the output speech signal estimated by forward structure [8]. Full analyses of these methods with and without post-filters are well described in [3,8]. We note also that, all of these structures/algorithms require manual-or-automatic voice activity detector system (VAD) to cancel the acoustic noise components at the outputs [12,13].

In this paper, we consider a two-channel convolutive mixing system. Several adaptive filtering algorithms have been proposed in time and frequency domains [14–16], as the double least mean square (DLMS), forward-and-backward normalized LMS (FNLMS and BNLMS), double affine projection algorithm (DAPA) [10], double fast Newton transversal filter (DFNTF) [17] and double pseudo APA [18]. These two-channel adaptive filtering BSS algorithms are used to identify the dispersive impulse responses (IRs) of two-channel convolutive mixture. It has been proven that the adaptive identification of unknown dispersive IR is equivalent to the problem of BSS technique [19–23]. In [20–23], the subband BSS algorithms have been proposed to improve the convergence rate. Recently, it was proposed efficient subband implementation and new variables step-sizes approaches of the two forward-and-backward BSS structures to improve their performances, i.e. the convergence speed and output speech quality [24–26]. The forward-and-backward symmetric adaptive decorrelation (SAD)

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https://doi.org/10.1016/j.apacoust.2018.03.014
Received 14 April 2017; Received in revised form 26 December 2017; Accepted 15 March 2018

Nomenclature

Abbreviations

μP-FND	μ-law proportionate forward normalized decorrelation
AD	adaptive decorrelation
ANR	acoustic noise reduction
APA	affine projection algorithm
BNLMS	backward normalized least-mean-squares
BSS	blind source separation
CD	cepstral distance
DAPA	double affine projection algorithm
DFNTF	double fast Newton transversal filter
DLMS	double LMS
FBSS	forward BSS
FND	forward normalized decorrelation
FNLMS	forward normalized least-mean-squares
IP-FND	improved proportionate forward normalized decorrelation
IRs	impulse responses
ISFT	inverse-short-Fourier-transform
LMS	least-mean-squares
MSD	mean-square-deviation
ND	normalized decorrelation
NEC	network echo cancellation
NLMS	normalized least-mean-squares
P-FND	proportionate forward normalized decorrelation
SAD	symmetric adaptive decorrelation
SFT	short-Fourier-transform
SegMSE	segmental Mean-square-error
SegSNR	segmental signal-to-noise ratio
SM	system mismatch
SQE	speech quality enhancement
VAD	voice activity detector

Notations

L	real impulse responses length
l	delay index
n	time index
U	mean averaging value
Fs	sampling frequency
D	number of sources
N	number of microphones

Mixture model

$s(n)$	original speech signal
$b(n)$	noise
$h_{11}(n)$, $h_{22}(n)$	direct acoustic impulse responses
$h_{12}(n)$, $h_{21}(n)$	sparse cross-coupling impulse responses
$p_1(n)$, $p_2(n)$	noisy speech signals
$\delta(n)$	Dirac impulse

Two-channel forward–backward structures

$w_{12}(n)$, $w_{21}(n)$	adaptive filters
$\mathbf{w}_{12}(n)$	adaptive filter vector
$\mathbf{w}_{21}(n)$	adaptive filter vector
$u_1(n)$	estimated speech by forward structure
$u_2(n)$	estimated noise by forward structure
$PF_1(n)$, $PF_2(n)$	post-filters
$s_1(n)$	estimated speech after post-filter
$s_2(n)$	estimated noise after post-filter

Parameters

ξ_{FND}	small positive constant in FND algorithm
ξ	small positive constant in proposed algorithm
α	small number take its values between −1 and 1
φ	very small positive number
ρ	positive number, ρ = 5/L
δ	positive number, δ = 0.01
μ	positive number, $\mu = \frac{1}{\varepsilon}$ where $\varepsilon = 0.001$
$\mathbf{Q}_{12}(n)$, $\mathbf{Q}_{21}(n)$	diagonals step-size control matrix (L × L)
μ_{12} and μ_{21}	normalized step-sizes parameters
λ_{12} and λ_{21}	step-sizes parameters
$\mu_{12,opt}$, $\mu_{21,opt}$	optimal step-sizes values
$\left.\begin{matrix}\boldsymbol{C}_{u_1u_2}(l)\\ \boldsymbol{C}_{u_2u_1}(l)\end{matrix}\right\}$	cross-correlation terms

algorithms [4,11,27] are important solutions to separate speech signal from noisy observations. These algorithms showed a good performance in two-channel convolutive mixing model with dispersive IRs. However, the main drawback of these algorithms is their poor performance when the IRs are sparse [28–30]. This inconvenience is well observed in transient phase. To overcomes these problems, these algorithms have to consider the following notes (i) the need to adapt a relatively long filter and (ii) the unavoidable adaptation noise occur at the inactive region of the tap weights [31].

In this paper, three efficient proportionate versions of two-channel forward normalized SAD algorithm that take into account the sparsity of the cross-channel IRs are presented. The proposed algorithms based on normalized step-sizes and proportionate techniques. The proposed proportionate forward algorithms allow improving the convergence speed and the misadjustement performances of their original versions.

This paper is presented as follows: in Section 2, the two-channel acoustic noise reduction problem is detailed. In Section 3, we present the mathematical formulations of modified normalized version of two-channel forward SAD algorithm. Section 4 and its sub-sections are reserved for the presentation of proposed two-channel proportionate forward algorithms and theirs full analysis. The simulation results are presented in Section 5 and finally the conclusion of this paper is presented in Section 6.

2. Acoustic noise reduction problem

In multi-sources situation, the blind source separation (BSS) is one of the efficient techniques which have shown a good performance. The major objective of the source separation technique is to estimate original signals using only the information of the observed signals in each channel. The blind source separation problem can be divided into two parts (Fig. 1) [32].

- **Mixture system modeling:** in the first step, we define the relationship between the original source signals and the observed signals in the output of the mixture system (i.e. instantaneous-or-convolutive mixtures).
- **BSS solution:** in the second step, we define the separation model and algorithm used to extract the estimated signals by using only the observed signal (identify the mixing system).

In the general case of a convolutive mixture model, we assume that we have D source signals which are real and statistically independent. The original signals are convoluted with all IRs channels to generate N observed signals. The relations between the original and the observed signals are given as follows: