Applied Soft Computing 42 (2016) 132–143

Contents lists available at ScienceDirect

Applied Soft Computing

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

Improved subband-forward algorithm for acoustic noise reduction and speech quality enhancement

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A R T I C L E I N F O

Article history:
Received 28 February 2015
Received in revised form 22 January 2016
Accepted 27 January 2016
Available online 7 February 2016

Keywords:
Subband structures
Acoustic noise reduction
Speech enhancement
Signal-to-noise ratio
Cepstral distortion

A B S T R A C T

This paper addresses the problem of speech enhancement and acoustic noise reduction by adaptive filtering algorithms. Recently, we have proposed a new Forward blind source separation algorithm that enhances very noisy speech signals with a subband approach. In this paper, we propose a new variable subband step-sizes algorithm that allows improving the previous algorithm behaviour when the number of subband is selected high. This new proposed algorithm is based on recursive formulas to compute the new variable step-sizes of the cross-coupling filters by using the decorrelation criterion between the estimated sub-signals at each subband output. This new algorithm has shown an important improvement in the steady state and the mean square error values. Along this paper, we present the obtained simulation results by the proposed algorithm that confirm its superiority in comparison with its original version that employs fixed step-sizes of the cross-coupling adaptive filters and with another fullband algorithm.

1. Introduction

Adaptive filtering is frequently employed in signal processing, control, communications and in many other applications because of its simplicity [1–3]. Noise reduction and speech enhancement applications have largely used adaptive filtering algorithm [4–8]. Several adaptive filtering algorithms have been proposed to enhance speech signal and cancel the acoustic noise. From these algorithms we have the recursive lest square (RLS) algorithms and its derived versions [8–10], the fast transversal filters and its derived versions as the fast Newton transversal filter (FNTF) [11–13], the fast sub-sampling Fast Transversal Filter (FTF) and FNTF algorithms [14], the affine projection algorithms and its fast versions [15], etc. However, the most popular adaptive filter is the least-mean square (LMS) algorithm and its normalized version (NLMS) [8–10]. It is well known that the stability of this algorithm is controlled by a step-size parameter and the good choice of this parameter reflects a compromise between fast convergence rate and small misadjustment such as required in most adaptive filtering applications [18].

Recently, a very important amount of papers have investigated the combination of the so called Forward and Backward blind source separation structures (i.e. FBSS and BBSS structures, respectively) with these different algorithms family to improve the behaviour of speech enhancement and noise reduction by these two structures [1,5–7,19–21]. The smart combinations between the two FBSS and BBSS structures and the adaptive filtering algorithm families have given a new insight in the field of acoustic noise reduction and speech intelligibility enhancement. For example in [19], authors of this paper used the Symmetric Adaptive Decorrelation (SAD) algorithm for signal separation. A several algorithms have been proposed and implemented with the two FBSS and BBSS structures [12,16,17]. In [5–8], several two and multi-sensors techniques have been proposed to separate the sources, which are mixed by a convolutive model [22].

Recently, several works have been conducted to increase convergence rates and reducing the steady-state estimation error (low misadjustment) of these different adaptive filtering algorithms in acoustic noise reduction and speech enhancement applications [23–38,1,39]. Furthermore, in the same direction, many variable step-size adaptive filtering algorithms for speech enhancement application have been also proposed [23–38,1,39]. For example, in references [23–26], variable step size LMS (VSS-LMS) algorithms were proposed to improve the performance of the fixed step size LMS algorithm by using particular recursive formulas. Others variable step-size normalized least mean square (NLMS) algorithms have been proposed to resolve the problem of the conventional NLMS algorithm in the steady state regime when a degradation of the final mean square error (MSE) occurs [27–29]. Another useful method to accelerate the convergence speed characteristic of the adaptive algorithm is to use the variable step-size affine

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http://dx.doi.org/10.1016/j.asoc.2016.01.049
1568-4946/

Nomenclature

D	decimation factor
I	interpolation factor
i	index varied between 0 and N-1
k	decimated time index
L	analysis and synthesis filters lengths
M	real and adaptive filters lengths
m	delay index
N	subbands number
n	time index
ε	small positive constant
ξ	small positive constant
α	small positive constant

Abbreviations

FTF	fast transversal filter
FNTF	fast Newton transversal filter
2CFB	two-channel fullband backward
2CSF	two-channel subband forward
AEC	acoustic echo cancellation
ANR	acoustic noise reduction
APA	affine projection algorithm
BSS	blind source separation
FBSS	Forward blind source separation
BBSS	Backward blind source separation
FSS	fixed step-size
IR(s)	impulse response(s)
LMS	least mean squares
NLMS	normalized LMS
2CFNLMS	two-channel Forward NLMS
RLS	recursive least square
SAF	subband adaptive filtering
SAD	symmetric adaptive decorrelating
SM	system mismatch
SNR	signal-to-noise ratio
SegSNR	segmental signal-to-noise ratio
CD	cepstral distance
MSE	mean square error
VAD	voice activity detector
VSS	variable-step-size
dB	decibel
VAD	voice activity detector
SVD	singular value decomposition
DWT	discrete wavelet transform
PWT	packet wavelet transform
E	expectation operator
φ	smoothed parameter

Mixture model

$s(n)$	original speech signal
$b(n)$	noise
$h_{12}(n)$ and $h_{21}(n)$	cross-coupling IRs
$p_1(n)$ and $p_2(n)$	noisy speech signals
$\delta(n)$	Kronecker impulse

Subband Forward structure

$\mathbf{h}_i$	analysis filters
$\mathbf{g}_i$	synthesis filters
$p_{1i,D}(k)$ and $p_{2i,D}(k)$	noisy speech sub-signals
$u_{1i,D}(k)$	estimated speech sub-signals
$u_{2i,D}(k)$	estimated noise sub-signals
$w_{12}(n)$ and $w_{21}(n)$	symmetric adaptive filters
$u_1(n)$	enhanced speech signal
$u_2(n)$	estimated noise
$\lambda_{12,i}$ and $\lambda_{21,i}$	Lagrange multipliers
f_s	sampling frequency

Step-sizes

μ_{12} and μ_{21}	fixed step-sizes
$\mu_{12,\min}$ and $\mu_{21,\min}$	minimal step-sizes
$\mu_{12,\max}$ and $\mu_{21,\max}$	maximal step-sizes
$\mu_{12}(k)$ and $\mu_{21}(k)$	controlled step-sizes
$\psi_{12}(k)$ and $\psi_{21}(k)$	cross-correlation factors
$f_1(\bullet)$ and $f_2(\bullet)$	functions
$\varepsilon_1(k)$ and $\varepsilon_2(k)$	deviation vectors

projection algorithm family (VSS-APAs) [30–32]. However, in all these algorithms, the selected variable step-size can modify the convergence rate and the steady-state mean square error properties (fast convergence rate and low steady-state or MSE).

In addition, the subband and fullband implementation techniques are lastly used in speech enhancement application to accelerate the acoustic noise reduction [33,34]. Another efficient method that is largely used in literature to improve these performances is the use of the Forward and Backward BSS structure combined with recursive and nonrecursive algorithms [35–38,1,39,40]. One of these efficient algorithms is the affine projection algorithm, and its Gauss–Seidel version that is recently proposed to be used in speech enhancement and acoustic noise reduction application [39–41]. In the literature, we can find other speech enhancement algorithms based on SVD techniques [42,43]. Further enhancement possibilities are allowed by using DWT and PWT to transform the noisy signals and to be enhanced in other orthogonal data-basses, these techniques are very promising and lead to important improvement in comparison with the classical speech enhancement and noise reductions algorithms [44–46].

Recently in [1], we have proposed a new two-channel subband implementation of the FBSS structure, this new algorithm is called two-channel subband Forward algorithm and is denoted by 2CSF. We proposed the use of this 2CSF algorithm in two-channel acoustic noise reduction and speech enhancement domain. The major benefit of this algorithm is to improve the convergence behaviour of the classical FBSS in acoustic noise reduction application with less output speech signal distortions. An important parameter of this 2CSF algorithm is the fixed step-size. The stability of the conventional algorithms is controlled by this parameter, which is time-independent. A compromise should be made in the selection of this important control parameter. A tradeoff between small steady state misadjustment (MSE) and fast speed of convergence rate must be made in the choice of the fixed step-size parameter of this 2CSF algorithm. However, A large value of the fixed step-size implies fast convergence rate and tracking, while a small value leads to low misadjustment and good robustness features again residual noise components. For example, in acoustic echo cancellation (AEC) there is a need for all these performance criteria, as a consequence, the step-size should be well controlled.

In this paper we focus on the FBSS structure and their implementation in subband approach (i.e. the 2CSF algorithm). We propose a new two-channel subband FBSS structure with variables step-size property that allows to resolve the problem and the drawbacks of the fixed step-size property of the 2CSF algorithm published in [1]. All the drawbacks of the previous 2CSF algorithm are resolved by the step-size techniques that we propose in this work. In the sequel of this paper, we show the efficiency of the new proposed variable step-size algorithm in comparison with fixed step-size ones, as the two-channel fullband normalized least means square (2CFNLMS)