WIDEBAND ROBUST ADAPTIVE BEAMFORMING VIA TARGET TRACKING

Sofiène Affès *Saeed Gazor* *Yves Grenier*
Télécom Paris, Département Signal, 46 rue Barrault, 75634 Paris Cedex 13, France.
E-mail: Affes@sig.enst.fr, Tel:++331 45817782, Fax:++331 45887935.

ABSTRACT

In this paper, we generalize a former work recently presented in the narrowband case of robust adaptive beamforming via target tracking to the wideband domain. The original algorithm is applied to each frequency component of the signal in an Analysis/Synthesis scheme. The source tracking and localization are simply performed in one frequency selected with the minimum location misadjustment. A more complex combination of location estimates can be computed in a specific set of frequencies, with a relatively better performance. Simulation results confirm in both cases the efficiency of the generalized algorithm regarding source localization and noise reduction.

1. INTRODUCTION

The main purpose of wideband adaptive beamforming is to design a processing multi-input filter of the signals collected by an array of sensors; so as to extract a desired source at a known location in space and spectral band. The filter coefficients of the adaptive beamformer are time-updated respectively to some criteria optimizing the reduction of uncoherent jammers [1-3]. It turns out however, that the performance of adaptive beamforming severely degrades at the presence of pointing errors [1,2].

To overcome the resulting phenomena of signal cancellation, some robust beamformers to location errors were proposed in the literature [4,5]. Their common idea is to maintain a minimum antenna gain over an allowable domain of location errors, at the expense of noise reduction efficiency.

Recently in [3,6], the idea of steering vector searching and DOA estimation has been explored with the method of Correlation-Constrained Minimization of Power (CCMP). In [7,8], we alternatively proposed a robust adaptive beamforming algorithm based on a target tracking procedure in the narrowband case. The performance analysis and simulations made in [7-9] proved the capacity of the algorithm to correct location errors, and even to track mobile sources with an efficient noise reduction.

In this paper, we generalize the proposed algorithm to the wideband case, and illustrate its efficiency by simulation.

The proposed scheme consists of applying the Generalized Sidelobe Canceller (GSC) structure as incorporated in the algorithm of [7,8] to each frequency component of the input signals, using DFT's over blocks of l samples and the Overlap Save (OLS) technique for the synthesis. The steps involving target tracking and source localization in the algorithm of [7,8] are simply performed in one tracking frequency say f_t, where the variance of location error is minimum. All the steering vectors at the remaining frequencies are driven by the estimate of the direction of arrival (DOA) at that frequency component f_t. Another alternative consists of optimally weighting a combination of location estimates in a specific set of frequency components. It is shown to yield a better performance regarding the location misadjustment. The combined estimate improves to some extent with the number of components. However, the set of selected frequencies should be restricted for computational complexity considerations.

The efficiency of the algorithm is confirmed by simulations in both schemes, with real signals recorded in a car acoustic environment.

2. PROBLEM FORMULATION

We consider the following model of wideband plane wave propagating signals received by a linear array at time t (see Figure 1):

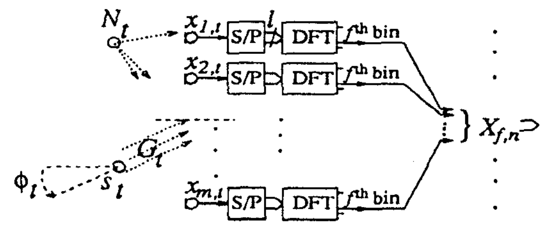

Figure 1. Serial to parallel and transform to the frequency domain of the observation signals.

$$X_t = [s_{t-\tau_{1,t}}, s_{t-\tau_{2,t}}, \cdots, s_{t-\tau_{m,t}}]^T + N_t, \quad (1)$$

$$\tau_{i,t} = \tau_{0,t} + \frac{x_i \sin(\phi_t)}{C} \quad \text{for } i = 1, 2, \cdots, m \ , \quad (2)$$

where X_t is the m-dimensional observation vector, s_t is the desired wideband signal to be extracted, and N_t is an additive zero mean noise vector. C is the celerity and $x_1, x_2, \cdots, x_m$ are the sensor positions. The DOA $\phi_t \in [-\pi/2, \pi/2[$ is assumed to be slowly time varying in comparison to the variations of N and s. $\tau_{0,t}$ and $\tau_{i,t}$ are respectively the time delays from the source to the array origin and the array sensors. As well, $\tau_{i,t}$ $(i = 0, 1, ...m)$, N and s are assumed to be mutually independent.

Let us define the transfer function (i.e. steering vector) $G_t = [\delta_{t-\tau_{1,t}}, \delta_{t-\tau_{2,t}}, \cdots, \delta_{t-\tau_{m,t}}]^T$ between the emitted source s_t and the m-sensor antenna array. On condition that ϕ_t be slowly time varying and almost piecewise constant over a l-sample blocks division[1], the Discrete Fourier Transform (DFT) of G_t can be written as follows:

$$\begin{aligned} G_{f,n} &= F(\theta_n, f) \quad \text{for } f = 0, 1/l, \cdots, (l-1)/l \ , && (3) \\ &= e^{-j2\pi\tau_{0,n} f}[e^{-j\nu_n f x_1}, e^{-j\nu_n f x_2}, \cdots, e^{-j\nu_n f x_m}]^T, \end{aligned}$$

where the subscripts f and n denote the DFT of the indexed quantity respectively at the normalized frequency f and the block iteration n. F is a parameterizing function of the plane wave propagation model where $\theta_n = [\nu_n \ \ \tau_{0,n}]^T$. $\nu_n \triangleq \frac{2\pi \sin(\phi_n)}{C}$ is the wavenumber at the normalized sampling frequency. $2\pi f \tau_{0,n}$ represents the phase delay from the origin to within about an integer multiple of 2π, and is obviously restricted to $[0, 2\pi[$.

Taking now the DFT of the observation signals, we can transpose the problem to the narrowband case as follows (see Figure 1):

$$X_{f,n} = G_{f,n} s_{f,n} + N_{f,n}. \qquad (4)$$

Hence, we are able to apply the algorithm presented in [7,8] at each frequency component, as shown in the following section. In this case, the estimated frequency components allow the synthesis of the desired signal in the time domain, using the OLS technique with overlapping blocks.

We finally assume that a possibly erroneous approximation of θ_0, say $\hat\theta_0$, is initially provided either by an approximate *a priori* guess, or by a given localization technique.

3. WIDEBAND ROBUST ALGORITHM

For wideband robust adaptive beamforming, we can apply the simple algorithm described in the narrowband case to each frequency component of the input signals.

As we stated in the previous section that the DOA ϕ_t is slowly time varying, we can use the steering vector $\hat G_{f,n-1}$ at the block iteration n, to adaptively estimate the desired signal component $s_{f,n}$ at the frequency f. Using the GSC structure, the beamforming unit can be summed up by the following steps (see Figure 2):

$$\begin{aligned} X^s_{f,n} &= \mathrm{diag}[\hat G^H_{f,n-1}]\ X_{f,n}, && (5) \\ X^P_{f,n} &= P\ X^s_{f,n}, && (6) \\ P &\triangleq \begin{bmatrix} 1 & -1 & 0 & \cdots & 0 \\ 0 & 1 & -1 & \cdots & 0 \\ \vdots & \ddots & \ddots & \ddots & \vdots \\ 0 & \cdots & 0 & 1 & -1 \end{bmatrix}_{(m-1)\times m}, \\ \hat s_{f,n} &= \frac{[1, 1, \cdots, 1]\ X^s_{f,n}}{m} - W^H_{f,n} X^P_{f,n}, && (7) \\ W_{f,n+1} &= W_{f,n} - \eta \hat s^H_{f,n}\ X^P_{f,n}. && (8) \end{aligned}$$

Notice in Figure 2 that the estimates of the location parameters should be supplied to the beamforming unit. These parameters are actually estimated in a LMS-like tracking and localization procedure, which corrects the steering vector and adjusts the location parameters adaptively.

However, this procedure described in [7,8] is simply performed for one tracking frequency, say f_t. The corresponding estimates are then forwarded to the beamforming units at the remaining frequencies.

[1] l is assumed to be large compared to the wavefront transit time across the array sensors.

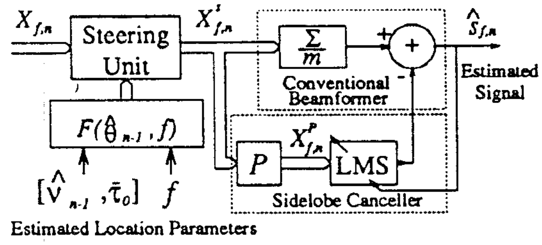

Figure 2. Beamforming unit of the algorithm in the frequency domain with the GSC structure at the frequency f.

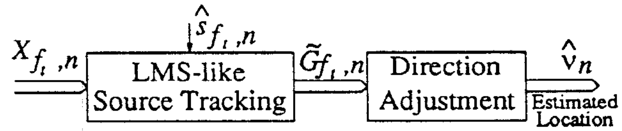

Figure 3. The target location estimation at the tracking frequency f_t.

We will see later in the following section how to select this frequency. We will also show a straightforward extension of the tracking and localization procedure at one frequency to an optimal combination of selected frequencies.

At present, this procedure can be described at the tracking frequency f_t as follows (see Figure 3):

$$\tilde G_{f_t,n} = \hat G_{f_t,n-1} + \mu\ (X_{f_t,n} - \hat G_{f_t,n-1}\hat s_{f_t,n})\hat s^H_{f_t,n}, \qquad (9)$$

$$\hat\theta_{f_t,n} = \hat\theta_{f_t,n-1} - K_{f_t}\begin{bmatrix} \sum_{i=1}^m x_i \mathrm{Im}\left(\tilde G_{f_t,n,i} e^{j\hat\nu_{n-1} f_t x_i}\right) \\ \sum_{i=1}^m \mathrm{Im}\left(\tilde G_{f_t,n,i} e^{j\hat\nu_{n-1} f_t x_i}\right) \end{bmatrix}, \qquad (10)$$

$$K_{f_t} \triangleq \frac{1}{f_t^2}\begin{bmatrix} \sum_{i=1}^m x_i^2 & 2\pi\sum_{i=1}^m x_i \\ 2\pi\sum_{i=1}^m x_i & 4m\pi^2 \end{bmatrix}^{-1}. \qquad (11)$$

In equation (9) denoted by LMS-like, we adaptively track and correct the steering vector into an intermediate variable $\tilde G_{f_t,n}$. This variable is not necessarily parametrized by the propagation modelling function F. Hence, we extract the location parameters in equation (10) in a projection step. It could be interpreted as a linear regression of $\tilde G_{f_t,n}$ components over the sensor positions x_i, where $\hat\nu_{f_t,n}$ and $\hat\tau_{f_t,n}$ are to be considered respectively as the slope and the constant of the required linear relation.

Eventually, we reconstruct the steering vector for the next iteration as follows:

$$\hat G_{f,n} = F\left([\hat\nu_n, \bar\tau_0]^T, f\right), \qquad (12)$$

where $\hat\nu_n$ is equal at present to $\hat\nu_{f_t,n}$. Obviously, it is not possible to identify the time delay $\tau_{0,n}$ in a passive detection of plane wave signals. However, we first estimate it to remove the constant part of the linear regression and properly extract $\hat\nu_n$. Then we set $\hat\tau_{0,n}$ to a constant time delay $\bar\tau_0$, possibly negative, so as to make the steering unit step with $\hat G^H_{f,n}$ causal for the wideband synthesis. It should be noted that this constant does not affect in any way the procedures presented above. In practice, we select the time delay constant $\bar\tau_0 = -\frac{|x_1 - x_m|}{C}$ (we select l s.t. $l >> \bar\tau_0$).

The estimated signals in the frequency domain are finally transformed to the time domain, using an inverse DFT (IDFT) in an OLS synthesis scheme.

4. TRACKING FREQUENCIES SELECTION

In the performance and convergence analyses made in [7,9], we proved that the DOA estimate is unbiased at any frequency. We also computed its covariance in the case of spatially diffuse noise, which is given at the frequency f by:

$$\sigma_f^2 \triangleq \mathrm{Cov}(\hat{\nu}_{f,n}) = f^{-2} \frac{\frac{\sigma_w^2}{2\mu\sigma_{s,f}^2} + \mu\sigma_{N,f}^2 \frac{1+\sigma_{N,f}^2/(m\sigma_{s,f}^2)}{4\sum_{i=1}^{m} x_i^2}}{1-\mu(\sigma_{s,f}^2+\sigma_{N,f}^2/(4m))}, \quad (13)$$

where $\sigma_w^2 = E[|\nu_{n+1} - \nu_n|^2]$ is the variance of the random walk of the location parameter, and where $\sigma_{N,f}^2$ and $\sigma_{s,f}^2$ are respectively the spectral densities of the noise and the source at the frequency f.

Hence, a simple criterion is to select the tracking frequency with the minimum variance σ_f^2 over the whole frequence band. In this case, the expression of the variance in equation (13) shows that this frequency can be selected respectively to some different conditions.

For instance, f_t can be chosen in a frequency band where the signal to noise ratio (SNR) is high enough, and where the energy of the source is omnipresent in time. It can be also chosen in the high frequency band where the antenna aperture and σ_f^2 are smaller. This allows the separation of close sources in the case of double talk, or in the presence of close jammers. Alternatively, it can be rather selected in the low frequency band where the locking range is higher, to enable the correction of relatively high DOA errors.

A weighted optimal combination of DOA estimates in a set of tracking frequencies can be performed to improve the localization.

To do so, we first assume that the localization errors at the different frequencies are independent. In this case, a linear optimal combiner can reduce the covariance of the estimated location parameter ν_n as follows:

$$\hat{\nu}_n = \frac{\sum_{f_t} \sigma_{f_t}^{-2} \hat{\nu}_{f_t,n}}{\sum_{f_t} \sigma_{f_t}^{-2}}, \quad (14)$$

where f_t possibly ranges over the whole frequency band. It is then straightforward to show the following inequality:

$$\mathrm{Cov}(\hat{\nu}_n) = \frac{1}{\sum_{f_t} \sigma_{f_t}^{-2}} \leq \sigma_{f_t}^2 \quad \forall\ f_t\ . \quad (15)$$

More complex criteria taking into account the spectral nonstationarities in time of the source and noise signals can be introduced, to optimally combine the location estimators over the frequency band. In such case, the estimates are combined so as to continuously make the best estimators at a given block compensate for the worst, and vice versa.

Equation (15) shows that the covariance of the combined location estimate decreases with the number of tracking frequencies. This enhancement in location misadjustment is however limited to some extent, basically at the low frequencies where σ_f^2 is relatively high. Hence, the significant increase in computational load is not worth selecting a high number of tracking frequencies, for a substantial improvement in location errors. In this case, a suboptimal combination of location estimates in a limited set of tracking frequencies could be made, with a satisfactory resolution of localization and a reasonable computational load.

5. SIMULATION RESULTS

To illustrate the efficiency of the present algorithm, we consider a linear equidistant antenna of 8 sensors and a planar waves propagation model of speech sources. The desired signal is a speech sentence of a female speaker repeated twice (see Figure 7-a). It is corrupted at a mean SNR of 5 dB, by a car noise recorded at a standstill with the engine on (see Figure 7-b). All the signals are lowpass filtered to $f_{max} = 4$ KHz, then sampled at $f_s = 8$ KHz. The spacing between sensors is $d = \frac{\lambda_{max}}{2}$, and the tracking frequency is selected at $f_t \simeq 2$ KHz.

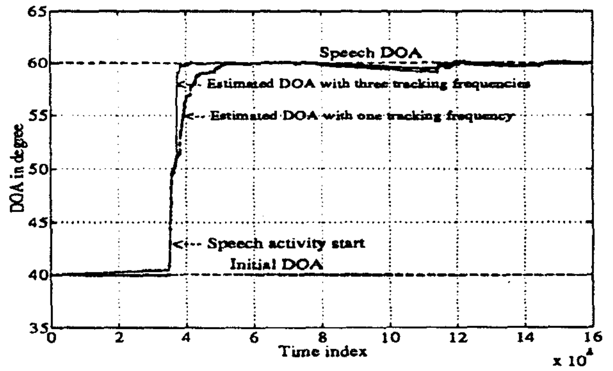

Figure 4. The proposed algorithm corrects the error, while the classical GSC maintains the steering at the erroneous DOA.

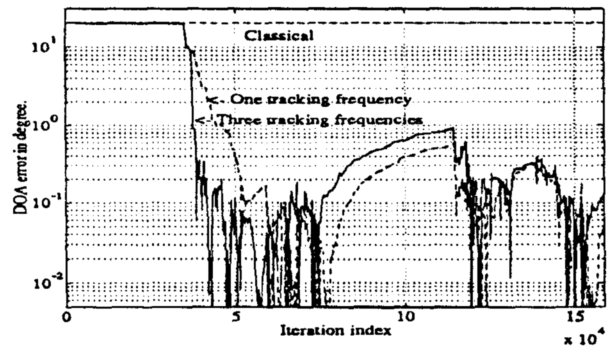

Figure 5. As soon as speech is uttered, location errors are reduced to a very small range of 10^{-2} deg.

Started with a DOA error as high as 20 deg, Figure 4 shows that the proposed algorithm is able to localize the speaker and to maintain the beamformer in the speech DOA as soon as the speech is uttered. Figure 5 shows that localization error is reduced to a range of 10^{-2} deg and remains stable in periods of silence.

The high resolution of localization gained by the proposed algorithm enables the beamformer to properly extract the speech source as shown in Figure 6-d. On the other hand, classical GSC maintains the steering in the erroneous DOA and causes a severe degradation of the speech as it is considered as a jammer (see Figure 6-c).

We notice in Figure 7 that both algorithms have the same performance during the first period of silence. Total distortion increases during speech utterance, due to the beamformer adaptation to the nonstationarities of the signal. In the case of classical GSC, the array gain is amplified outside

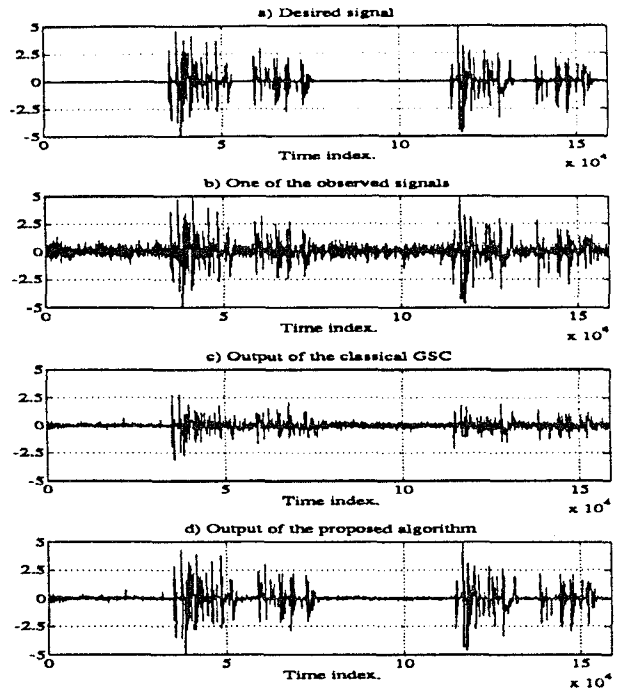

Figure 6. a: original speech, b: noisy speech at the 4^{th} sensor, c: output of classical GSC, d: output of proposed algorithm.

the speech DOA, so as to cancel the speech and to maintain a unit gain in the erroneous direction. This has the effect of degrading the noise reduction even in periods of silence, while the proposed algorithm maintains the same performance.

We finally tested the algorithm in the case of multiple tracking frequencies. We selected three frequencies at 2.03, 2.16 and 2.29 KHz. Figures 4 and 5 show that the performance of localization improves regarding the misadjustment and convergence time. This is actually due to the diversity brought by the estimators at the different frequencies. However, we notice that signal distortion and noise reduction are not enhanced significantly with the tracking in multiple frequencies, as the localization in one simple frequency is high enough.

6. CONCLUSIONS

We described in this paper the generalization we have made of a former work recently presented on robust adaptive beamforming via target tracking, to the wideband case.

This generalization is performed in the frequency domain, using DFT's and the OLS technique in an Analysis/Synthesis scheme. Robust adaptive beamforming is applied at each frequency bin, to estimate the source signal components at the corresponding frequencies. In the first place, source tracking and DOA estimation are simply performed in one tracking frequency. The corresponding DOA estimate is then forwarded to each steering unit, to adaptively drive the beamformers at the different frequency components in a master-slave structure.

We also gave the expression of misadjustment at a given frequency, and confirmed some of the intuitive expectations regarding the best choice of the tracking frequency. In addition, we proved the tracking step in a selected set of frequencies to improve the location estimation, to some extent however, due to the resulting increase in computational load.

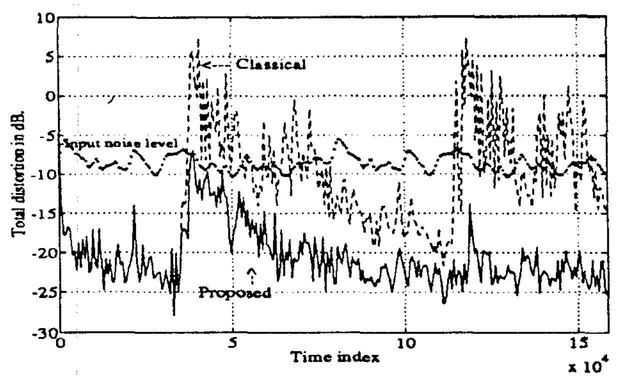

Figure 7. Total distortion in dB $E[|s_t - \hat{s}_t|^2]$, for classical GSC and the proposed algorithm.

We finally illustrated by simulations the efficiency of the generalized algorithm in source localization and noise reduction, in a car acoustic environment.

At present, we are studying the generalization to the near field propagation model, successfully tested in the narrowband case. In parallel, we are investigating about the capacity of the present algorithm to be directly implemented in the time domain.

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