An Iterative Blind Receiver : Implementation and Performance

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Summary
In this paper, we introduce a new blind MIMO communication scheme allowing the transmission over a frequency selective channel based on the deflation procedure described in [6]. The propose method is based on an iterative receiver, we introduce an error correcting code and we propose a new structure of a receiver. It consists in extracting symbols emitted by sources. Our scheme uses 2 transmit antennas and K (K>2) received antennas. This kind of blind system is well suited for many applications of communication network such as acoustic network, neural network, ... and where we do not have a strong constraint for the number of antennas.

Key words:
MIMO system, Deflation, Source separation, turbo decoding, communication network.

1. Introduction

The blind source separation problem has recently become an intense area of research in many situations of practical interest and is receiving increasing attention in both signal processing and neural network [1]. It consists in extracting emitting symbols of independant sources from only an observed linear mixture of them. To solve this problem, many algorithms have been proposed. The first one was proposed in 1985 [1] and based on concellation of higher order moments. Other criteria are based on the minimization of cost functions, such as contrast function [2, 3], or likelihood function [4, 5].

We introduce in this paper a new iteratif process based on the deflation approach [11, 12] where the separation is achieved by the contrast function : Constant Modulus Algorithm (CMA). Therefore, we consider a MIMO communication system using 2 transmit antennas and K received antennas (K>2). The received signal $y(t) = y_1(t) + y_2(t)$ is made up of the contribution of the signal $y_p(t)$ emitted by antenna p {p=1,2}. In the telecommunication framework, $y_p(t)$ is a convolutive mixture shown in equation (1) :

$$y_p(t) = \sum_k s_p(k) h_p(t - kT) \qquad (1)$$

Manuscript received September 5, 2007
Manuscript revised September 20, 2007

where h_p denotes the impulse response of the channel relative to antenna p, i.e. a filter stemming from the cascade of a band-limited pulse-shaping filter and multi-path transmission effects, $(s_p(k))_k$ is an i.i.d. sequence of discrete Q-PSK symbols and T is the symbol period. At the receiver side, y is observed through an array of K sensors.
The mean idea of our method developed in this paper consists in the following : the emitter has 2 transmit antennas, each one send a sequence of bits. At the receiver side, the signal sent by each antenna can be estimated by applying blind source separation techniques. We based our receiver on the deflation procedure [10, 11]. The deflation allows us to estimate successively the symbols sent by each transmit antenna.
Now, we consider a product code word C of size $N_1 \times N_2$ obtained from two elementary codes $C_1(N_1, K_1, \delta_1)$ and $C_2(N_2, K_2, \delta_2)$ where (N_i, K_i, δ_i) is the length of the code word C_i, the length of information word and the minimum distance of the code word C_i [7, 14, 15]. We divide vertically the code word C_i into two blocks as depicted at figure (Fig. 2).
The first one is sent on antenna 1 and the second one is sent on antenna 2. Let's note $S_i(n) = [s_i(nN), ..., s_i(nN + N - 1)]^T$ with $N = \left\lfloor \frac{N_1 \times N_2}{2} \right\rfloor$ the sequence of symbols sent by antenna i.

The extraction of $S_i(n)$ is done by applying thedeflation approach [10, 11] based onthe minimization of a contrast function [12, 13]. We recall that a contrast function is a function of the statistics of the received signal and its minimization allows the extraction of one source.

We develop in this paper a method based on the deflation approach and the turbo decoding procedure allowing the estimation of the emitted symbols.Therefore, Section 2 describes our iterative procedure and the principle of our symbol estimation procedure. We evaluate our method performance in section 3.

2. Our Method Description

Our method is based on two process : the deflation procedure and the turbo decoding principle [15, 16]. First of all, let us describe our iterative procedure.

2.1 Description of the iterative procedure

While having two sources, our method is splitted into two stages. Each stage allows us the estimation of symbols of one source. Our method is based on the deflation procedure which is the key-stone of our reception scheme :

Stage 1 :

1) The observation y is sampled at rate 1/T and passed through a digital K×1 vector-filter $G_1(z)$. The coefficients of $G_1(z)$ are adapted by the minimization of a contrast function. The minimum is reached if and only if the sequence $Z_1(n)=\left[z_1(nN),\ldots,z_1(nN+N-1)\right]^T$ obtained at the output of $G_1(z)$ equals the symbols of one of the antenna up to a delay and a complex multiplicative factor. Let us assume that $Z_1(n)$ is an estimation of the sequence $S_1(n)$.

2) By an adaptive subtraction procedure, we compute from $Z_1(n)$ the contribution of $S_1(n)$ on each captor and we subtract this contribution from the observed signal. More precisely, we search $t_1(z)$ of size K×1 that minimizes the following expression:

$$\mathrm{E}\left[\left\|y(n)-\left[t_1(z)\right]z_1(n)\right\|^2\right] \quad (2)$$

$t_1(z)$ is an estimate of the channel impulse response corresponding to the first estimated sequence of symbols. And $y'(n)=y(n)-\left[t_1(z)\right]z_1(n)$ is a signal of dimension K corresponding to the contribution of the sequence $S_2(n)$.

Stage 2 :

We iterate stage 1 but on the signal $y'(n)$ in order to adapt a filter $G_2(z)$, to estimate the sequence $S_2(n)$ and the corresponding channel impulse response.

At each stage, we compute from $Z_k(n)$ an extrinsic information that ca be used as a priori information at stage 1 of the next iteration. Then, we decode iteratively the computed posteriori information

Therefore, the first iteration of our method consists in the deflation procedure with the computation of the respective extrinsic information and the turbo decoding of the posteriori information at each stage of our receiver.

At the end of this first iteration, we obtain the signal $e^{(1)}(n)$ (the upper-script (m) stands for iteration number m) :

$$e^{(1)}(n)=y(n)-\left[t_1^{(1)}(z)\right]z_1^{(1)}(n)-\left[t_2^{(1)}(z)\right]z_2^{(1)}(n)$$

that can be considered as a residual signal containing the part of the signal not exploited during the first iteration. In order to start the next iteration and to re-extract the first sequence of symbols, we add to $e^{(1)}(n)$ the estimated contribution of S_1 at the previous iteration (i.e. $\left[t_1^{(1)}(z)\right]z_1^{(1)}(n)$). Once the symbols of the source are estimated (using the a priori information of the previous iteration), we estimate $t_1^{(2)}(z)$ and subtract the contribution of this source and add the previous estimate of the sequence S_2 (as shown by figure Fig. 1) before starting stage 2 of iteration 2.

2.2 Estimation of Symbol Sequence

Figure (Fig. 1) shows the principle of our method at iteration m.

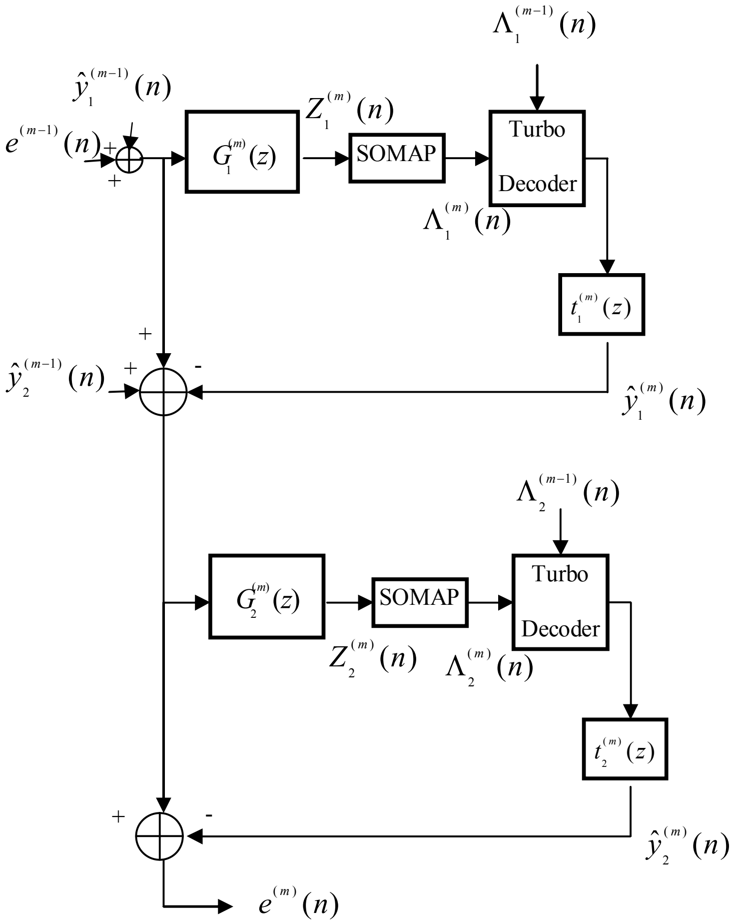

Fig. 1 : Our Method Principle

As shown at figure (Fig. 1), we note $G_k^{(m)}(z)$ the Multiple Input Single Output equalizer adapted by the CMA [11] at stage k, {k=1,2}, of iteration m. The output $Z_k^{(m)}(n)$ of $G_k^{(m)}(z)$, is an estimation of one of the sequence of symbols (i.e. S_1 or S_2) up to a delay and a multiplicative factor. We note the following assumptions:

i) The delay and the multiplicative factors are known by the receiver.
ii)The receiver knows the source currently estimated.

Without restriction, we assume that the sequence $S_1(n)$ is extracted at the first stage. Therefore, we have the following equation :

$$Z_k^{(m)}(n) = S_k(n) + b_{k,m}(n) \tag{3}$$

where:
$Z_k^{(m)}(n) = \left[z_k^{(m)}(nN), \ldots, z_k^{(m)}(nN+N-1)\right]^T$ and $b_{k,m}(n)$ is Gaussian White Noise vector of length N with zero mean and unknown variance $v_{k,m}^2$. Note that the variance $v_{k,m}^2$ of the noise can be empirically estimated by:

$$\hat{v}_{k,m}^2 = \frac{1}{N}\sum_{n=1}^{N}\left(z_k^{(m)}(n) - \hat{s}_k^{(m)}(n)\right)^2 \tag{4}$$

where $\hat{s}_k^{(m)}(n)$ is the hard decision of the symbol $z_k^{(m)}(n)$. $\hat{v}_{k,m}$ measures the reliability of the estimated symbols.

We consider Q-PSK symbols in this paper. Therefore, we treat separately their real and imaginary part. The problem now consists in estimating a vector $U_1(n) = \left[\Re\left(S_1(n)\right), \Im\left(S_1(n)\right)\right]^T$ of size $2.N \times 1$ with value in {+1,-1}, where $\Re(x)$ (respectively $\Im(x)$) denotes the real (respectively imaginary) part of x. Let us consider the iteration m. The estimated $\hat{U}_1^{(m)}(n)$ of $U_1^{(m)}(n)$ is obtained by [14] :

$$\hat{U}_1^{(m)}(n) = \tanh\left(\frac{L_1^{(m)}\left(U_1(n) \mid Z_1^{(m)}(n)\right)}{2}\right) \tag{5}$$

where $L_1^{(m)}\left(U_1(n) \mid Z_1^{(m)}(n)\right)$ is *a posteriori* information, the Log-likelihood ratio of $U_1(n)$ computed as follows :

$$L_1^{(m)}\left(U_1(n) \mid Z_1^{(m)}(n)\right) = L_1^{(m)}\left(U_1(n)\right) + \Lambda_1^{(m)}(n)$$

where $\Lambda_1^{(m)}(n)$ is the extrinsic information estimate of the sequence $U_1(n)$. By equation (3), $Z_1^{(m)}(n)$ may be viewed as the output of an equivalent AWGN channel. Thus, we have :

$$\Lambda_1^{(m)}(n) = \left[\frac{2\Re\left(Z_1^{(m)}(n)\right)}{v_{1,m}^2}, \frac{2\Im\left(Z_1^{(m)}(n)\right)}{v_{1,m}^2} \right]^T . \quad (6)$$

$L_1^{(m)}\left(U_1(n)\right)$ is the estimate at iteration m of the *a priori* information of $U_1(n)$ and $\Lambda_1^{(m)}(n)$ is an extrinsic information of $U_1(n)$.

Now, we have to compute $L_1^{(m)}\left(U_1(n)\right)$.

As $S_1(n)$ and $S_2(n)$ do not send the same information, the *priori* information is the *extrinsic* information computed at stage 1 of the previous iteration. Then, we have the following expression :

$$L_1^{(m)}\left(U_1(n)\right) = \Lambda_1^{(m-1)}(n) \quad (7)$$

Finally, we decode iteratively [8, 9] the computed *posteriori* information.

At the next stage, we proceed as above to estimate $S_2(n)$.

3. Simulation Results

In this section, we consider a Q-PSK modulation. The symbols transmitted are coded by a product turbo code of size $N_1 \times N_2$. We have chosen the same elementary code (i.e. BCH(64,57,4)) for the lines and the columns. While having two transmit antennas, each one transmit a part of product code word as depicted in figure (Fig. 2).

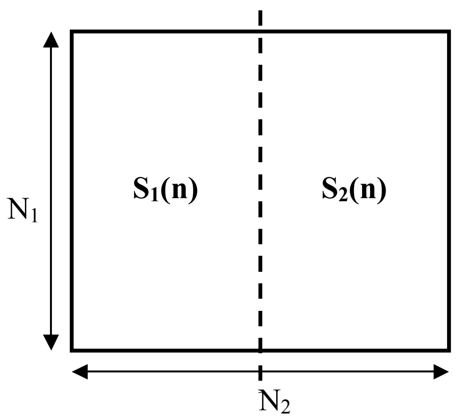

Fig. 2 : Our Method Principle

The considered contrast function is the Godard criterion and $G_k^{(m)}(z)$ is obtained as follows :

$$G_k^{(m)}(z) = ArgMin \frac{1}{N} \sum_{n=0}^{N-1} \left(\left| z_k^{(m)}(n) \right|^2 - 1 \right)^2 . \quad (8)$$

The filter G has a fixed number of taps set to 5 and the minimization is done by the Newton algorithm. We now specify the propagation model. The excess bandwidth factors are all equal to 0.2. The propagation channel results from the superposition of 3 paths : the delays and directions of arrival and directions of departure are uniformly chosen in respectively $[0;3.T]$ and $[0;2\pi]$. The attenuation of the path follows a Rayleigh distribution. The distance between the received antennas is $\frac{\lambda}{2}$ where λ is the wave length of the signal. The number of received antennas set to K=3.

All the following results have been averaged over 1000 trials, where the channel and the symbol sequences are randomly generated for each new trial. The method is tested in a noisy context. The noise is assumed Gaussian, complex, temporally and spatially white with zero mean.

Our method performance is measured by the residual Bit Error Rate (BER) at the output of each iteration of our method. Figure (Fig. 3) shows the performances of the estimated symbols. We note that at each stage of an iteration, we have 4 iterations of turbo decoding.

Figure (Fig. 3) shows that the BER of our method decreases considerably while the number of iterations proceeds. Furthermore, curves show a slope changing while the iterations increases.

Therefore, our method provides an important turbo effect. In addition, the gain summarized in table (Tab. 1) shows the convergence of our method.

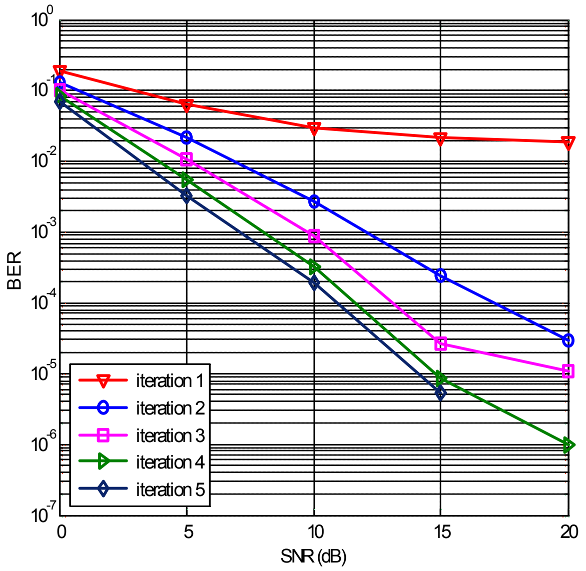

Fig. 3 : Our Method performance : BER versus SNR

Tab. 1 : The gain between iterations

Iteration	1^{st} — 2^{nd}	2^{nd} — 3^{rd}	3^{rd} — 4^{th}	4^{th} — 5^{th}
Gain	> 10 dB	4 dB	1 dB	≈ 0.5 dB

4. Conclusion

In this paper, we have developed a new method allowing us the estimation of symbols emitted by sources of a MIMO system. Our receiver based on the deflation procedure and the turbo decoding process consists in extracting and decoding iteratively symbols of the two sources. The evaluation of our method over a frequency selective fading channel shows good results in term of good estimating emitted symbols.

The main contribution of this paper consists in introducing a new iteratif receiver of estimating symbols where the two sources do not transmit the same information. Our new scheme can be used in many applications of communication network.

References

[1] C. Jutten and J. Hcrault, "Détection de grandeurs primitives dans un message composite par une architecture de calcul neuromimetrique en apprentissage non supervisk", in *Proc. Gretsi,* (Nice), 1985.

[2] P. Comon, "Independant component analysis, a new concept ? ", Signal processing, vol. 3, pp. 287-314, 1994.

[3] E. Moreau and O. Macchi, "New self-adaptive algorithms for source separation bsed on contrast functions", in Proc. IEEE SP Workshop on Higher-Order Stat., Lake Tahoe, USA, 1993.

[4] A. Belouchrani and J.-F. Cardodo, "Maximum likelihood source separation for discrete sources", in Proc. EUSIPCO, pp. 768-771, 1994.

[5] A. Belouchrani and J.-F. Cardodo, "Maximum likelihood source separation by the exceptation maximization technique : Deterministic and stochastic implementation", in proc. International Symposium on Nonlinear Theory and its Applications, Nolta'95, Las vegas, Nevada, USA, Dec. 1995.

[6] S.Houcke and O.Benrhouma, "Transmit Diversity over frequency selective fading channel : A blind approach", EUSIPCO 2004.

[7] C. Berrou, A.Glavieux and P. Thitimajshima, "Near Shannon limit error-correcting coding and decoding : Turbo-codes (1)", IEEE Int. Conf. on Comm.ICC'93, vol. 2/3, May 1993, pp. 1064-1071.

[8] S. M. Reddy, "On decoding iterated codes", IEEE Trans. Inform. Theory, vol. IT-16, Sept 1970, pp. 624-627.

[9] R. Pyndiah, A. Glavieux, A. Piccart and S. Jacq, "Near optimum decoding of products codes, in proc. of IEEE GLOBECOM'94, vol. 1/3, Nov-Dec.1994, pp. 339-343.

[10] P. Loubaton and P. Regalia, "Blind Deconvolution of multivariate signals : A deflation approach", IEEE International Conference on Communications, vol. 2, pp. 1160-1164, October 1993.

[11] N. Delfosse and P. Loubaton, "Adaptive Blind Separation of Independant source : A deflation approach", Signal Processing, 45, PP.59-83, 1995.

[12] P.Comon, "Contrast for multichannel blind deconvolution", IEEE Signal Processing Letters", vol.3, N°7, pp. 209-211, 1996.

[13] S. U. H. Qureshi, "Adaptive Equalization", Proc. IEEE, vol. 73, pp.1349-1387, Sept. 1985.

[14] D. Chase, "A class of algorithms for decoding block codes with channel measurement information", IEEE Trans. Inform. Theory, vol. IT-18, Jan. 1972, pp.170-182.

[15] S. Lin and D. J. Costello, Jr. "Error Control Coding : Fondamentals and Applications", 2nd Edition, Prentice Hall : Englewood Cliffs, NJ, 2005
[16] K. R. Parthasarthy, "Quantum Error Correcting Codes and Information Theory", Jan. 2006

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