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# A Gradient-Based Optimum Block Adaptation ICA Technique for Interference Suppression in Highly Dynamic Communication Channels

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The fast fixed-point independent component analysis (ICA) algorithm has been widely used in various applications because of its fast convergence and superior performance. However, in a highly dynamic environment, real-time adaptation is necessary to track the variations of the mixing matrix. In this scenario, the gradient-based online learning algorithm performs better, but its convergence is slow, and depends on a proper choice of convergence factor. This paper develops a gradient-based optimum block adaptive ICA algorithm (OBA/ICA) that combines the advantages of the two algorithms. Simulation results for telecommunication applications indicate that the resulting performance is superior under time-varying conditions, which is particularly useful in mobile communications.

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## 1. INTRODUCTION

Independent component analysis (ICA) is a powerful statistical technique that has a wide range of applications. It has attracted huge research efforts in areas such as *feature extraction* [1], *telecommunications* [2–4], *financial engineering* [5], *brain imaging* [6], and *text document analysis* [7]. ICA can extract statistically independent components from a set of observations that are linear combinations of these components.

The basic ICA model is  $\mathbf{X} = \mathbf{AS}$ . Here,  $\mathbf{X}$  is the observation matrix,  $\mathbf{A}$  is the mixing matrix, and  $\mathbf{S}$  is the source signal matrix consisting of independent components. The objective of ICA is to find a separation matrix  $\mathbf{W}$ , such that  $\mathbf{S}$  can be recovered when the observation matrix  $\mathbf{X}$  is multiplied by  $\mathbf{W}$ . This is achieved by making each component in  $\mathbf{WX}$  as independent as possible. Many principles and corresponding algorithms have been reported to accomplish this task, such as *maximization of nongaussianity* [8, 9], *maximum likelihood estimation* [10, 11], *minimization of mutual information* [12, 13], and *tensorial methods* [14–16].

The Newton-based fixed-point ICA algorithm [8], also known as the fast-ICA, is a highly efficient algorithm. It typically converges within less than ten iterations in a stationary environment. Moreover, in most cases the choice of the learning rate is avoided. However, when the mixing matrix is highly dynamic, fast-ICA cannot successfully track the time

variation. Thus, a gradient-based algorithm is more desirable in this scenario.

The previously reported online gradient-based algorithm [17, page 177] suffers from slow convergence and difficulty in the choice of the learning rate. An improper choice of the learning rate, which is typically determined by trial and error, can result in slow convergence or divergence. In the adaptive learning and neural network area, many research efforts have been devoted to the selection of learning rate in an intelligent way [18–23]. In this paper, we propose a gradient-based block ICA algorithm OBA/ICA, which automatically selects the optimal learning rate.

ICA has been previously proposed to perform blind detection in a multiuser scenario. In [2, 24], Ristaniemi and Joutsensalo proposed to use fast-ICA as a tuning element to improve the performance of the traditional RAKE or MMSE DS-CDMA receivers. Other techniques exploiting antenna diversity have also been presented for interference suppression [25, 26] or multiuser detection [27]. These ICA-based approaches have attractive properties, such as near-far resistance and little requirement on channel parameter estimation. In this contribution, the new OBA/ICA algorithm is applied for baseband interference suppression in diversity BPSK receivers. Simulation results confirm OBA/ICA's effectiveness and advantage over the existing fast-ICA algorithm in highly dynamic channels. Naturally, OBA/ICA is still useful for slowly time-varying or stationary channels.

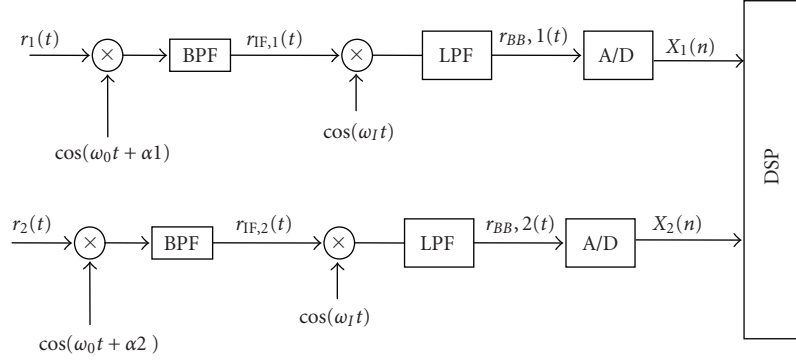


FIGURE 1: Diversity BPSK wireless receiver structure with ICA interference suppression.

The rest of the paper is organized as follows. Section 2 presents the system model for diversity BPSK receiver structure. Section 3 discusses the motivation and basic strategy of OBA/ICA. Section 4 formulates OBA/ICA, and it is also shown that OBA/ICA reduces to online gradient ICA in the simplest case. Section 5 deals with several practical implementation issues regarding OBA/ICA. Section 6 applies OBA/ICA for interference suppression in mobile communications assuming two different types of time-varying channels, and the performance is compared with fast-ICA. Finally, conclusions are given in Section 7.

## 2. SIGNAL MODEL FOR DIVERSITY BPSK RECEIVERS

Figure 1 shows the simplified structure of a dual-antenna diversity BPSK receiver. We assume the image signal is the primary interferer to be suppressed. The extension to the cases of multiple interferers and/or cochannel interference (CCI) is straightforward, and it is accomplished by the addition of antenna elements. For each receiver processing chain, the received signal is first downconverted from RF to IF, followed by a bandpass filter to perform adjacent channel suppression. Then, the IF signal  $r_{IF}(t)$  is downconverted to baseband and lowpass filtered. The baseband signal  $r_{BB}(t)$  is digitized to obtain the signal observation  $\mathbf{X}(\mathbf{n})$ , which is fed into the digital signal processor (DSP) for further processing.

In our signal analysis, frequency-flat fading is assumed. For the  $k$ th antenna ( $k = 1, 2$ ), the channel's fading coefficients for the desired signal  $s(t)$  and the image signal  $i(t)$  are defined as

$$\begin{aligned} f_{sk} &= \alpha_{sk} e^{j\psi_{sk}}, \\ f_{ik} &= \alpha_{ik} e^{j\psi_{ik}}, \end{aligned} \quad (1)$$

where  $\alpha_{sk}$ ,  $\alpha_{ik}$  and  $\psi_{sk}$ ,  $\psi_{ik}$  are the channel's amplitude and phase responses, respectively. The distributions of  $\alpha_{sk}$  and  $\alpha_{ik}$  are determined by the type of fading channels the signals encounter. Since the signals travel random paths,  $\psi_{sk}$  and  $\psi_{ik}$  can be modeled as uniformly distributed random phases over the interval  $[0, 2\pi)$ .

The received signal from the  $k$ th antenna,  $r_k(t)$ , can be expressed as

$$r_k(t) = 2 \operatorname{Re} \{s(t) f_{sk} e^{j(\omega_0 + \omega_I)t} + i(t) f_{ik} e^{j(\omega_0 - \omega_I)t}\}, \quad (2)$$

where  $\operatorname{Re}\{\cdot\}$  denotes the real part of a signal,  $\omega_0$  and  $\omega_I$  denote the frequency of the first and the second local oscillators (LO). The multiplication by 2 is introduced for convenience.

After the RF-IF downconversion, the bandpass filtered signal is given by

$$\begin{aligned} r_{IF,k}(t) &= s(t) f_{sk} e^{-j\alpha} e^{j\omega_I t} + s^*(t) f_{sk}^* e^{j\alpha} e^{-j\omega_I t} \\ &+ i(t) f_{ik} e^{-j\omega_I t} e^{-j\alpha} + i^*(t) f_{ik}^* e^{j\omega_I t} e^{j\alpha}, \end{aligned} \quad (3)$$

where the superscript  $*$  denotes complex conjugate, and  $\alpha$  is the phase difference between the received signal and the first LO signal.

The baseband signal after downconversion to baseband and lowpass filtering is expressed as

$$r_{BB,k}(t) = \operatorname{Re} \{s(t) f_{sk} e^{-j\alpha}\} + \operatorname{Re} \{i(t) f_{ik} e^{-j\alpha}\}. \quad (4)$$

For BPSK signals,  $s(t)$  and  $i(t)$  are real-valued, so (4) can be written as

$$r_{BB,k}(t) = a_k s(t) + b_k i(t), \quad (5)$$

where the coefficients  $a_k = \operatorname{Re}\{f_{sk} e^{-j\alpha}\}$ , and  $b_k = \operatorname{Re}\{f_{ik} e^{-j\alpha}\}$ .

Thus, after A/D converter, the baseband observation is

$$X_k(n) = a_k s(n) + b_k i(n). \quad (6)$$

Each of  $s(n)$ ,  $i(n)$ , and  $X_k(n)$  in (6) represents a one sample signal. Since the signals are processed in frames of length  $N$ ,  $s_N$ ,  $i_N$ , and  $X_{N,k}$  are used to represent frames of  $N$  successive samples. Hence,

$$X_{N,k} = a_k s_N + b_k i_N. \quad (7)$$

Therefore, the baseband signal observation matrix is expressed as

$$X = \begin{bmatrix} X_{N,1} \\ X_{N,2} \end{bmatrix} = \begin{bmatrix} a_1 & b_1 \\ a_2 & b_2 \end{bmatrix} \begin{bmatrix} s_N \\ i_N \end{bmatrix} = AS. \quad (8)$$

In system model (8),  $X$  is the 2 by  $N$  observation matrix,  $A$  is the unknown 2 by 2 mixing matrix, and  $S$  is the 2 by  $N$  source signal matrix, which is to be recovered by ICA algorithm based on the assumption of statistical independence between the desired signal and the interferer. From the above derivation process, it is clear that the mixing matrix is determined by the wireless channel's fading coefficients, which are often time varying. ICA requires that the mixing matrix should be nonsingular, and this is guaranteed due to the randomness of the wireless channel. ICA poses no requirement regarding the relative strength of the source signals, so the operating range for input signal-to-interference ratio (SIR) is quite large. However, in practice, if the interference is too strong, the front-end synchronization becomes problematic. Therefore, there are practical limitations to the application of the proposed technique.

ICA processing has the inherent order ambiguity. Therefore, reference sequences need to be inserted into source signals for the receiver to identify the desired user. Fortunately, in most communication standards, such reference sequences are available.

In this paper, we are primarily concerned about the interference-limited scenario. Therefore, thermal noise is not explicitly included in the signal model. However, ICA algorithm is able to perform successfully in the presence of thermal noise. In Section 6, simulation results will be presented with thermal noise included.

### 3. BACKGROUND AND MOTIVATIONS

The fast-ICA algorithm is a block algorithm. It uses a block of data to establish statistical properties. Specifically, the "expectation" operator is estimated by the average over  $L$  data points, where  $L$  is the block size [8]. The performance is better when the estimation is more accurate, that is,  $L$  is larger. However, it is very important that the mixing matrix stays approximately constant within one processing block, that is, quasistationary. Thus, the problem with convergence arises when the mixing matrix is rapidly time varying, in which case a large  $L$  violates the assumption of quasistationarity.

On the other hand, the online gradient-based algorithm, which updates the separation matrix once for every received symbol, can better track the time variation of the mixing matrix. But it directly drops the "expectation" operator, which results in worse performance than a block algorithm.

Therefore, an algorithm is needed that can better accommodate time variations by processing signals in blocks and automatically selecting the optimal convergence factor. In the following section, such a technique is developed, which is denoted OBA/ICA.

The idea is to tailor the learning rates in a gradient-based block algorithm to each iteration and every coefficient in the

separation matrix, in order to maximize a performance function that corresponds to a measure of independence. In [28], Mikhael and Wu used a similar idea to develop a fast block-LMS adaptive algorithm for FIR filters, which proved to be useful, especially when adapting to time-varying systems.

### 4. FORMULATION OF OBA/ICA

The algorithm developed here is used for estimating one row,  $w$ , of the demixing matrix  $W$ . The algorithm is run for all rows. The performance function adopted is the absolute value of kurtosis. Other ICA-related operations, such as mean centering, whitening, and orthogonalization, are identical as fast-ICA. First, the following parameters are defined:

- (i)  $j$ : iteration index,
- (ii)  $M$ : number of observations,
- (iii)  $L$ : length of the processing block,
- (iv)  $\underline{w}(j) = [w_1(j), w_2(j), \dots, w_M(j)]^T$ : the current row of the separation matrix for the  $j$ th iteration. ( $i = 1, 2, \dots, M$ ),
- (v)  $x_{l,i}(j)$ : the  $i$ th signal in the  $l$ th observation data vector for the  $j$ th iteration. ( $l = 1, 2, \dots, L$ ),
- (vi)  $\underline{X}_l(j) = [x_{l,1}(j), x_{l,2}(j), \dots, x_{l,M}(j)]^T$ :  $l$ th signal observation for the  $j$ th iteration,
- (vii)  $[G]_j = [X_1(j), X_2(j), \dots, X_L(j)]^T$ : observation matrix for the  $j$ th iteration.

The  $l$ th kurtosis value for the  $j$ th iteration is

$$\text{kurt}_l(j) = E \left\{ \left[ \underline{w}^T(j) \underline{X}_l(j) \right]^4 \right\} - 3, \quad (9)$$

where it is assumed that the signals and  $\underline{w}(j)$  both have been normalized to unit variance.

Then, the kurtosis vector for the  $j$ th iteration is

$$\underline{\text{kurt}}(j) = [\text{kurt}_1(j), \text{kurt}_2(j), \dots, \text{kurt}_L(j)]^T. \quad (10)$$

Now the updating formula can be written in a matrix-vector form as

$$\underline{w}(j-1) = \underline{w}(j) - [\text{MU}]_j \nabla_B(j), \quad (11)$$

where

$$\begin{aligned} \nabla_B(j) &= \frac{\partial \{\underline{\text{kurt}}^T(j) \underline{\text{kurt}}(j)\}}{\partial \underline{w}(j)} \\ &= \frac{1}{L} \left[ \frac{\partial \{\underline{\text{kurt}}^T(j) \underline{\text{kurt}}(j)\}}{\partial w_1(j)} \quad \dots \quad \frac{\partial \{\underline{\text{kurt}}^T(j) \underline{\text{kurt}}(j)\}}{\partial w_M(j)} \right]^T, \end{aligned} \quad (12)$$

$$[\text{MU}]_j = \begin{bmatrix} \mu_{B1}(j) & \dots & 0 \\ \dots & \dots & \dots \\ 0 & \dots & \mu_{BM}(j) \end{bmatrix}. \quad (13)$$

Note that in (11), a “+” sign is used instead of “-” as in the steepest descent algorithm. Because our performance function is the absolute value of kurtosis rather than error signal, we wish to maximize the function to achieve maximal non-Gaussianity.

To evaluate (12), we have

$$\begin{aligned} & \frac{\partial \{\text{kurt}^T(j)\text{kurt}(j)\}}{\partial w_i(j)} \\ &= \sum_{l=1}^L \frac{\partial [E\{[w^T(j)x_l(j)]^4\} - 3]^2}{\partial w_i(j)} \\ &= 8 \sum_{l=1}^L [w^T(j)X_l(j)]^3 \text{kurt}_l(j)x_{l,i}(j). \end{aligned} \quad (14)$$

In the derivation of (14), the expectation operator was dropped.

The block gradient vector can be written as

$$\begin{aligned} \nabla_B(j) &= \frac{8}{L} \left[ \sum_{l=1}^L [w^T(j)X_l(j)]^3 \text{kurt}_l(j)x_{l,1}(j) \cdots \right. \\ & \quad \left. \sum_{l=1}^L [w^T(j)X_l(j)]^3 \text{kurt}_l(j)x_{l,M}(j) \right]^T \\ &= \frac{8}{L} [G]_j^T [C]_j^3 \text{kurt}(j), \end{aligned} \quad (15)$$

where

$$[C]_j = \begin{bmatrix} \frac{w^T(j)X_1(j)}{L} & \cdots & 0 \\ \cdots & \cdots & \cdots \\ 0 & \cdots & \frac{w^T(j)X_L(j)}{L} \end{bmatrix} \quad (16)$$

is a diagonal matrix.

From (15), the updating formula (11) becomes

$$\underline{w}(j+1) = \underline{w}(j) + \frac{8}{L} [\text{MU}]_j [G]_j^T [C]_j^3 \text{kurt}(j). \quad (17)$$

Now, the primary task is to identify the matrix  $[\text{MU}]_j$  in an optimal sense, so that the total squared kurtosis  $\text{kurt}^T(j)\text{kurt}(j)$  is maximized. In order to do that, we express the  $l$ th kurtosis value in the  $(j+1)$ th iteration by Taylor's series expansion:

$$\begin{aligned} \text{kurt}_l(j+1) &= \text{kurt}_l(j) \\ &+ \sum_{i=1}^M \frac{\partial \text{kurt}_l(j)}{\partial w_i(j)} \Delta w_i(j) \\ &+ \frac{1}{2!} \sum_{m=1}^M \sum_{n=1}^M \frac{\partial^2 \text{kurt}_l(j)}{\partial w_m(j) \partial w_n(j)} \Delta w_m(j) \Delta w_n(j) \\ &+ \cdots, \quad l = 1, 2, \dots, L, \end{aligned} \quad (18)$$

where

$$\Delta w_i(j) = w_i(j+1) - w_i(j), \quad i = 1, 2, \dots, M. \quad (19)$$

In (18), the complexity of the terms increases as the order of the derivative increases. However, if  $\Delta w_i(j)$  is small enough, higher-order derivative terms can be omitted. In our experimentation, it is found that this is indeed the case.

The expectation operator in (9) is dropped. Thus,

$$\frac{\partial \text{kurt}_l(j)}{\partial w_i(j)} = 4x_{l,i}(j) [w^T(j)X_l(j)]^3. \quad (20)$$

Then, (18) becomes

$$\begin{aligned} \text{kurt}_l(j+1) &= \text{kurt}_l(j) + 4 [w^T(j)X_l(j)]^3 \sum_{i=1}^M x_{l,i}(j) \Delta w_i(j) \\ &= \text{kurt}_l(j) + 4 [w^T(j)X_l(j)]^3 [X_l^T(j) \Delta w(j)]. \end{aligned} \quad (21)$$

Writing (21) for every  $l$ , the matrix-vector form of the Taylor expansion becomes

$$\underline{\text{kurt}}(j+1) = \underline{\text{kurt}}(j) + 4 [C]_j^3 [G]_j \underline{\Delta w}(j). \quad (22)$$

From (17),

$$\underline{\Delta w}(j) = \frac{8}{L} [\text{MU}]_j [G]_j^T [C]_j^3 \underline{\text{kurt}}(j). \quad (23)$$

Substituting (23) into (22), one obtains

$$\underline{\text{kurt}}(j+1) = \underline{\text{kurt}}(j) + \frac{32}{L} [C]_j^3 [G]_j [\text{MU}]_j [G]_j^T [C]_j^3 \underline{\text{kurt}}(j). \quad (24)$$

Defining  $\underline{q}(j)$  and  $[R]_j$  as

$$\underline{q}(j) = [G]_j^T [C]_j^3 \underline{\text{kurt}}(j) = [q_1(j), \dots, q_M(j)]^T, \quad (25)$$

$$[R]_j = [G]_j^T [C]_j^6 [G]_j = [R_{mn}(j)], \quad 1 \leq m, n \leq M. \quad (26)$$

The total squared kurtosis for the  $(j+1)$ th iteration can be written as

$$\underline{\text{kurt}}^T(j+1)\underline{\text{kurt}}(j+1) = S_1 + S_2 + S_3, \quad (27a)$$

where

$$S_1 = \underline{\text{kurt}}^T(j)\underline{\text{kurt}}(j), \quad (27b)$$

$$S_2 = \frac{64}{L} \sum_{i=1}^M q_i^2(j) \mu_{Bi}(j), \quad (27c)$$

$$S_3 = \frac{1024}{L^2} \underline{q}^T(j) [\text{MU}]_j [R]_j [\text{MU}]_j \underline{q}(j). \quad (27d)$$

In order to identify  $[\text{MU}]_j$  optimally, the following condition

must be met:

$$\frac{\partial\{\text{kurt}^T(j+1)\text{kurt}(j+1)\}}{\partial\mu_{Bi}(j)} = 0, \quad i = 1, 2, \dots, M. \quad (28)$$

Combining (27a) and (28) yields

$$\frac{\partial S_1}{\partial\mu_{Bi}(j)} + \frac{\partial S_2}{\partial\mu_{Bi}(j)} + \frac{\partial S_3}{\partial\mu_{Bi}(j)} = 0. \quad (29)$$

Substituting (27b), (27c), and (27d) into (29), and using the symmetry property of the matrix  $[R]_j$  given in (26), the following is obtained:

$$\sum_{k=1}^M [q_k(j)\mu_{Bk}^*(j)r_{ki}(j)] = -\frac{L}{32}q_i(j), \quad (30)$$

where  $*$  denotes the optimal value.

Writing (30) for every  $i$ , the following matrix-vector equation is obtained:

$$[R]_j[\text{MU}]_j^* \underline{q}(j) = -\frac{L}{32}\underline{q}(j). \quad (31)$$

From (31), we have

$$[\text{MU}]_j^* \underline{q}(j) = -\frac{L}{32}[R]_j^{-1}\underline{q}(j). \quad (32)$$

From (25), (32), and (17), the OBA/ICA algorithm is obtained:

$$\begin{aligned} \underline{w}(j+1) &= \underline{w}(j) + \frac{8}{L}\left(-\frac{L}{32}\right)[R]_j^{-1}\underline{q}(j) \\ &= \underline{w}(j) - 0.25[R]_j^{-1}\underline{q}(j), \end{aligned} \quad (33)$$

where  $[R]_j$  and  $\underline{q}(j)$  are given by (25) and (26).

Now we show that online gradient-based ICA can be obtained as a special case of the more general OBA/ICA formulation presented above. Let  $L = 1$  and let  $\mu_{B1}(j) = \mu_{B2}(j) = \dots = \mu_{BM}(j) = \mu_B(j)$ , then OBA/ICA simplifies to

$$\underline{w}(j+1) = \underline{w}(j) - 0.25\mu_B^*(j)\underline{X}(j)[\underline{w}^T(j)\underline{X}(j)]^3 \text{kurt}(j), \quad (34)$$

where

$$\mu_B^*(j) = \frac{1}{[\underline{w}^T(j)\underline{X}(j)]^6 [\underline{X}^T(j)\underline{X}(j)]}. \quad (35)$$

If we let  $\mu = 0.25\mu_B^*(j)|\text{kurt}(j)|$ , the online gradient-based ICA is obtained [17, page 177]:

$$\underline{w}(j+1) = \underline{w}(j) - \mu \left( \text{sign}[\text{kurt}(j)]\underline{X}(j)[\underline{w}^T(j)\underline{X}(j)]^3 \right). \quad (36)$$

## 5. IMPLEMENTATION ISSUES

### 5.1. Elimination of the matrix inversion operation

OBA/ICA algorithm, (33), gives the optimal updating formula to extract one row of the separation matrix  $\mathbf{W}$ . The

update equation, (33), involves the inversion of the  $[R]$  matrix, whose dimensionality is equal to the order of the system  $M$ . This operation could be inefficient in the case of a high-order system. This is because the computational complexity of the matrix inversion operation is  $O(M^3)$ . When  $M$  is large, an estimate of  $[R]$  can be used. The method proposed here is to use a diagonal matrix  $[R]_D$  which contains only the diagonal elements of  $[R]$ . Thus, the complexity of the inverse operation becomes  $O(M)$ . From extensive simulations, it is found that the adaptive system repairs itself from this approximation and converges to the right solution in a few additional iterations.

### 5.2. Computational complexity

Having eliminated the inversion problem, the dominant factor determining the computational complexity is the block size  $L$  for most applications of ICA.  $L$  is typically larger than the order of the system  $M$ . It is easily seen that the number of multiplications and divisions of OBA/ICA is  $O(L)$  per iteration, which is equivalent to fast-ICA.

### 5.3. An optional scaling constant

In practice, a parameter  $k$  can be introduced in (33) to further optimize the algorithm performance if a priori information is available regarding the speed of time variation of the channel. Also, since the high-order derivative terms in (18) are dropped in our formulation, an additional adaptation parameter can help to ensure reliable convergence. However, the value of  $k$  is not critical, and the algorithm successfully converges over a wide range of  $k$ , as is confirmed by our simulations.

Therefore, the optimized updating formula is obtained based on (33) as

$$\underline{w}(j+1) = \underline{w}(j) - 0.25k[R]_j^{-1}\underline{q}(j), \quad (37)$$

where the choice of  $k$  is made according to the convergence property and the speed of mixing matrix's time variation.

### 5.4. Types of time variations

In our simulations two types of time variations are studied, which correspond to two scenarios that can arise in mobile communication applications.

In the first case, the change of the channel is modeled as a continuous linear time variation in the mixing matrix's coefficients. In this case, the ICA algorithm seeks a compromise separation matrix that recovers the source signals with minimum error.

The second type of time variation arises when the user is experiencing handover between two service towers. In this scenario, the mixing matrix's coefficients are modeled by an abrupt change. Note that the ICA processing will only be affected when the abrupt change occurs within one processing block. This is the case studied in our simulation.



When an abrupt change occurs within a processing block, the performance for the block degrades significantly, especially when the block size is large. This is because the converged demixing vector is a compromise between two completely different channel parameters. In order to deal with this situation, we propose to locate the position of the abrupt change within the block. This technique will improve the performance if the performance degradation is due to an abrupt change within the block.

In the search procedure, the demixing matrices obtained through the previous block  $\mathbf{W1}$  and the subsequent block  $\mathbf{W2}$  are utilized.

First, the block is evenly divided into two subblocks.  $\mathbf{W1}$  is used to process the first subblock, while  $\mathbf{W2}$  is used to process the second subblock.

If the separation performance for the second subblock is better, it is concluded that the abrupt change occurs within the first subblock. Otherwise, it is concluded that the abrupt change occurs within the second subblock.

Thus, the location of the abrupt change is narrowed down to a subblock. The search process can be continued by dividing that subblock evenly and using  $\mathbf{W1}$  and  $\mathbf{W2}$  to process the two subblocks, respectively. This procedure can be repeated until the location of the abrupt change is narrowed down to a very small range.

Once the location is identified, the symbols before the abrupt change are processed by  $\mathbf{W1}$ , and the symbols after the abrupt change are processed by  $\mathbf{W2}$ .

## 6. APPLICATION IN MOBILE TELECOMMUNICATIONS

To study the performance of OBA/ICA, computer simulations are performed. The performance measures are the signal-to-interference ratio (SIR) and the number of iterations to convergence  $N_c$ . SIR represents the average ratio of the desired signal power to the power of the estimation error, defined as

$$\text{SIR} = 10 \log_{10} \left( \frac{1}{L} \sum_{k=1}^L \frac{s(k)^2}{[s(k) - y(k)]^2} \right), \quad (38)$$

where  $s(k)$  is the  $k$ th sample of the desired signal,  $y(k)$  is the estimate of the  $s(k)$  obtained at the output of the ICA processing unit.

For continuous linear time variation, the mixing matrix simulated is chosen as

$$\mathbf{A} = \begin{bmatrix} 1 + l\Delta & 0.5 \\ 0.7 & 2 + l\Delta \end{bmatrix}, \quad (39)$$

where  $l = 1, 2, \dots, L$ , and  $\Delta$  is the parameter reflecting the speed of channel variation. Here, it is assumed that the channel's transfer function is frequency-flat over the signal band. Also, the sampling interval of the receiver's A/D converter is negligible compared with  $1/\Delta$ , which represents the rate of the channel's time variation.

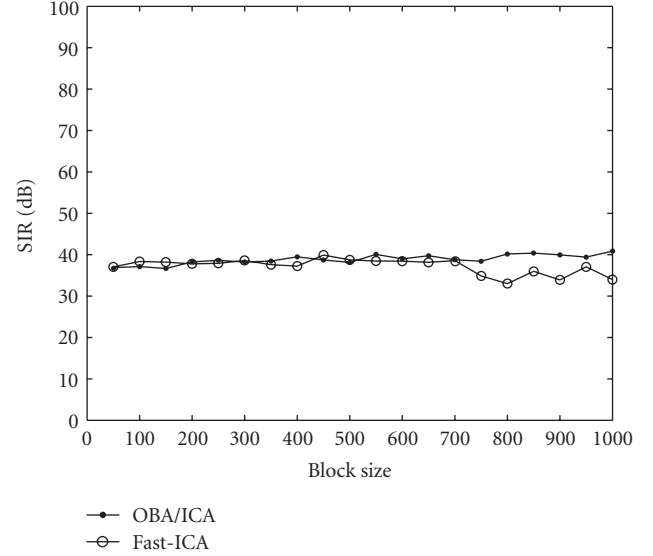


FIGURE 2: Signal-to-interference ratio (SIR) achieved in dB versus the processed block size employing fast-ICA and OBA/ICA ( $k=0.5$ ) when channel conditions vary linearly with time:  $\Delta = 0.01$  in (39).

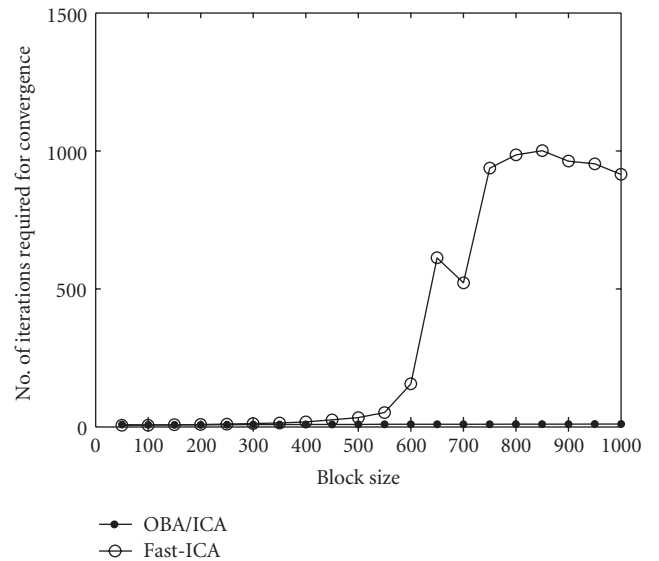


FIGURE 3: Convergence speed of fast-ICA and OBA/ICA ( $k=0.5$ ) versus the processed block size when channel conditions vary linearly with time:  $\Delta = 0.01$  in (39).

In our simulations, the block size is varied from 50 symbols to 1000 symbols, with a step size of 50. For each  $L$ , SIR and  $N_c$  are computed and averaged over 100 simulation runs.

Figures 2 and 3 show the performance and convergence speed of OBA/ICA and fast-ICA for relatively slow time-varying channel condition, that is,  $\Delta = 0.01$ . The additional scaling factor  $k$  in OBA/ICA (37) is 0.5. It is seen that the two algorithms have similar performance except for longer blocks, in which case OBA/ICA has better performance. This indicates OBA/ICA has better capability in dealing with time

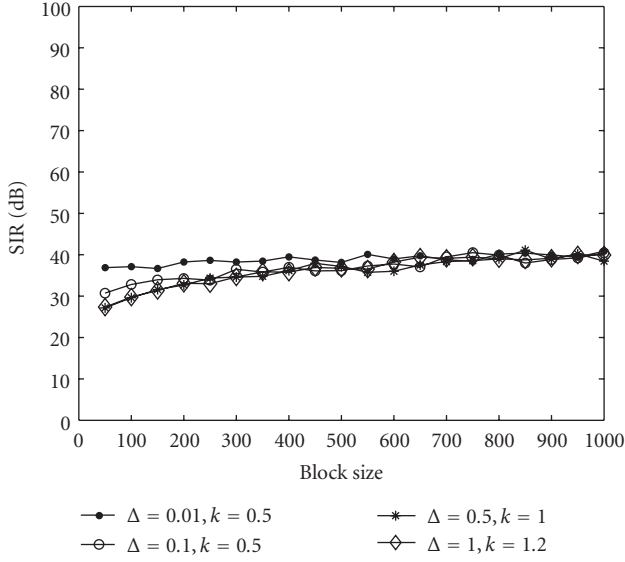


FIGURE 4: SIR achieved in dB versus the processed block size employing OBA/ICA when channel conditions vary linearly with time.

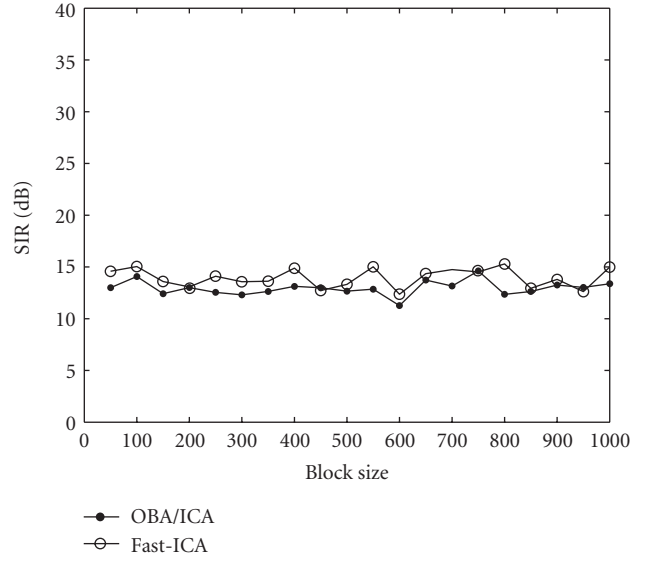


FIGURE 6: SIR achieved by OBA/ICA ( $k = 0.5$ ) and fast-ICA when channel conditions change abruptly.

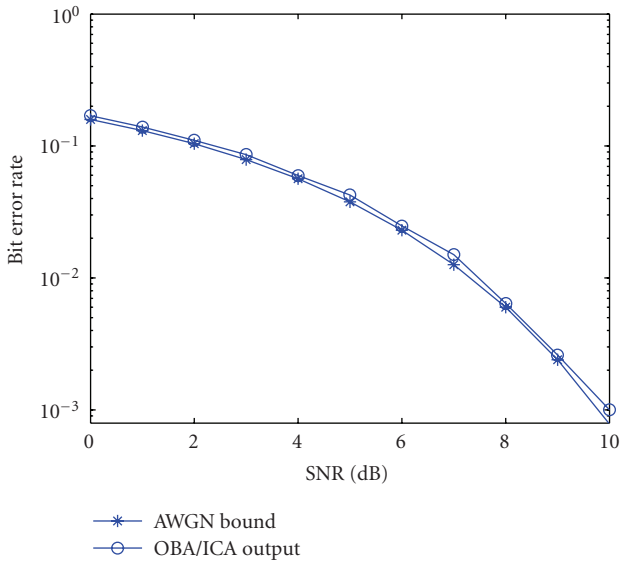


FIGURE 5: Bit error rate (BER) versus SNR employing OBA/ICA.

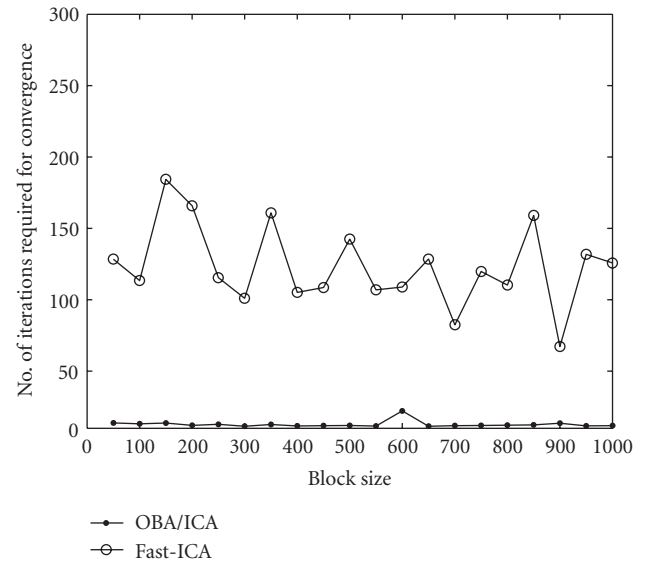


FIGURE 7: Convergence of OBA/ICA ( $k = 0.5$ ) and fast-ICA when channel conditions change abruptly.

variation within one processing block. Also, fast-ICA converges very slowly for long blocks, while OBA/ICA always converges within 20 iterations regardless of the block size.

For faster time variation, that is,  $\Delta = 0.1, 0.5, 1$ , fast-ICA fails to converge within one thousand iterations, which makes it impractical to use. On the other hand, OBA/ICA always converges within 20 iterations. This is why only the OBA/ICA results are given. The performance for OBA/ICA is given in Figure 4. The optimal  $k$  values are given for every  $\Delta$ . It is observed that a larger  $k$  should be used for faster time variation, as expected.

To study the performance of OBA/ICA under noisy conditions, simulations are performed with  $\Delta = 0.01$  and thermal noise added. The resulting bit error rate (BER) is plotted versus signal-to-noise ratio (SNR) in Figure 5. As a reference, the BER with additive noise only, known as the AWGN (additive white Gaussian noise) bound, is also shown for comparison. It is clearly seen that OBA/ICA successfully achieves interference suppression in noisy conditions, and the obtained BER is close to the AWGN bound, which corresponds to the interference-free scenario. The convergence of OBA/ICA under noisy conditions requires about 7 to 16



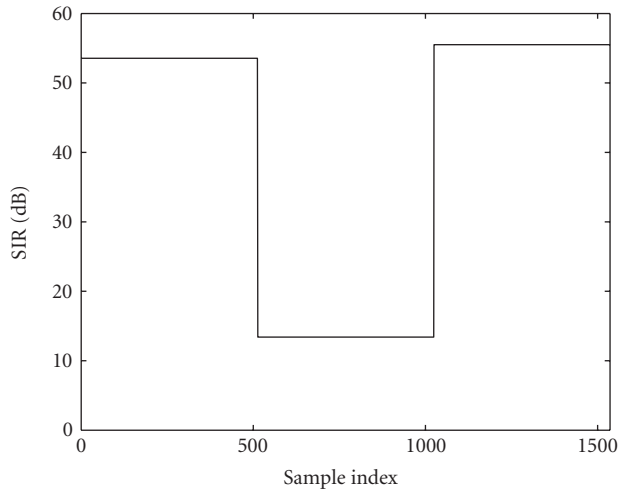


FIGURE 8: SIR achieved by OBA/ICA for three blocks when channel conditions change abruptly in time without finding the location of the sudden change (block size = 512).

iterations, compared to 7 to 10 iterations in the noiseless case. Therefore, a slight increase in the processing time may be required for OBA/ICA in the presence of thermal noise.

Next, fast-ICA and OBA/ICA are compared under abruptly changing channel conditions. To simulate this condition, an abrupt change of the mixing matrix is introduced within the processing block. Figures 6 and 7 compare fast-ICA and OBA/ICA in terms of average SIR and convergence speed without any knowledge about the abrupt change. As expected, the performance of both algorithms degrades when compared to the case of continuous time variation. However, OBA/ICA converges much faster than fast-ICA.

Following the detection of an abrupt change within a certain block, the binary search technique described in Section 5.4 is simulated to detect the location of the abrupt change. As before, one hundred simulation runs are performed and the average performance is given. The block size is chosen to be 512 samples. Figure 8 shows the performance of OBA/ICA for three consecutive blocks when a sudden channel change is simulated at the middle of the second block. Since the adaptive algorithm tries to converge to a compromising demixing matrix for two completely different mixing matrices, the performance for the second block degraded significantly. Figure 9 describes the performance of OBA/ICA after the application of binary search for the second block. As seen, the technique successfully identified the position of the abrupt change denoted by “a,” and the resulting performance for the second block is substantially improved compared to Figure 8.

In addition to these simulation results, in Figures 10 and 11 the residue interference power and the SIR value are shown as a function of the iteration index. Although the whole block is processed with a converged demixing matrix, the two figures illustrate the convergence process of OBA/ICA algorithm.

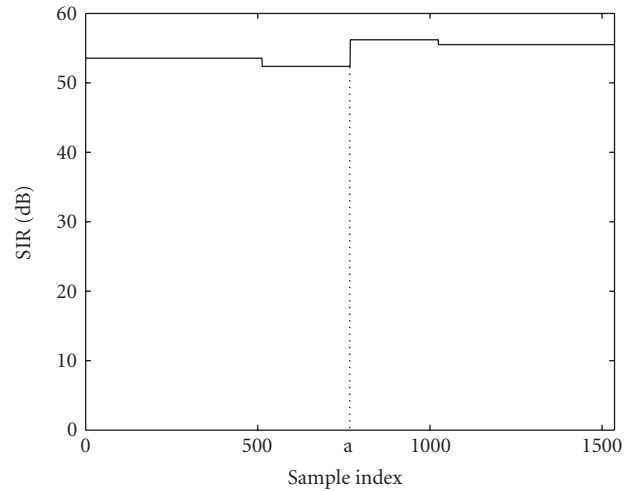


FIGURE 9: SIR achieved by OBA/ICA for three blocks when channel conditions change abruptly in time after finding the location of the sudden change (block size = 512).

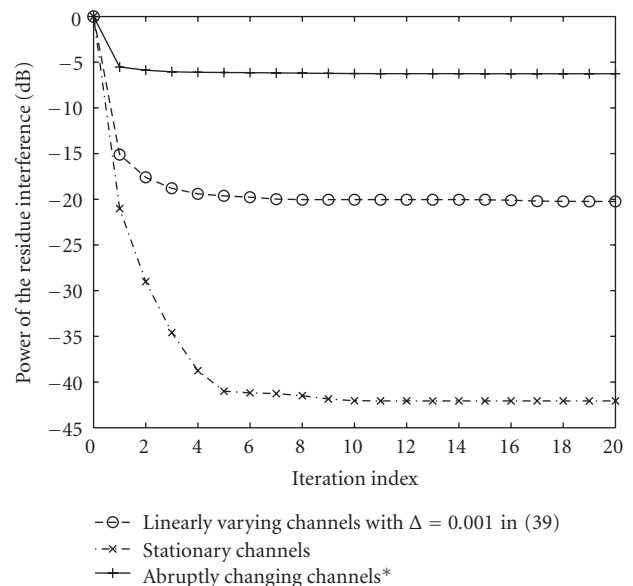


FIGURE 10: Residue interference power averaged over a hundred simulation runs versus iteration number for OBA/ICA assuming block size = 100. \*Without finding the location of the abrupt change within the block.

## 7. CONCLUSIONS

In this paper, a gradient-based ICA algorithm with optimum block adaptation (OBA/ICA) is developed, which tailors the learning rate for each coefficient in the separation matrix and updates those rates at each block iteration. The computational complexity of OBA/ICA for each iteration is equivalent to the fast-ICA. When the channel is time varying, the

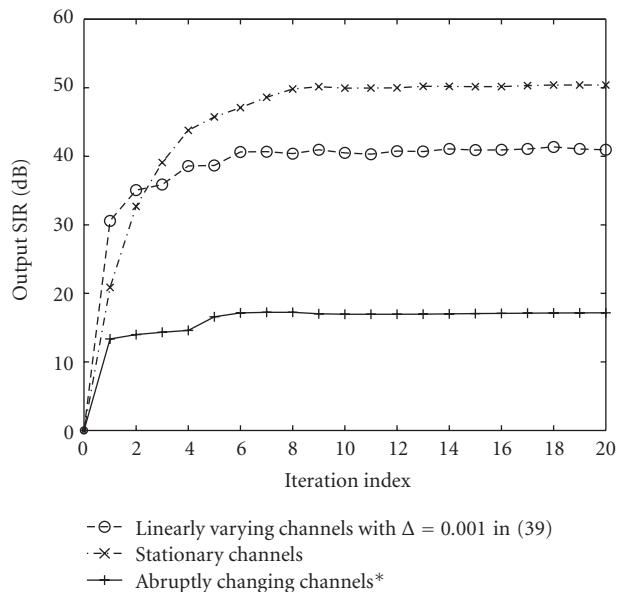


FIGURE 11: Output SIR averaged over a hundred simulation runs versus iteration number for OBA/ICA assuming block size = 100. \*Without finding the location of the abrupt change within the block.

proposed technique is superior to the fast-ICA, especially in terms of convergence properties. This is true for changes that are linear or abrupt in nature.

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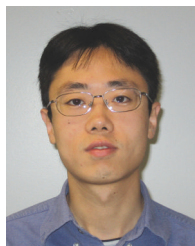
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# Special Issue on Transforming Signal Processing Applications into Parallel Implementations

## Call for Papers

There is an increasing need to develop efficient “system-level” models, methods, and tools to support designers to quickly transform signal processing application specification to heterogeneous hardware and software architectures such as arrays of DSPs, heterogeneous platforms involving microprocessors, DSPs and FPGAs, and other evolving multiprocessor SoC architectures. Typically, the design process involves aspects of application and architecture modeling as well as transformations to translate the application models to architecture models for subsequent performance analysis and design space exploration. Accurate predictions are indispensable because next generation signal processing applications, for example, audio, video, and array signal processing impose high throughput, real-time and energy constraints that can no longer be served by a single DSP.

There are a number of key issues in transforming application models into parallel implementations that are not addressed in current approaches. These are engineering the application specification, transforming application specification, or representation of the architecture specification as well as communication models such as data transfer and synchronization primitives in both models.

The purpose of this call for papers is to address approaches that include application transformations in the performance, analysis, and design space exploration efforts when taking signal processing applications to concurrent and parallel implementations. The Guest Editors are soliciting contributions in joint application and architecture space exploration that outperform the current architecture-only design space exploration methods and tools.

Topics of interest for this special issue include but are not limited to:

- modeling applications in terms of (abstract) control-dataflow graph, dataflow graph, and process network models of computation (MoC)
- transforming application models or algorithmic engineering
- transforming application MoCs to architecture MoCs
- joint application and architecture space exploration

- joint application and architecture performance analysis
- extending the concept of algorithmic engineering to architecture engineering
- design cases and applications mapped on multiprocessor, homogeneous, or heterogeneous SOCs, showing joint optimization of application and architecture

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# Special Issue on Video Adaptation for Heterogeneous Environments

## Call for Papers

The explosive growth of compressed video streams and repositories accessible worldwide, the recent addition of new video-related standards such as H.264/AVC, MPEG-7, and MPEG-21, and the ever-increasing prevalence of heterogeneous, video-enabled terminals such as computer, TV, mobile phones, and personal digital assistants have escalated the need for efficient and effective techniques for adapting compressed videos to better suit the different capabilities, constraints, and requirements of various transmission networks, applications, and end users. For instance, Universal Multimedia Access (UMA) advocates the provision and adaptation of the same multimedia content for different networks, terminals, and user preferences.

Video adaptation is an emerging field that offers a rich body of knowledge and techniques for handling the huge variation of resource constraints (e.g., bandwidth, display capability, processing speed, and power consumption) and the large diversity of user tasks in pervasive media applications. Considerable amounts of research and development activities in industry and academia have been devoted to answering the many challenges in making better use of video content across systems and applications of various kinds.

Video adaptation may apply to individual or multiple video streams and may call for different means depending on the objectives and requirements of adaptation. Transcoding, transmoding (cross-modality transcoding), scalable content representation, content abstraction and summarization are popular means for video adaptation. In addition, video content analysis and understanding, including low-level feature analysis and high-level semantics understanding, play an important role in video adaptation as essential video content can be better preserved.

The aim of this special issue is to present state-of-the-art developments in this flourishing and important research field. Contributions in theoretical study, architecture design, performance analysis, complexity reduction, and real-world applications are all welcome.

Topics of interest include (but are not limited to):

- Heterogeneous video transcoding
- Scalable video coding
- Dynamic bitstream switching for video adaptation

- Signal, structural, and semantic-level video adaptation
- Content analysis and understanding for video adaptation
- Video summarization and abstraction
- Copyright protection for video adaptation
- Crossmedia techniques for video adaptation
- Testing, field trials, and applications of video adaptation services
- International standard activities for video adaptation

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# Special Issue on Knowledge-Assisted Media Analysis for Interactive Multimedia Applications

## Call for Papers

It is broadly acknowledged that the development of enabling technologies for new forms of interactive multimedia services requires a targeted confluence of knowledge, semantics, and low-level media processing. The convergence of these areas is key to many applications including interactive TV, networked medical imaging, vision-based surveillance and multimedia visualization, navigation, search, and retrieval. The latter is a crucial application since the exponential growth of audiovisual data, along with the critical lack of tools to record the data in a well-structured form, is rendering useless vast portions of available content. To overcome this problem, there is need for technology that is able to produce accurate levels of abstraction in order to annotate and retrieve content using queries that are natural to humans. Such technology will help narrow the gap between low-level features or content descriptors that can be computed automatically, and the richness and subjectivity of semantics in user queries and high-level human interpretations of audiovisual media.

This special issue focuses on truly integrative research targeting of what can be disparate disciplines including image processing, knowledge engineering, information retrieval, semantic, analysis, and artificial intelligence. High-quality and novel contributions addressing theoretical and practical aspects are solicited. Specifically, the following topics are of interest:

- Semantics-based multimedia analysis
- Context-based multimedia mining
- Intelligent exploitation of user relevance feedback
- Knowledge acquisition from multimedia contents
- Semantics based interaction with multimedia
- Integration of multimedia processing and Semantic Web technologies to enable automatic content sharing, processing, and interpretation
- Content, user, and network aware media engineering
- Multimodal techniques, high-dimensionality reduction, and low level feature fusion

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# Special Issue on Super-resolution Enhancement of Digital Video

## Call for Papers

When designing a system for image acquisition, there is generally a desire for high spatial resolution and a wide field-of-view. To achieve this, a camera system must typically employ small f-number optics. This produces an image with very high spatial-frequency bandwidth at the focal plane. To avoid aliasing caused by undersampling, the corresponding focal plane array (FPA) must be sufficiently dense. However, cost and fabrication complexities may make this impractical. More fundamentally, smaller detectors capture fewer photons, which can lead to potentially severe noise levels in the acquired imagery. Considering these factors, one may choose to accept a certain level of undersampling or to sacrifice some optical resolution and/or field-of-view.

In image super-resolution (SR), postprocessing is used to obtain images with resolutions that go beyond the conventional limits of the uncompensated imaging system. In some systems, the primary limiting factor is the optical resolution of the image in the focal plane as defined by the cut-off frequency of the optics. We use the term “optical SR” to refer to SR methods that aim to create an image with valid spatial-frequency content that goes beyond the cut-off frequency of the optics. Such techniques typically must rely on extensive a priori information. In other image acquisition systems, the limiting factor may be the density of the FPA, subsequent postprocessing requirements, or transmission bitrate constraints that require data compression. We refer to the process of overcoming the limitations of the FPA in order to obtain the full resolution afforded by the selected optics as “detector SR.” Note that some methods may seek to perform both optical and detector SR.

Detector SR algorithms generally process a set of low-resolution aliased frames from a video sequence to produce a high-resolution frame. When subpixel relative motion is present between the objects in the scene and the detector array, a unique set of scene samples are acquired for each frame. This provides the mechanism for effectively increasing the spatial sampling rate of the imaging system without reducing the physical size of the detectors.

With increasing interest in surveillance and the proliferation of digital imaging and video, SR has become a rapidly growing field. Recent advances in SR include innovative algorithms, generalized methods, real-time implementations,

and novel applications. The purpose of this special issue is to present leading research and development in the area of super-resolution for digital video. Topics of interest for this special issue include but are not limited to:

- Detector and optical SR algorithms for video
- Real-time or near-real-time SR implementations
- Innovative color SR processing
- Novel SR applications such as improved object detection, recognition, and tracking
- Super-resolution from compressed video
- Subpixel image registration and optical flow

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## Special Issue on

# Advanced Signal Processing and Computational Intelligence Techniques for Power Line Communications

### Call for Papers

In recent years, increased demand for fast Internet access and new multimedia services, the development of new and feasible signal processing techniques associated with faster and low-cost digital signal processors, as well as the deregulation of the telecommunications market have placed major emphasis on the value of investigating hostile media, such as powerline (PL) channels for high-rate data transmissions.

Nowadays, some companies are offering powerline communications (PLC) modems with mean and peak bit-rates around 100 Mbps and 200 Mbps, respectively. However, advanced broadband powerline communications (BPLC) modems will surpass this performance. For accomplishing it, some special schemes or solutions for coping with the following issues should be addressed: (i) considerable differences between powerline network topologies; (ii) hostile properties of PL channels, such as attenuation proportional to high frequencies and long distances, high-power impulse noise occurrences, time-varying behavior, and strong inter-symbol interference (ISI) effects; (iv) electromagnetic compatibility with other well-established communication systems working in the same spectrum, (v) climatic conditions in different parts of the world; (vii) reliability and QoS guarantee for video and voice transmissions; and (vi) different demands and needs from developed, developing, and poor countries.

These issues can lead to exciting research frontiers with very promising results if signal processing, digital communication, and computational intelligence techniques are effectively and efficiently combined.

The goal of this special issue is to introduce signal processing, digital communication, and computational intelligence tools either individually or in combined form for advancing reliable and powerful future generations of powerline communication solutions that can be suited with for applications in developed, developing, and poor countries.

Topics of interest include (but are not limited to)

- Multicarrier, spread spectrum, and single carrier techniques
- Channel modeling

- Channel coding and equalization techniques
- Multiuser detection and multiple access techniques
- Synchronization techniques
- Impulse noise cancellation techniques
- FPGA, ASIC, and DSP implementation issues of PLC modems
- Error resilience, error concealment, and Joint source-channel design methods for video transmission through PL channels

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# Special Issue on Numerical Linear Algebra in Signal Processing Applications

## Call for Papers

The cross-fertilization between numerical linear algebra and digital signal processing has been very fruitful in the last decades. The interaction between them has been growing, leading to many new algorithms.

Numerical linear algebra tools, such as eigenvalue and singular value decomposition and their higher-extension, least squares, total least squares, recursive least squares, regularization, orthogonality, and projections, are the kernels of powerful and numerically robust algorithms.

The goal of this special issue is to present new efficient and reliable numerical linear algebra tools for signal processing applications. Areas and topics of interest for this special issue include (but are not limited to):

- Singular value and eigenvalue decompositions, including applications.
- Fourier, Toeplitz, Cauchy, Vandermonde and semi-separable matrices, including special algorithms and architectures.
- Recursive least squares in digital signal processing.
- Updating and downdating techniques in linear algebra and signal processing.
- Stability and sensitivity analysis of special recursive least-squares problems.
- Numerical linear algebra in:
  - Biomedical signal processing applications.
  - Adaptive filters.
  - Remote sensing.
  - Acoustic echo cancellation.
  - Blind signal separation and multiuser detection.
  - Multidimensional harmonic retrieval and direction-of-arrival estimation.
  - Applications in wireless communications.
  - Applications in pattern analysis and statistical modeling.
  - Sensor array processing.

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## Special Issue on

# Wavelets in Source Coding, Communications, and Networks

### Call for Papers

Wavelet transforms are arguably the most powerful, and most widely-used, tool to arise in the field of signal processing in the last several decades. Their inherent capacity for multiresolution representation akin to the operation of the human visual system motivated a quick adoption and widespread use of wavelets in image-processing applications. Indeed, wavelet-based algorithms have dominated image compression for over a decade, and wavelet-based source coding is now emerging in other domains. For example, recent wavelet-based video coders exploit techniques such as motion-compensated temporal filtering to yield effective video compression with full temporal, spatial, and fidelity scalability. Additionally, wavelets are increasingly used in the source coding of remote-sensing, satellite, and other geospatial imagery. Furthermore, wavelets are starting to be deployed beyond the source-coding realm with increasing interest in robust communication of images and video over both wired and wireless networks. In particular, wavelets have been recently proposed for joint source-channel coding and multiple-description coding. This special issue will explore these and other latest advances in the theory and application of wavelets.

Specifically, this special issue will gather high-quality, original contributions on all aspects of the application of wavelets and wavelet theory to source coding, communications, and network transmission of images and video. Topics of interest include (but are not limited to) the theory and applications of wavelets in:

- Scalable image and video coding
- Motion-compensated temporal filtering
- Source coding of images and video via frames and overcomplete representations
- Geometric and adaptive multiresolution image and video representations
- Multiple-description coding of images and video
- Joint source-channel coding of images and video
- Distributed source coding of images and video
- Robust coding of images and video for wired and wireless packet networks

- Network adaption and transcoding of images and video
- Coding and communication of images and video in sensor networks

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