Independent Component Analysis Based MIMO Transceiver With Improved Performance In Time Varying Wireless Channels

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Abstract

Independent component analysis (ICA) is a signal processing technique used for un-mixing of the mixed recorded signals. In wireless communication, ICA is mainly used in multiple input multiple output (MIMO) systems. Most of the existing work regarding the ICA applications in MIMO systems assumed static or quasi static wireless channels. Performance of the ICA algorithms degrades in case of time varying wireless channels and is further degraded if the data block lengths are reduced to get the quasi stationarity. In this paper, we propose an ICA based MIMO transceiver that performs well in time varying wireless channels, even for smaller data blocks. Simulation is performed over quadrature amplitude modulated (QAM) signals. Results show that the proposed transceiver system outperforms the existing MIMO system utilizing the FastICA and the OBAICA algorithms in both the transceiver systems for time varying wireless channels. Performance improvement is observed for different data blocks lengths and signal to noise ratios (SNRs).

Keywords: Wireless Systems, Time Varying Channel, Independent Component Analysis, MIMO Transceiver

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1. Introduction

Independent component analysis (ICA) is a technique of blind source separation (BSS) and extensively has been used in several systems due to its simplicity and applicability in practical scenarios [1-3] from the last two decades. Various applications of ICA have been reported in the literature e.g., vibration analysis [4, 5], robotics [6], fault diagnosis [7], biomedical signal processing [8], speech processing [9], and wireless communication systems [10, 11]. In wireless communication, ICA is mainly used in multiple input multiple output (MIMO) systems. The MIMO system consists of multiple transmit and receive antennas that make it an ideal candidate for the ICA algorithms.

In general, the ICA algorithms assumed static or quasi static wireless channels [1]. Several ICA based communication systems are presented in the literature. We summarized a few important research articles as follows. A semi-blind receiver structure is presented in [12], that overcomes the phase and permutation indeterminacies in the ICA post processed data by using a minimum number of pilot bits attached with each transmit data block in a quasi-static scenario. Although, the authors in [12] have added a small number of bits with low computational cost but achieved good bit-error-rate (BER) performance. Another MIMO based communication system is presented in [13], that have improved bandwidth efficiency without considering the spatial coding. An ICA based low complexity orthogonal frequency division multiplexing (OFDM) based receiver structure for MIMO system is proposed in [14] with enhanced performance for large sub-carriers.

1.1 Motivation

Fast and accurate ICA algorithms are presented in the literature [15] for static and quasi-static wireless channels. Performance of these algorithms degrades if variations in the wireless channels are such that the quasi stationarity condition does not hold. However, the practical wireless channels are time varying and un-mixing of the mixed received signals becomes a challenging work for ICA to blindly un-mix them in a time varying scenario [16]. If we reduce the lengths of the processing data blocks to get the quasi stationarity, then the performance of the batch ICA algorithms further degrades. Performance degradation occurs due to the statistical parameters used in the update rules of the ICA algorithms. To the best of our knowledge, we are the first to dicuss the applications of the batch ICA algorithms in time varying wireless channels.

1.2 Contribution

In this paper, we propose an ICA based MIMO transceiver for time varying wireless channels. The proposed transceiver system also performs well if variations in the channels are such that the quasi stationarity condition becomes invalid even for smaller data blocks. To the best of our knowledge, no such a transceiver is presented in the literature that efficiently utilized the existing ICA algorithms in a time varying scenario. The transmitter side of this transceiver system transmits multiple copies of each sample of the source data. The receiver receives mixtures of the transmitted source signals. The multiple mixed copies are averaged and then up-sampled. The up-sampled data is un-mixed through the ICA algorithms and then down sampled to get back the original data blocks lengths.

Due to the use of samples copier (SC), samples averaging (SA) and up/down sampling operations used in the proposed transceiver system, we name it the MIMO SCSA-UD transceiver system. The MIMO SCSA-UD transceiver system improves the quality of the received signal in highly dynamic communication environment at the cost of reduced bandwidth efficiency due to the transmission of multiple copies of each source data sample. By highly dynamic or highly time varying we mean that the channels characteristics are changing for each transmitted sample. Moreover, we also propose the automatic values adjustment (AVA) technique in this paper. This technique controls the functions of the SC and the UD sampling operations blocks in such a way if there are no variations in the channel or the channel is quasi static then the SC block will copy each sample once. By this way the bandwidth efficiency will not be affected in quasi static channels condition. In short we can say that the bandwidth efficiency of the proposed transceiver becomes equal to the bandwidth efficiency of the existing MIMO system in quasi static channels conditions.

Performance of the proposed transceiver is evaluated over quadrature amplitude modulated (QAM) signals. The performance evaluation criterion is signal to interference ratio (SIR).

1.3 Notations and Organization

The notations used in this paper are such that the capital size bold letters are used for matrices and the small size bold letters represent vectors. Symbols and abbreviations used in this paper are given in **Table 1** and **2** respectively.

Remaining paper is organized as follows. The signal model and overview of ICA is given in Section 2, Section 3 presents the proposed MIMO SCSA-UD transceiver system. Section 4 illustrates the simulation results. Finally, conclusion is given in Section 5.

Table 1. List of Symbols

$\boldsymbol{H}$	Channel matrix of size $M\times M$
L	Length of the processing data block
M	Maximum number of the transmitted source signals
$\boldsymbol{S}$	Matrix contains source signals of size $M\times L$
$\boldsymbol{s}$	Source signal vector of length L
$\boldsymbol{W}$	Un-mixing matrix of size $M\times M$
$\boldsymbol{X}$	Received mixed data matrix of size $M\times QL$
$\boldsymbol{x}$	Vector represents received mixed data of length L
$\boldsymbol{Y}$	Un-mixed data matrix of size $M\times L$
$\boldsymbol{y}$	Un-mixed data vector of length L
$\uparrow$	This symbol represents up-sampling
$\downarrow$	This symbol represents down sampling
Q	Number of samples which SC copies
Z	Up/Down sampling parameter
$\delta_1, \delta_2, \delta_3, \delta_4$	Complex random variables represent time varying channel matrix
$\boldsymbol{S}'$	The SC post processed source data matrix of size $M\times QL$
$\boldsymbol{X}'$	The SS and SA post processed mixed data matrix of size $M\times L$
$\boldsymbol{X}^{\uparrow}$	The up-sampled mixed data matrix of size $M\times ZL$
$\boldsymbol{x}^{\uparrow}$	The up-sampled mixed data vector of length ZL
$\boldsymbol{Y}^{\uparrow}$	The un-mixed source data matrix of size $M\times ZL$
$\boldsymbol{y}^{\uparrow}$	The un-mixed source data vector of length ZL
μ	An integer value used in the AVA technique for the values of Z & Q

M	Integer values ranges from 1 to M
δ	Unit impulse function

Table 2. List of Abbreviations

BSS	Blind source separation
ICA	Independent component analysis
QAM	Quadrature amplitude modulation
MIMO	Multiple input multiple output
SIR	Signal to interference ratio
US	Up-sampling
DS	Down sampling
SC	Samples copiers
SA	Samples averaging
AWGN	Additive white gassian noise
SNR	Signal to noise ratio
CDF	Commulative densisty function
PDF	Probability density function
FDICA	Frequency domain ICA
AVA	Automatic values adjustment
SIR_{avg}	Average SIR

2. The Signal Model and Overview of ICA

We consider a multi-user MIMO wireless communication system with M_T transmit and M_R receive antennas. For the sake of simplicity, it is considered that $M_T = M_R = M$. The transmitted signals vectors are $\boldsymbol{s}_1, \boldsymbol{s}_2, \ldots, \boldsymbol{s}_M$, where $\boldsymbol{s}_m = [s_{m1}, s_{m2}, \ldots, s_{mL}]$ and the received mixed signals are $\boldsymbol{x}_1, \boldsymbol{x}_2, \ldots, \boldsymbol{x}_M$. The system model for ICA signals processing is shown in **Fig. 1**. The ICA post process reconstructed signals are represented by $y_1, y_2, \ldots, y_M$. Where the post processed signals are the resultant un-mixed signals of the ICA algorithms. The un-mixed signals are the estimated resultant source signals transmitted by different antennas. The ICA algorithms un-mix the received mixed signals in order to find the actual transmitted source signals. The received mixed signals can be modeled as follows

$$\boldsymbol{X}=\boldsymbol{HS} \tag{1}$$

where, $\boldsymbol{X}$ is $M \times L$ mixed data matrix, $\boldsymbol{H}$ is $M \times M$ complex valued channel matrix and $\boldsymbol{S}$ is $M \times L$ matrix that contains the independent and non-Gaussian source signals. Though the ICA research can be divided into the following categories

(a) Equal number of transmit and receive antennas [14, 16]: In this case we get the mixing coefficients in such a way that the mixing matrix becomes square. For square mixing matrix stable and efficient ICA algorithms exist like FastICA algorithms [25], OBAICA algorithm [16], and Infomax algorithm [26] etc.

(b) The number of transmitted signals greater than the number of received signals: This area of research in the ICA literature is called over-complete ICA [17].

(c) The number of transmitted signals less than the number of received signals: This area of research in the field of ICA is known as the under-complete ICA [18].
(d) More transmit signal and one receive antenna or sensor: This is a different area of research in the field of ICA signal processing, where we receive one mixed signal for further processing through ICA algorithms [19, 20].

We use the square channel or mixing matrix in our analysis. As a future work, we would like to extend it for non-square case utilizing the techniques of non-square mixing scenario.

Equation (1) represents the noise free model of ICA. The noisy ICA model can be represented as

$$\boldsymbol{X}=\boldsymbol{HS}+\boldsymbol{N} \tag{2}$$

where, $\boldsymbol{N}$ is $M\times L$ matrix, which represents the complex valued additive white Gaussian noise with variance one and assumed independent to the source signals. The addition of noise also degrades the performance of the ICA algorithms. The role of the ICA algorithms is to estimate the un-mixing matrix $\boldsymbol{W}$, where $\boldsymbol{W}=\boldsymbol{H}^{-1}$. After estimating the un-mixing matrix the desired un-mixed signals can be represented as follows

$$\boldsymbol{Y}=\boldsymbol{WX} \tag{3}$$

Once the un-mixing matrix $\boldsymbol{W}$ is determined, then, the individual component signals can be separated from the mixed signals by using equation (3). Accuracy of the ICA algorithms depends on the criterion used for estimating the un-mixing matrix.

In case of highly time varying channels, the coefficients of the channel matrix $\boldsymbol{H}$ changes for each incoming data sample and the un-mixing becomes a challenging work for the ICA algorithms.

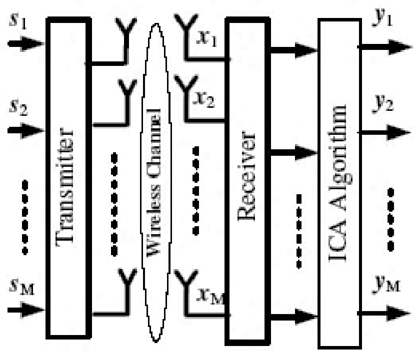

Fig. 1. The existing MIMO transceiver system for ICA signals processing.

Assumptions used in this paper are given below

- The source signals are mutually statistically independent. If different signals are related to different sources then the independence assumption becomes valid.
- The source signals have non-Gaussian distributions. The non-Gaussian structure of the signals is also practical, because most of the signals from different physical processes have non-Gaussian distribution. Moreover, the transmitted signals in digital modulation schemes like quadrature amplitude modulation (QAM), binary phase shift keying (BPSK), and quadrature phase shift keying (QPSK) are non-

Gaussian in nature as discussed in [21-23], where we select finite alphabets from a finite alphabet set.

- The channel matrix ***H*** is assumed to be square. If channel matrix is square then estimation of the un-mixing matrix ***W*** becomes easy. This condition means that the number of the underlying component signals and the mixture signals is equal.

There are some indeterminacies in the ICA post processed data, as given below

- Order of the ICA post processed signals changes for each incoming data block.
- Phase and amplitude of the ICA post processed signals also change for each processing data block.

3. The Proposed MIMO SCSA-UD transceiver system

In this section, the proposed transceiver system called the MIMO SCSA-UD transceiver is presented. The MIMO SCSA-UD transceiver improves the separation performance of the batch ICA algorithms in highly time varying channels. In case of highly time varying channels the quasi stationary condition becomes invalid even for smaller data blocks. The proposed transceiver system is given in **Fig. 2**.

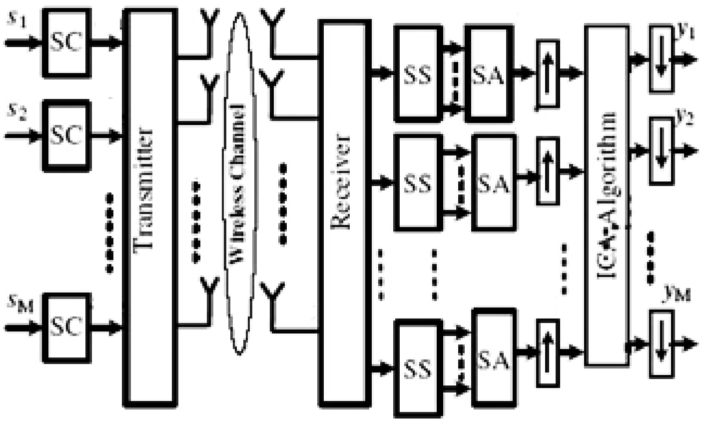

Fig. 2. The proposed MIMO SCSA-UD transceiver system.

The proposed transceiver system first processes the source signals $\boldsymbol{s}_1, \boldsymbol{s}_2, \ldots, \boldsymbol{s}_M$, through samples copier (SC) block that copies each sample Q times before transmission. The source signals are the information signals which we want to transmit. These signals may be speech or any other information signals, which we want to transmit. In order to generate these signals, we first generate random data in matlab using the rand (random) function and then modulate it using the qammod (quardrature amplitude modulation) function of matlab for further processing through the SC block. The transmitted source signals after the SC block are $\boldsymbol{s}'_1, \boldsymbol{s}'_2, \ldots, \boldsymbol{s}'_M$, where $\boldsymbol{s}'_m = [s'_{m1}, s'_{m2}, \ldots, s'_{mLQ}]$. The SC process for each source data vector is demonstrated as follows

$$\boldsymbol{s}'_m = [(s^1_{m1}, s^2_{m1}, \ldots, s^Q_{m1}), (s^1_{m2}, s^2_{m2}, \ldots, s^Q_{m2}), \ldots, (s^1_{mL}, s^2_{mL}, \ldots, s^Q_{mL})] \tag{4}$$

At the receiver side linearly mixed data vectors $\boldsymbol{x}_1, \boldsymbol{x}_2, \ldots, \boldsymbol{x}_M$ are received and passed through the samples separator (SS) and samples averaging (SA) blocks. The SS block separates the mixed received samples inserted by the SC block and the SA block calculates their average values. The signal processing of the SS and the SA blocks is shown in **Fig. 3**, where 1*D* represents shift right and -1*D* represents shift left by one sample. This figure shows that first the mixed data vector $\boldsymbol{x}(n)$ is decimated by integer value *Q* to discard the extra inserted samples, then delayed and down sampled $\boldsymbol{x}(n)$ *Q* times and then shift back all the decimated vectors to remove the delays, finally add them up and divide by *Q* to get the average values. This process is demonstrated mathematically as follows

$$\boldsymbol{x}'_m=[(x^1_{m1}+ x^2_{m1}+\dots x^Q_{m1})/Q,\ (x^1_{m2}+ x^2_{m2}+\dots x^Q_{m2})/Q,\ \dots,\ (x^1_{mQL}+ x^2_{mQL}+\dots x^Q_{mQL})/Q] \tag{5}$$

where $x^1_{m1}, x^2_{m1}, \dots, x^Q_{m1}$ represent the mixed received copies of samples that are transmitted *Q* times, where the transmitted source symbols are $s^1_{m1}, s^2_{m1}, \dots, s^Q_{m1}$.

In the next step, the mixed data vectors $\boldsymbol{x}'_m$ are up-sampled before un-mixing. The up-sampling (US) operation increases the data blocks lengths by using an integer factor *Z*. The US block generates (*Z* - 1) samples between two adjacent samples. The up-sampling is a two steps process, (1) zero insertion, and (2) the samples generation. The zero insertion process for a mixed received data block $\boldsymbol{x}'(n)$ can be represented as follows

$$\boldsymbol{x}^{\uparrow}(n) = \sum_{k=1}^{L} \boldsymbol{x}'(n)\delta(n-kZ) \tag{6}$$

where, δ is the unit impulse function.

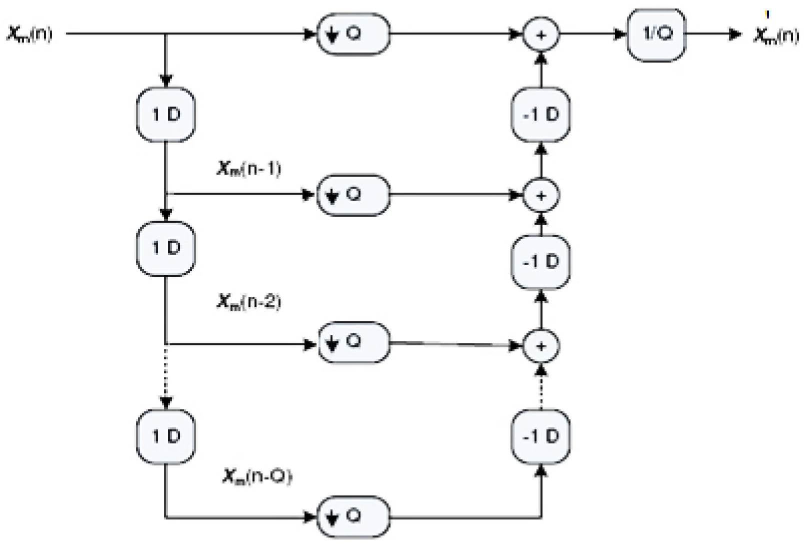

Fig. 3. The signal processing of the samples separator and samples averaging blocks.

After zero insertion the samples generation process is performed using the linear interpolator of [24]. The impulse response of the linear interpolator represented by $\boldsymbol{h}(n)$ and can be written as follows

$$\boldsymbol{h}(n) = \begin{cases} 1, & if\ \frac{|n|}{Z} \leq Q \\ 0, & otherwise \end{cases} \tag{7}$$

The up-sampled mixed data vectors are $\boldsymbol{x}^{\uparrow}_m = [x^{\uparrow}_{m1}, x^{\uparrow}_{m2}, \dots, x^{\uparrow}_{mZL}]$, where m=1,2,…,*M*. The up-sampled mixed data is un-mixed by using the complex valued FastICA algorithm [25]. This algorithm is used as a bench mark to evaluate the effectiveness of the proposed transceiver system. The resultant un-mixed data vectors are $\boldsymbol{y}^{\uparrow}_1, \boldsymbol{y}^{\uparrow}_2, \dots, \boldsymbol{y}^{\uparrow}_M$. The un-mixed data is then down sampled using the decimation parameter Z to get back the original data blocks lengths. The DS or decimator discards (Z -1) number of samples and keeps the remaining samples. The decimated version of the data block $\boldsymbol{y}^{\uparrow}(n)$ can be represented as follows

$$\boldsymbol{y}(n) = \boldsymbol{y}^{\uparrow}(nZ) \tag{8}$$

The decimated data vectors are $\boldsymbol{y}_1, \boldsymbol{y}_2, \ldots, \boldsymbol{y}_M$, where $\boldsymbol{y}_m = [y_{m1}, y_{m2}, \ldots, y_{mL}]$.

The indeterminacies in the resultant reconstructed data can be resolved by transmitting few known symbols with each data block as performed in [12].

Now, the important thing is how to select the optimal values of the key parameters Z and Q. For this purpose, we propose the automatic values adjustment (AVA) technique. This technique is responsible for selection of the desired values of Z and Q according to the channels conditions. The AVA technique is presented in **Fig. 4**. The AVA technique initially selects the SC parameter Q and the UD sampling parameter Z equal 1 and computes the signals to interference ratios (SIRs) of the resultant un-mixed signals $\boldsymbol{y}_1, \boldsymbol{y}_2, \ldots, \boldsymbol{y}_M$. The SIR represents the average ratio of the desired signal power to the power of the estimation error and is defined as follows

$$SIR_{dB}^{i} = 10\log_{10}\left(\frac{1}{L}\sum_{n=1}^{L}\left[\frac{(s^i(n))^2}{(s^i(n)-y^i(n))^2}\right]\right) \tag{9}$$

where i=1,2,...,M.

SIR is used as a criteria to evaluate the results of the proposed and existing MIMO systems using the FastICA and OBAICA algorithms in both the transceivers. Basically, we are transmitting M number of signals utilizing M number of antennas. Equation (9) calculates the SIRs of all the transmitted data blocks. After calculating the SIRs we compute the average SIR (SIR_{avg}) as follows

$$SIR_{avg} = \frac{SIR^1 + SIR^2 + \cdots + SIR^M}{M} \tag{10}$$

Now compare the value of the SIR_{avg} with the predefined threshold μ, where this constant defines the quality of the resultant un-mixed signals. If $SIR_{avg} > \mu$, then keep the same values of Q and Z. In this scenario, the signal processing of the proposed transceiver system becomes equal to the signal processing of the existing MIMO system shown in **Fig. 1**. Moreover, if $SIR_{avg} \leq \mu$, due to variations in the wireless channels, then the AVA technique will increase the values of Q and Z iteratively up to the desired limit. Furthermore, the signal processing steps of the proposed transceiver system with the AVA technique are summarized as follows

- Generate the signals which we want to transmit
- Perform the SC operation by using equation (4)
- Perform the SS & SA operations by using equation (5)
- Perform the up-sampling by using equation (6) and (7)
- Perform un-mixing using the Complex valued FastICA algorithm of [25]
- Perform down sampling using equation (8)
- Calculate the SIRs of each un-mixed data block by using equation (9)
- Calculate the SIRavg through using equation (10)
- Define the value of μ
- Compare the SIRavg with μ, if $SIR_{avg} > \mu$ then go to steps 1, 3, 5, and keep the same values of Z and Q, otherwise increase them up to the desired values.

Computational complexity of the FastICA algorithm utilizing the existing MIMO system shown in Fig 1 is O(ML). The proposed transceiver system has the complexity of O(MLZQ). Though, the proposed transceiver system will increase the computational complexity as compared to the complexity of the existing MIMO system in time varying channels. However, the proposed system gives increased accuracy of the results in time varying scenario, which is shown in Section 4. Moreover, in case if the channel is quasi static then the complexity of the proposed transceiver becomes equal to the computational complexity of the existing MIMO system, which we shown in detail in the automatic values adjustment (AVA) technique in **Fig. 4**. In short, we can say that the computational complexity and bandwidth utilization of the proposed system increases if the channels become highly time varying, where the channels are varying for each transmitted source symbol.

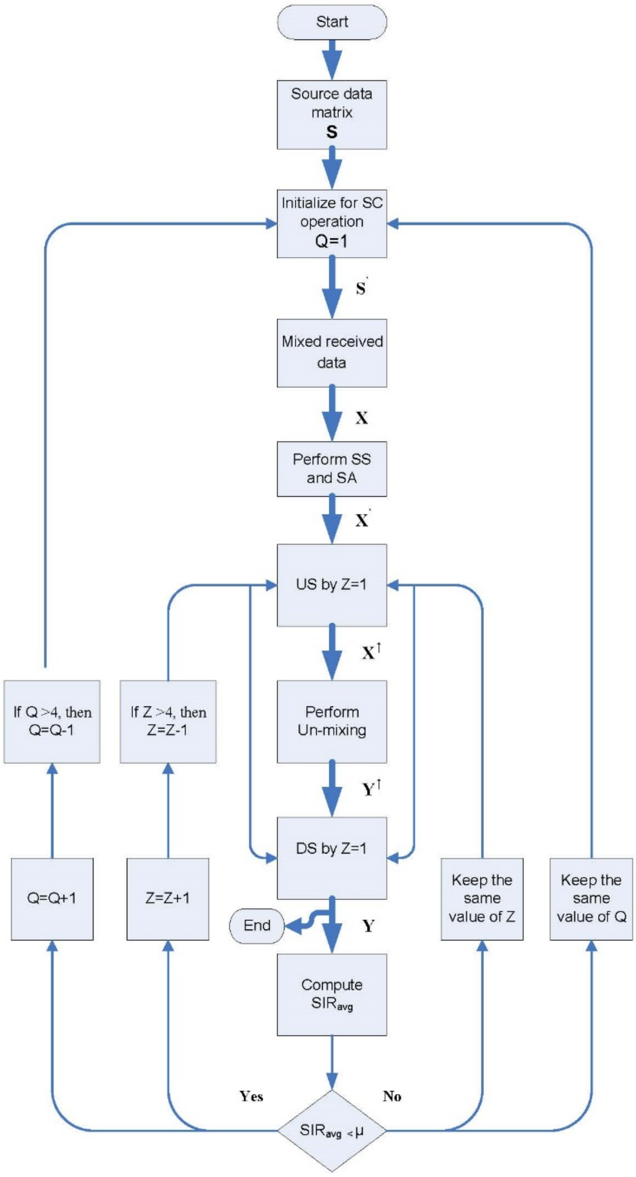

Fig. 4. The automatic values adjustment (AVA) technique for the key parameters Q and Z.

4. Results and Discussion

In this section, the performance of the proposed transceiver system is simulated and discussed. A dual antenna MIMO transceiver system is considered for simulation purpose. The transmitter transmits quadrature amplitude modulated (QAM) signals with equal carrier frequencies and power. Signal to noise ratio (SNR) ranges from -3dB to 10dB, throughout the simulations. Additive white Gaussian noise (AWGN) channels with frequency flat fading are used. The receiver receives linear mixtures of the transmitted source signals.

We consider both the time varying and the quasi static wireless channels. In case of time varying channels the condition of the quasi stationarity becomes invalid even for smaller lengths of the processing data blocks and the coefficients of the channel matrix $\boldsymbol{H}$ are changing within the processing data blocks. The channel matrix used in our simulation is given below,

$$\boldsymbol{H} = \begin{bmatrix} \delta_1 & \delta_2 \\ \delta_3 & \delta_4 \end{bmatrix} \tag{11}$$

where coefficients δ_1, δ_2, δ_3 and δ_4 of the matrix $\boldsymbol{H}$ are complex valued random variables. We generate them randomly in our simulation for time varying as well as quasi static channels. In quasi static case, we keep the channel coefficients fixed during the processing data blocks, while in time varying scenario these coefficients are changed randomly for each transmitted data sample.

The performance measure used in our simulation is signal to interference ratio (SIR). Simulation is performed using Monte Carlo simulation for each input data block. The input data blocks lengths ranges from 20 to 200 samples.

Different parameters used in our simulation are summarized in **Table 3**.

Table 3. Simulation parameters

Parameters	**Values**
Constellation	16 QAM
M	2
L	20 to 200
SNR	-2 to 10dB
Z	1 to 4
Q	1 to 4

In our simulations, first, we evaluate the performance of the proposed MIMO SCSA-UD transceiver system utilizing the quasi static as well as the time varying wireless channels, without utilizing the automatic value adjustment (AVA) technique. SNR is kept constant with a value of 10dB. In this simulation setup, we keep the value of Q equal 1 to clearly observe the effect of up/down sampling parameter Z with the values ranges from 1 to 4. We consider the smaller values of Z to protect oversampling and also to reduce the computational complexity of the proposed transceiver system. For detailed discussion regarding the value of Z one can refer to [24]. Results are demonstrated in **Fig. 5**. Performance improvements can be observed from this figure for different values of Z even in time varying channels.

Secondly, we evaluate the performance of the proposed system using the AVA technique. The AVA technique is used with fixed value of Q equal 1, for variable blocks lengths. Other parameters of the simulation setup are kept same as in the above experiment. Simulation is performed over quasi static and time varying conditions with different values of the predefined threshold. In case of quasi static channels we use μ= 11, 8, 7, 6, and in case of time varying scenario we use μ= 8, 7, 6, 5 respectively. Results are given in **Fig. 6**. One can observe from this figure that once the value of the predefined threshold is set to any value, then the AVA technique updates the value of Z from 1 to the desired value depending upon the value of μ for different blocks lengths and channel conditions.

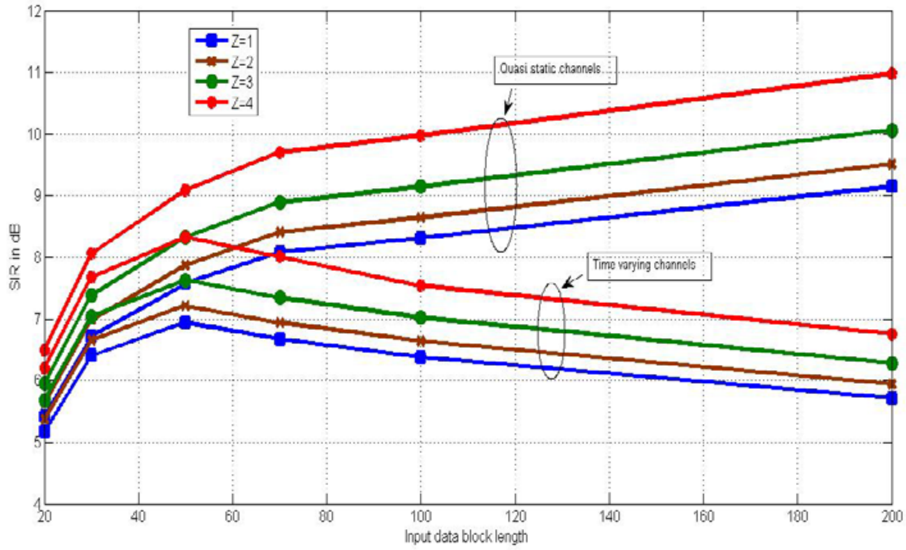

Fig. 5. Results of the proposed MIMO SCSA-UD transceiver utilizing the SC parameter Q=1, and changing the value of Z from 1 to 4, in quasi static as well as time varying channels conditions.

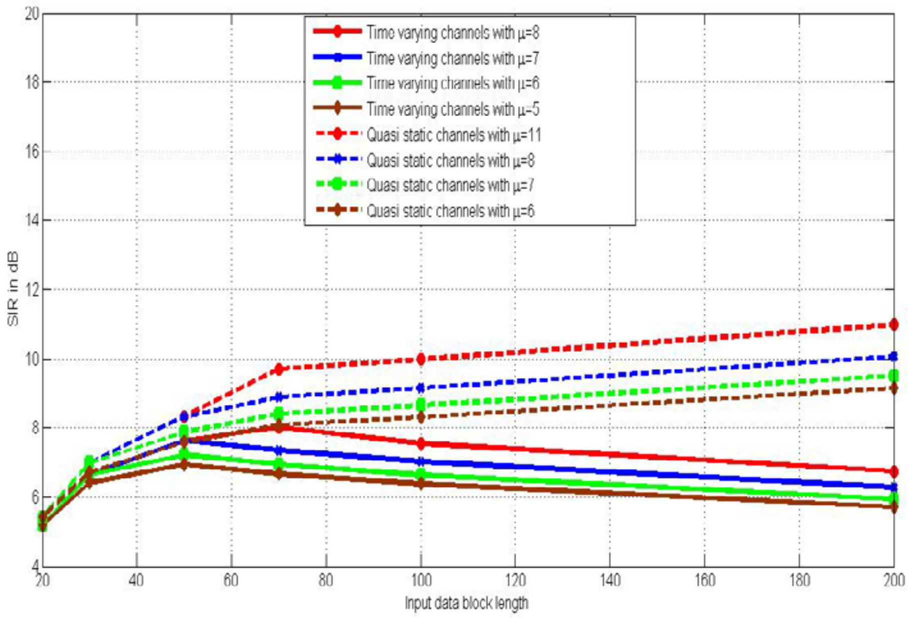

Fig. 6. Performance of the proposed transceiver system utilizing the AVA technique with the value of Q equal 1, and Z changes from1 to 4 in both the quasi static and time varying scenarios.

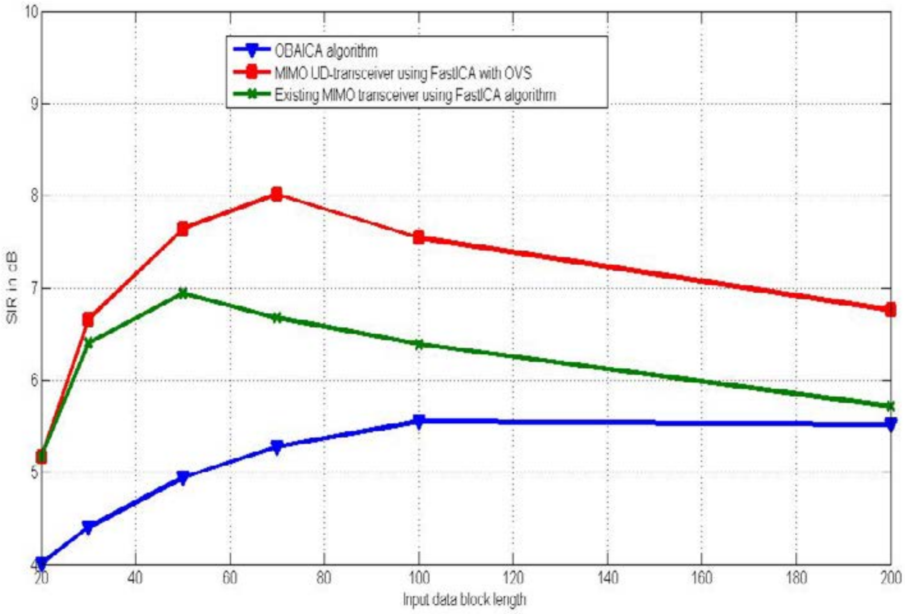

Fig. 7. Performance comparison of the proposed system with the FastICA and OBAICA algorithms in time varying channels. The AVA technique is used in the proposed system with Q=1, and μ=8, where values of Z changes from 1 to 4.

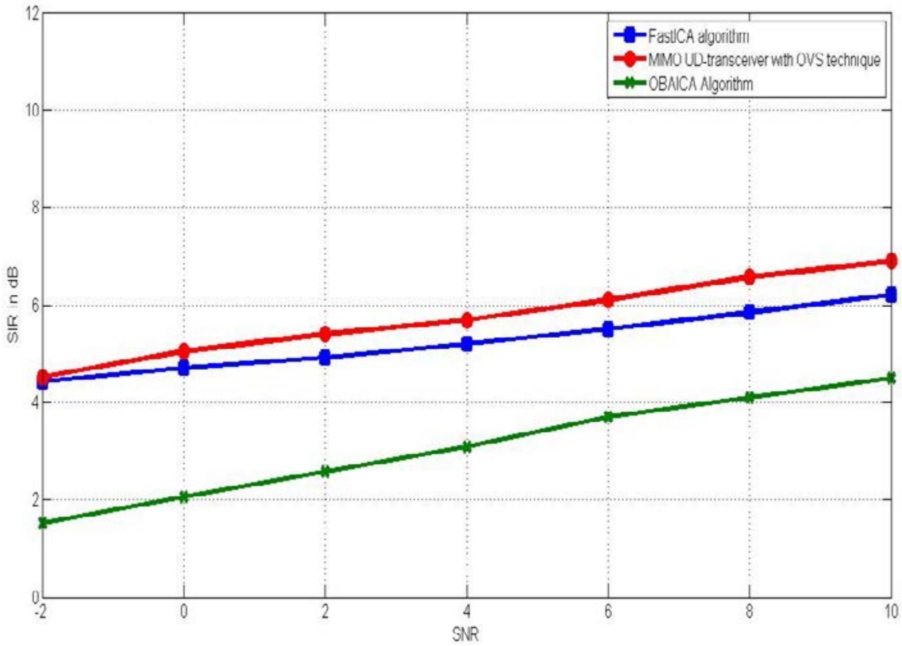

Fig. 8. Performance of the proposed transceiver system in comparison with the existing MIMO system utilizing the FastICA and OBAICA algorithms in both the transceivers for a block length of 30 samples and SNRs ranges from -2 to 10dB.

In the third simulation setup, performance of the MIMO SCSA-UD transceiver using the AVA technique with Q=1, and μ= 8, is compared with the FastICA [25] and the OBAICA [16] algorithms for different blocks lengths and fixed SNR of 10dB, in time varying scenario. Results are shown in **Fig. 7**. Performance improvement of the MIMO SCSA-UD transceiver can be observed from this figure even at larger blocks lengths in time varying scenario. Actually, for smaller data blocks, the expectation operator used in the update rule of the algorithm fails to produce correct results. By increasing the data block length, the accuracy of the results also increases. However, in a time varying scenario, the batch ICA algorithms are unable to track the variations and at larger block length the variations increases. Due to this reason, in our simulation, first the accuracy of the results increases and then decreases. The important point is that the results of the proposed system are better at smaller as well as larger data blocks.

Furthermore, we also compared the results for different SNRs utilizing a constant block length of 30 samples. Results are given in **Fig. 8**. This figure shows superior performance of the proposed system for different values of the SNRs and smaller data block length of 30 samples.

As a cross check we compared the bit error rate (BER) performance of the proposed transceiver with the FastICA and OBAICA algorithms for data block length of 30 samples. Other parameters of the simulation setup are kept same as in the previous experiment. Results are demonstrated in **Fig. 9** where the proposed system shows superior performance in terms of BER for a block length of 30 samples and different SNRs. It must be noted that the value of Q=1, and Z varies from 1 to 4.

In the next simulation setup, we consider the proposed system with different values of Q and Z, where both the values range from 1 to 4. We manually adjust these parameters to clearly observe their effect over un-mixing. The value of Q also kept smaller due to more bandwidth utilization in time varying scenario. Results are given in **Fig. 10**. In these results first we keep the value of Q=2 and change the value of Z from 1 to 4. Secondly, we keep Q=4 and change Z from 1 to 4, where more improvements in results observed while increasing the value of Q from 2 to 4 for larger blocks lengths in time varying scenario.

In **Fig. 11**, we demonstrate the results of the proposed system with the AVA technique. Different values of the predefined threshold μ are used which are 5, 7, and 9. SNR is kept constant with the value of 10dB. The channels are time varying. The AVA technique continuously checks the value of SIR_{avg} in comparison with the predefined threshold μ i.e, for μ=5 the SIR_{av} is greater than 5 for initial block length of 20 samples, so the values of Q and Z remained 1. In case of μ=7, and 9, the values of Q and Z are increased from 1 to the desired values, because the SIR_{avg} is less than 7 or 9 for block length of 20 samples. In this simulation more performance improvements are observed for μ=7, and 9, which is the main contribution of the proposed system. Moreover, the proposed system is also evaluated for different SNRs, with the data block length of 100 samples. Other parameters are same as in the above experiment. Results are given in **Fig. 12**, which shows performance improvements for larger value of μ.

Finally, we demonstrate the performance of the MIMO SCSA-UD transceiver (with μ=7) in comparison with the existing MIMO system utilizing the FastICA and OBAICA algorithms in both the transceivers for time varying scenario. Results are tabulated in **Table 4**. Results shown in this table are such that the SIR performance of the FastICA algorithm degrades for increased length of the processing data blocks while performance of the proposed transceiver further improves. This is the main achievement of the proposed transceiver system.

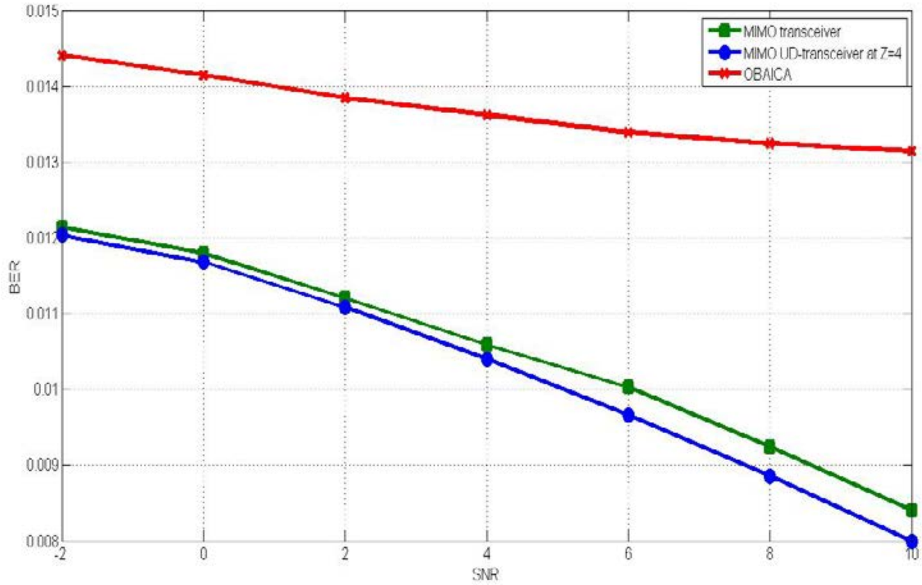

Fig. 9. Bit error rate (BER) performance of the proposed system in comparison with the FastICA and OBAICA algorithms.

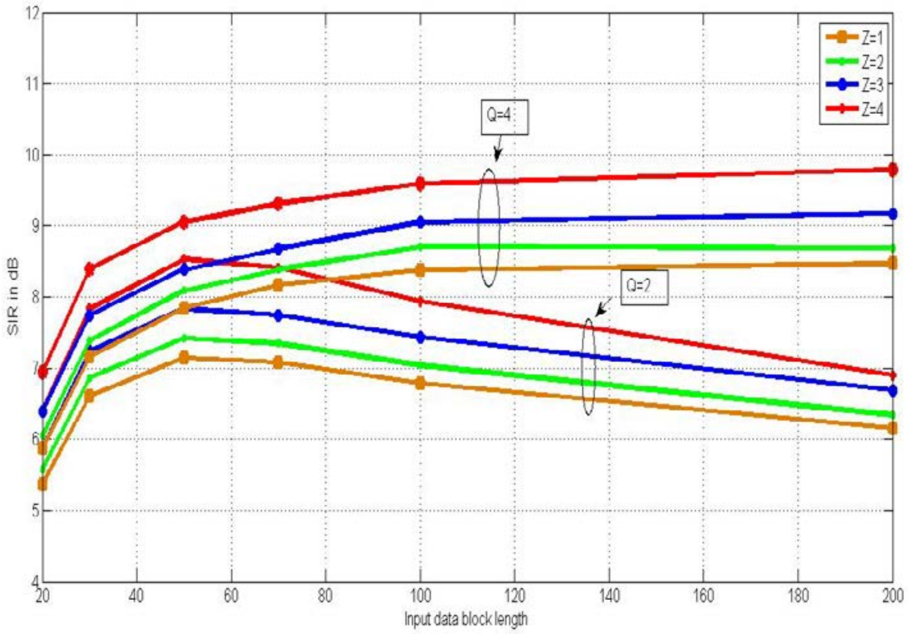

Fig. 10. Results of the proposed MIMO SCSA-UD transceiver system for different values of Q and Z without using the AVA technique**.**

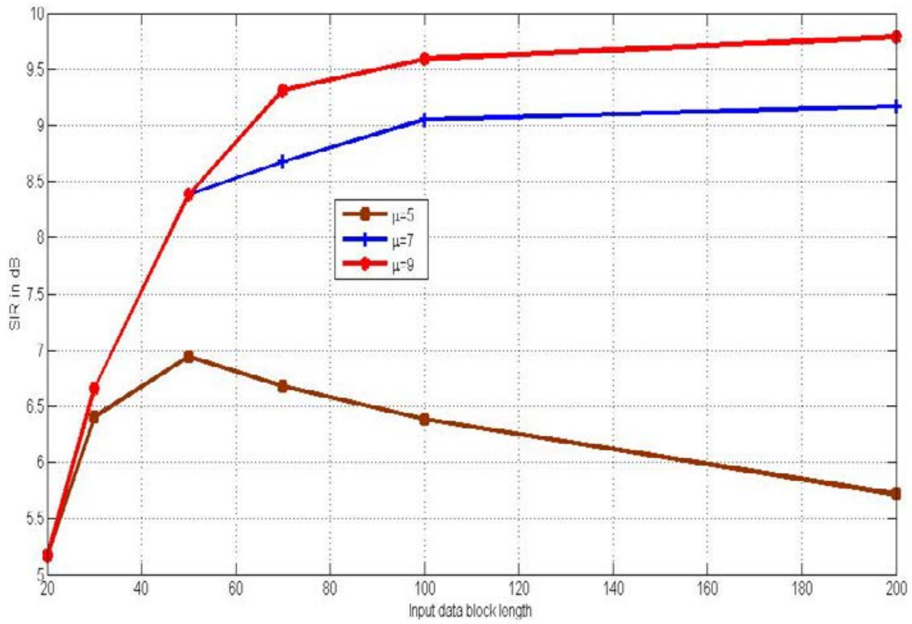

Fig. 11. Results of the proposed MIMO SCSA-UD transceiver system utilizing the AVA technique for μ=5, 7, and 9.

The results for different SNRs with block length of 100 are given in **Fig. 12**. This figure also shows superior performance of the proposed MIMO SCSA-UD transceiver system for time varying wireless scenario.

Table 4. The SIR performance comparison of the proposed transceiver system with the FastICA and the OBAICA algorithms in time varying scenario.

S No.	Data blocks lengths (L)	FastICA Algorithm	MIMO SCSA-UD transceiver with μ=9	OBAICA Algorithm
1	20	5.2 dB	5.3 dB	4.0 dB
2	30	6.4 dB	6.8 dB	4.3 dB
3	50	7.0 dB	8.4 dB	5.0 dB
4	70	6.8 dB	9.4 dB	5.2 dB
5	100	6.6 dB	9.6 dB	5.6 dB
6	200	5.8 dB	9.8 dB	5.7 dB

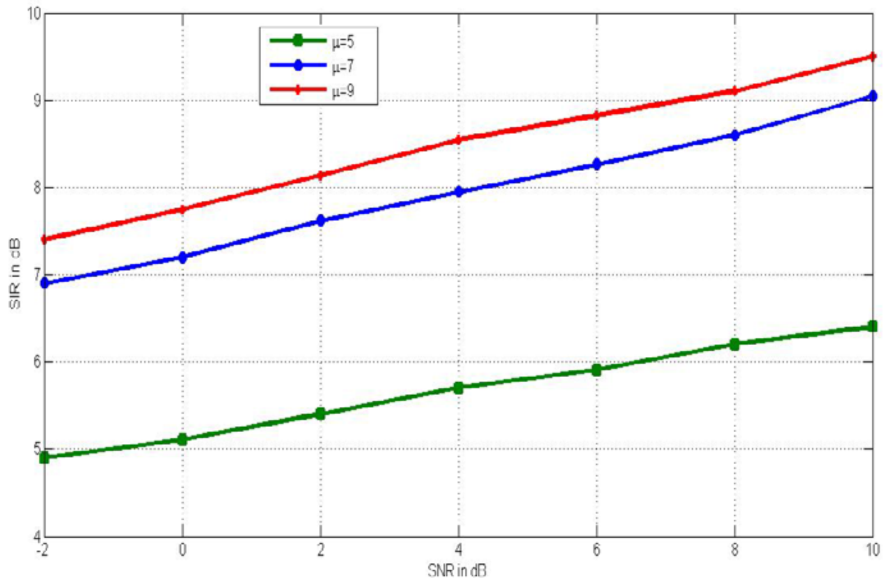

Fig. 12. Results of the proposed transceiver system utilizing the AVA technique with μ=5, 7, and 9 for data block length of 100 samples and SNRs ranges from -2 to 10dB.

5. Conclusion

In most of the ICA algorithms implemented in wireless communication systems, the channels are considered static or quasi static. Its use in time varying channels to blindly estimate the original signal from the mixed received signals is limited and inefficient. In this paper, the MIMO SCSA-UD transceiver system is proposed that performs well in time varying scenario. The channels are assumed highly time varying with the channel characteristics changing for each transmitted sample. We use the complex valued FastICA algorithm as a benchmark to evaluate the effectiveness of the proposed system. Simulation is performed over QAM modulated signals for various SNRs and input data block lengths. Performance of the MIMO SCSA-UD transceiver is compared with the OBAICA algorithm and the FastICA algorithm. Results show that the proposed transceiver outperforms the OBAICA and FastICA algorithms in highly time varying wireless channels.
As a future work, we would like to extend the proposed approach for non-square case utilizing the techniques of non-square mixing scenario.

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