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Blind Channel Estimation and Equalization

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Abstract

This paper resolves the problem of channel estimation and identification of a non-minimum phase system using three algorithms. These algorithms play an important role for estimating blindly the parameters of radio mobile channel. Thus studying the blind's problem channel equalization based on the following, Proposed Algorithm, CMA and SKMAA equalizers. The simulation results in the noisy environment and for different SNR values demonstrate that Proposed Algorithm is more performing than the other algorithms. In addition, the Sign Kurtosis Maximization Adaptive Algorithm (SKMAA) is more powerful compared to Constant Modulus Algorithm (CMA), that is to say it gives the right blind channel equalization.

Keywords: *Adaptive blind equalization, Blind channel Estimation, Higher Order Cumulants (HOC), CMA, SKMAA, SNR, SER*

1. Introduction

From the output measurements, the channel system identification is a well defined problem in several science and engineering areas such as speech signal processing, adaptive filtering, spectral estimation, communication[1-3]. Signal processing techniques using Higher-Order Statistics (HOS) or cumulants have attracted a considerable attention in the literature [4]. The information contained in the second-order statistics (autocorrelation functions) would suffice for the complete statistical description of a Gaussian signal. However, there are a practical situation where we have to look beyond the autocorrelation of a signal to extract information regarding deviations from Gaussianity and presence of phase relation. Thus the interest in higher order cumulants (HOC) or higher order statistics (HOS) is permanently growing in the last years [5]. Also the finite impulse response system identification based on HOC of system output has received more attention. Actually the blind equalization of radio mobile channel has become an important topic in digital communications. that why, we try to study the popular adaptive blind equalization Bussgang algorithm (the Godard algorithm [6]) or Constant Modulus Algorithm (CMA) [7], and Sign Kurtosis Maximization Adaptive Algorithm (SKMAA) [8] in the context of non constant modulus data with spatio-temporal diversity. The paper also describes three algorithms, Safi1, Safi2 and Said1, which are used to estimate the parameters of radio mobile channel in noisy and non noisy case. Moreover we attest the blind equalization of channel using the Constant Modulus Algorithm (CMA), Sign Kurtosis Maximization Adaptive Algorithm (SKMAA) and Said1 algorithms.

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2. Problem Statement

The output of a Finite Impulse Response (FIR) channel, excited by an unobservable input sequences, i.i.d. zero-mean symbols with unit energy, across a selective channel with memory p and additive noise (Figure 1) is given by the following relationship.

$$X(k) = h_p(k)S(k) + V(k \tag{1}$$

Where $h_p = [h(1), h(2), \dots, h(p)]$ represents the channel parameter, $S(k)$ input system and $V(k)$ is the additive colored Gaussian noise.

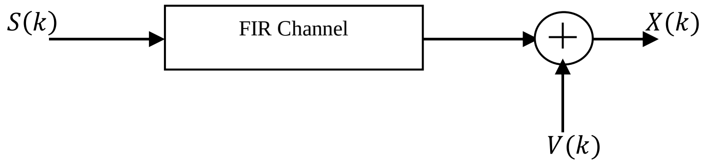

Figure 1. Channel Model

3. Blind Algorithms

3.1. Safi1 Algorithm[9]

Hypothesis:
Let us suppose that:

- The additive noise $V(k)$ is Gaussian, colored or with symmetric distribution, zero mean, with variance σ^2, i.i.d. with the mth order cumulants vanishes for m > 2
- The noise $V(t)$ is independent of $S(k)$ and $X(k)$.
- The channel (FIR system) order p is supposed to be known and $h(0) = 1$.
- The system is causal, i.e., $h(i) = 0$ if $i < 0$.
- The mth order cumulant of the output signal is given by the following equation [5], [10], [11]:

$$C_{my}\,(t_1, \dots, t_{m-1}) = \gamma_{mx} \sum_{-\infty}^{+\infty} h(i)h(i+t_1)\cdots h(i+t_{m-1}) \tag{2}$$

Where γ_{mx} represents the mth order cumulants of the excitation signal ($S(k)$). Based on equation (2), Safi1 algorithm is developed by the following relationship:

$$\sum_{i=0}^{p} h(i)C_{3y}(t_1 - i, t_2 - i) = \epsilon \sum_{i=0}^{p} h(i)h(i + t_2 - t_1)C_{2y}(t_1 - i) \tag{3}$$

If we use the property of the ACF of the stationary process, such as $C_{2y}(t) \neq 0$ only for $(-p \leq t \leq p)$ and vanishes else were.

$$\sum_{i=0}^{p} h(i)C_{3y}(-p - i, t_2 - i) = \epsilon h(0)h(t_2 + p)C_{2y}(-p) \tag{4}$$

Else, if we suppose that $t_2 = -p$, Eq. (4) will become

$$C_{3y}(-p, -p) = \epsilon h(0)C_{2y}(-p) \tag{5}$$

Using Eqs. (4) and (5) we obtain the following relation

$$\sum_{i=0}^{p} h(i)C_{3y}(-p-i, t_2-i) = h(t_2+p) \tag{6}$$

Else, if we suppose that the system is causal, i.e., $h(i) = 0$ if $i < 0$. So, for $t_2 = -p, \ldots, 0$, the system of Eq. (6) can be written in matrix form as:

$$\begin{pmatrix} C_{3y}(-p-1,-p-1) & \cdots & C_{3y}(-2q\,,-2q) \\ C_{3y}(-p-1\,,-p)-\alpha & \cdots & C_{3y}(-2p,-2p+1) \\ \vdots & \ddots & \vdots \\ C_{3y}(-p-1,1) & \cdots & C_{3y}(-2p,-p)-\alpha \end{pmatrix} \begin{pmatrix} h(1) \\ h(2) \\ \vdots \\ h(p) \end{pmatrix}$$

$$= \begin{pmatrix} 0 \\ -C_{3y}(-p,-p+1) \\ \vdots \\ -C_{3y}(-p,0) \end{pmatrix} \tag{7}$$

Were $\alpha = C_{3y}(-p,-p)$.
The above Eq. (7) can be written in compact form as

$$Mh_p = d_1 \tag{8}$$

Where M is the matrix of size $(p+1) \times (p)$ elements $h_p$is a column vector constituted by the unknown impulse response parameters $h(n)$: $n = 1, \ldots, p$ and d_1 is a column vector of size $(p+1) \times (1)$ as indicated in the Eq. (7). The least squares solution (LS) of the system of Eq. (8), permits blindly identification of the parameters $h(n)$ and without any "information" of the input selective channel. So, the solution will be written under the following form

$$h_p = (M^T M)^{-1} M^T d_1 \tag{9}$$

3.2. Safi2 Algorithm[9]

From the Eq. (2), the mth and nth cumulants of the output signal, $\{y(n)\}$, and the coefficients $\{h(i)\}$, where $n > m$, are linked by the following relationship:

$$\sum_{j=0}^{p} h(j)C_{ny}(j+t_1, \ldots, j+t_{m-1}, t_m, \ldots, t_{n-1})$$
$$= \frac{\gamma_{ne}}{\gamma_{me}} \sum_{i=0}^{p} h(i) \left[\prod_{k=m}^{n-1} h(i+t_k)\right] C_{my}(i+t_1, \ldots, i+t_{m-1}) \tag{10}$$

If we take $n = 4$ and $m = 3$ into Eq. (10), we find the basic relationship developed in [12,13]. If we take $n = 3$ and $m = 2$ into Eq. (9), we find the basic relationship of the algorithms developed in [14]. So, the equation proposed in [15] presents the relationship between different nth cumulant slices of the output signal $\{X(n)\}$, as follows

$$\sum_{j=0}^{p} h(j) \left[\prod_{k=1}^{r} h(j+t_k)\right] C_{ny}(\beta_1, \ldots, \beta_r, j+\alpha_1, \ldots, \alpha_{n-r-1})$$
$$= \sum_{i=0}^{p} h(i) \left[\prod_{k=1}^{r} h(i+\beta_k)\right] C_{ny}(t_1, \ldots, t_r, i+\alpha_1, \ldots, i+\alpha_{n-r-1}) \tag{11}$$

Where $1 \le r \le n-2$.

If we take n=3 we obtain that r=1, so the Eq. (10) will be

$$\sum_{j=0}^{p} h(j)h(j+t_1)C_{3y}\,(\beta_1, j+\alpha_1) = \sum_{i=0}^{p} h(i)h(i+\beta_1)C_{3y}\,(t_1, i+\alpha_1) \tag{12}$$

In the following, we develop an algorithm based only on 4th order cumulants. If we take $n = 4$ into Eq. (11) we obtain the following equation:

$$\sum_{i=0}^{p} h(i)h(i+t_1)h(i+t_2)\, C_{4y}(\beta_1, \beta_1, i+\alpha_1) = \sum_{j=0}^{p} h(j)h(j+\beta_1)h(j+\beta_2)C_{4y}\,(t_1, t_2, +\alpha_1) \tag{13}$$

If $t_1 = t_2 = p$ and $\beta_1 = \beta_2 = 0$, Eq. (13) takes the form:

$$h(0)h^2(p)C_{4y}(0,0,i+\alpha_1) = \sum_{j=0}^{p} h^3\,(j)C_{4y}(p,p,j+\alpha_1) \tag{14}$$

As the system is a FIR, and is supposed causal with an order p, so, the $j+\alpha_1$ will be necessarily into the interval $[0,p]$, this implies that the determination of the range of the parameter α_1 is obtained as follows: $0 \le j + \alpha_1 \le p \Rightarrow -j \le \alpha_1 \le q - j$, and we have $0 \le j \le p$. From these two inequations, we obtain:

$$-p \le \alpha_1 \le p. \tag{15}$$

Then, from the Eqs. (13) and (14) we obtain the following system of equations:

$$\begin{pmatrix} C_{4y}(p,p,-p) & \cdots & C_{4y}(p\,,p,0) \\ \vdots & \ddots & \vdots \\ C_{4y}(p,p,0) & \cdots & C_{4y}(p,p,p) \\ \vdots & \ddots & \vdots \\ C_{4y}(p,p,p) & \cdots & C_{4y}(p,p,2p) \end{pmatrix} \begin{pmatrix} h^3(0) \\ \vdots \\ h^3(i) \\ \vdots \\ h^3(p) \end{pmatrix} = h(0)h^2(p) \begin{pmatrix} C_{4y}(0,0,-p) \\ \vdots \\ C_{4y}(0,0,0) \\ \vdots \\ C_{4y}(0,0,p) \end{pmatrix} \tag{16}$$

and as we have assumed that $h(0) = 1$, if, we consider that $h(p) \ne 0$ and the cumulant $C_{my}(t_1, \ldots, t_{m-1}) = 0$, if one of the variables $t_k > p$, where $k = 1, \ldots, m-1$, the system of Eq. (16) will be written as follows:

$$\begin{pmatrix} 0 & & \cdots & 0 & C_{4y}(p,p,0) \\ \vdots & & \ddots & & \vdots \\ 0 & & & & \\ C_{4y}(p,p,0) & \cdots & & & C_{4y}(p,p,p) \\ \vdots & & \ddots & & 0 \\ & & & & \vdots \\ C_{4y}(p,p,p) & 0 & \cdots & & 0 \end{pmatrix} \begin{pmatrix} \frac{1}{h^2(p)} \\ \vdots \\ \frac{h^3(i)}{h^2(p)} \\ \vdots \\ \frac{h^3(p)}{h^2(p)} \end{pmatrix} = \begin{pmatrix} C_{4y}(0,0,-p) \\ \vdots \\ C_{4y}(0,0,0) \\ \vdots \\ C_{4y}(0,0,p) \end{pmatrix} \tag{17}$$

In more compact form, the system of Eq. (17) can be written in the following form:

$$Mb_p = d_2 \tag{18}$$

where M, b_p and d_2 are defined in the system of Eq. (17). The least squares solution of the system of Eq. (18) is given by:

$$\hat{b}_p = (M^T M)^{-1} M^T d_2 \tag{19}$$

This solution give us an estimation of the quotient of the parameters $h^3(i)$ and $h^3(p)$, with $b_p(i) = \left(\frac{\hat{h}^3(i)}{h^2(p)}\right), i = 1, \ldots, p$

So, in order to obtain an estimation of the parameters $\hat{h}(i), i = 1, \ldots, p$ we proceed as follows:

- The parameters $h(i)$ for $i = 1, \ldots, p-1$ are estimated from the estimated values $b_p(i)$ using the following equation:

$$\hat{h}(i) = sign\left[\hat{b}_p(i)\left(\hat{b}_p(p)\right)^2\right]\left\{abs\left(\hat{b}_p(i)\right)(\hat{b}_p(p))^2\right\}^{\frac{1}{3}} \tag{20}$$

where $sign(x) = \begin{cases} 1, & if \quad x > 0; \\ 0, & if \quad x = 0; \\ -1 & if \quad x < 0. \end{cases}$

and $abs(x) = |x|$ indicates the absolute value of x

- The $\hat{h}(p)$ parameters is estimated as follows :

$$\hat{h}(p) = \frac{1}{2} sign\left[\hat{b}_p(p)\right]\left\{abs\left(\hat{b}_p(p)\right) + \left(\frac{1}{\hat{b}_p(1)}\right)^{1/2}\right\} \tag{21}$$

3.3 Proposed algorithm Said1

The general problem of interest here may be characterized using the following model:

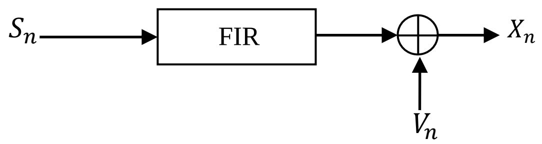

Figure 2. System Model

The convolution model is represented by:

$$X_n = \sum_{l=0}^{L} h_l S_{n-l} + V_n \tag{22}$$

h_l represents the channel parameter, S_{n-l} input system, X_n output system and V_n is the additive colored Gaussian noise.

Or model vector of equation (22) is:

$$X_n = [h_0 \ldots h_l]\begin{bmatrix} S_n \\ \vdots \\ S_{n-l} \end{bmatrix} + V_n$$

$$X_n = h^T S_n + V_n \tag{23}$$

The matrix system will be written as follows:

$$\begin{bmatrix} X_n \\ \vdots \\ X_{n-N} \end{bmatrix} = \begin{bmatrix} h_0 & \ldots & h_l & 0 & 0 \\ 0 & \ddots & \ddots & \ddots & 0 \\ 0 & 0 & h_0 & \ddots & h_l \end{bmatrix}\begin{bmatrix} S_n \\ \vdots \\ S_{n-N-l} \end{bmatrix} \tag{24}$$

Or

$$X(n) = h(n).S(n) + V(n) \tag{25}$$

With $X(n)$ of size $(N+1)\times 1$, $h(n)$ is of size $(N+1)\times(N+L+1)$ and $S(n)$ of size $(N+L+1)\times 1$. The Said1 algorithm is a general method for blind solving estimation problems of channel parameters. The considerations leading to the Said1 algorithm is given below.

- Let $S_0, \dots., S_M$ be known
  Let N=L,
  Then:

$$E[X(n)S_{n-L}^{*}] = \begin{bmatrix} h_0 \\ \vdots \\ h_L \end{bmatrix} \tag{26}$$

  Where in practice,

$$E[X(n)S_{n-L}^{*}] = \frac{1}{M-L+1}\left(\sum_{n=L}^{M} X(n)S_{n-L}^{*}\right) \tag{27}$$

- Calculate mean square error of channel estimation:
  From equation (26), we have estimation:

$$\hat{h} = \begin{bmatrix} h_0 \\ \vdots \\ h_L \end{bmatrix} \tag{28}$$

  Normalization of h and $\hat{h}$ is given by:

$$h = {h}/{\|h\|}, \qquad \hat{h} = {\hat{h}}/{\|\hat{h}\|}$$

  So, MSE is calculated by the following equation:

$$MSE = \|h - \hat{h}\| \tag{29}$$

  The principle of this algorithm is given by the following diagram:

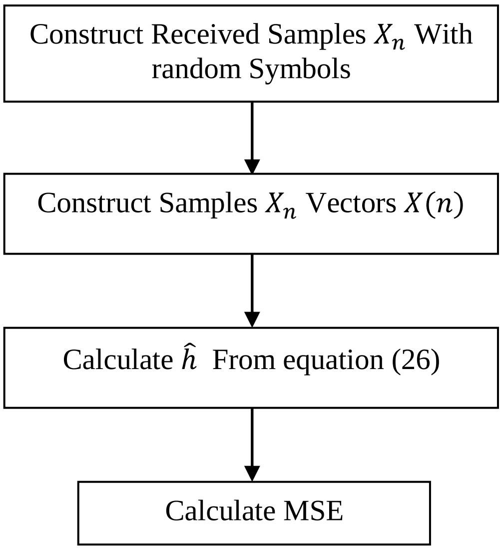

Figure 3. Diagram of Proposed Algorithm (Said1)

4. Said1Equalization

4.1. ZF Equalization

In order to compensate the effect of power line communication (PLC) channel, we consider the ZF equalizer, wich satisfies the condition shown below.

$$f_{ZF} H = I \tag{30}$$

Where $f_{ZF} = (H^H H)^{-1} H^H$ is the ZF decoding matrix, $(.)^H$ denotes Hermitian transpose, H is the channel matrix and I is the identity matrix. Given the received signal X, the receiver can obtain the estimated signal by using the ZF equalization, which is given by:

$$\hat{S}(n) = f_{ZF}\, X(n) \tag{31}$$

$$\hat{S}(n) = f_{ZF} H\, S(n) + f_{ZF}\, V(n)$$
$$\hat{S}(n) = I\, S(n) + f_{ZF}\, V(n)$$

Where $\hat{S}$ is an estimate matrix of the transmitted signal. If the determinant of H is not zero so that there exists the inverse matrix of H the decoding matrix can be expressed as

$$f_{ZF} = H^{-1} \tag{32}$$

The ZF equalization is ideal when the channel is noiseless. However, when the channel is noisy, the ZF equalization plays an important role to cancel the additive colored Gaussian noise.

4.2 MMSE equalization

In order to minimize the power of the noise component, we employ the MMSE equalization, which is given by:

$$f_{MMSE} = \underbrace{min}_{f_{MMSE}}\; E\left|\left\|S_{n-d} - f_{MMSE}^H X(n)\right\|^2\right| \tag{33}$$

Where f_{MMSE} is the MMSE decoding matrix and $\|.\|$ the norm of $S_{n-d} - f_{MMSE}^H X(n)$.
Or:

$$f_{MMSE} = R_X \hat{h}_d \tag{34}$$

Where $R_X = E[X(n)X^H(n)]$ and $\hat{h}_d = E[X(n)S^*_{n-d}]$. In practice the expression of R_X and $\hat{h}_d$ is:

$$R_X = \frac{1}{k}\sum_{n=1}^{k} X(n)X^H(n) \tag{35}$$

$$\hat{h}_d = \frac{1}{M}\sum_{n=1}^{M} X(n)S^*_{n-d} \tag{36}$$

The MMSE equalization can be used to estimate the channel effect with varying the decoding matrix in accordance with SNR. Besides, it prevents the noise component from being amplified.

So, equalization said algorithm is defined by the steps following:

- Transmit k symbols, the first M is assured known,
- Obtain received samples X_n,
- Construct sample vectors $X(n)$,
- Calculate R_X and $\hat{h}_d$,
- Calculate $f_{MMSE} = R_X^{-1}\hat{h}_d$,
- Calculate symbol error rate (SER):
  - ✓ Use $f^H X(n)$ to estimate symbol $\hat{S}_{n-d}$.
  - ✓ Compare $\hat{S}_{n-d}$ with S_{n-d} .

5. Adaptive Equalization

One approach to removing inter-symbol interference in a communications channels is to employ adaptive blind equalization to reduce the Symbol Error Rate (SER). The most popular class of algorithms used for blind equalization are those that minimize the Godard (or constant modulus) criteria [6,16]. In this paper we study the performance of the Constant Modulus Algorithm (CMA).

5.1. The Constant Modulus Algorithm (CMA)

The problem with blind adaptation techniques is their poor convergence property compared to traditional techniques using training sequences. Generally a gradient descent based algorithm is used with the blind adaptation schemes. The most commonly used gradient descent based blind adaptation algorithm is the Constant Modulus Algorithm (CMA). The Constant Modulus Algorithm (CMA) [16,17] has gained widespread practical use as a blind adaptive equalization algorithm for digital communications systems operating over inter-symbol interference channels.

The constant modulus (CM) criterion can be expressed by the cost function $J_{CM} = \frac{1}{4}E\{(|X_n|^2 - \gamma)^2\}$, where γ is a positive constant known as the Godard radius [6]. The equalizer update algorithm leading to a stochastic gradient descent of J_{CM} is known as the Constant Modulus Algorithm (CMA) and is specified by:

$$f(n+1) = f(n) + \mu S^*(n)\underbrace{X_n(\gamma - |X_n|^2)}_{\triangleq\psi(X(n))} \tag{37}$$

Where μ is a step-size and $S^*(n)$ is the equalizer input vector at time index n. the asterisk denotes conjugation. The function $\psi(.)$ identified in (37) is referred to as the CMA error function.

5.2. KMAA Algorithm

Sign Kurtosis Maximization Adaptive Algorithm (SKMAA) is used to blind channel equalization, that is to say SKMAA is developed based on kurtosis of stochastic signals, the stochastic ascend approach, and sign algorithm for restoring the blind equalizer weight vectors [8]. Furthermore it is also necessary to calculate the output sample $y(n)$ of the equalizer filter once per algorithm update. The general idea is to maximize the chosen cost function. To assure that the SKMAA algorithm ascends for each time step n the following requirement is made

$$J\big(f(n+1)\big) > J(f(n)) \tag{38}$$

Or the cost function is given by

$$J_{KMA}(f) = \frac{|Kurtosis[y(n)]|}{E^2|y(n)|^2} \tag{39}$$

where $y(n)$ is the equalizer output and $Kurtosis[y(n)]$ is defined by:

$$Kurtosis[y(n)] = E\{y^4(n)\} - 3E^2\{y^2(n)\} \tag{40}$$

The update is then given by

$$\begin{aligned} f(n+1) &= f(n) + \mu\Delta f J_{KMA}(f) \\ &= f(n) + \mu F(y)[S(n) * h(n)] \\ &= f(n) + \mu F(y) X(n) \end{aligned} \tag{41}$$

or

$$\Delta f J_{KMA}(f) = F(y)X(n) \tag{42}$$

where F is the feedback function defined by:

$$F(y) = \frac{4(E[y^2]y^2 - E[y^4])y}{E^3[y^2]} \tag{43}$$

a Sign Algorithm is introduced into KMAA. So the SKMAA is written by:

$$f(n+1) = f(n) + \alpha Sign[F(y)] \cdot [X(n)] \tag{44}$$

where $\text{Sign}[.]$ is a simple sign function, and α is a forgetting factor which is used to replace the ensemble averages by empirical averages which are then adaptively updated $(0 < \alpha < 1)$[8].

6. Simulation Results

6.1. Performance of Safi1, Safi2 and Said1 algorithms in the without Noise Case

In this part, we will test the performance of the presented algorithms (Safi1, Safi2 and Said1) in order to estimate the channel's parameters. Table1 represents the obtained results using different data length and the Real parameters of Proakis (B) channel [18], by applying the modulation QPSK or 4 QAM symbol sequence channel without noisy case. In the simulation we utilized the relations (9), (20) and (26) with the matlab logiciel.

Table 1. Coefficients of Proakis (B) channel using the Three Algorithms

N	*Algorithms*	*h*(1)	*h*(2)	*h*(3)	*MSE*
512	*Safi1*	-0.1852	0.3324	0.1456	1.6804
	Safi2	0.669	0.8962	0.7719	1.2283
	Said1	0.3426	0.7944	0.4622	0.1140
1024	*Safi1*	-0.2575	0.5186	0.2792	1.3604
	Safi2	0.6464	0.9017	0.7657	1.1343
	Said1	0.3997	0.8082	0.4286	0.0630
2048	*Safi1*	0.0944	0.1009	0.1135	1.2774
	Safi2	0.5692	0.8614	0.6953	0.7639
	Said1	0.0553	0.8136	0.4065	0.0553
4096	*Safi1*	0.1204	0.4851	0.1675	0.7387
	Safi2	0.5004	0.8067	0.6329	0.5609
	Said1	0.4135	0.8131	0.4096	0.0140
	Real Parameters	0.407	0.815	0.407	

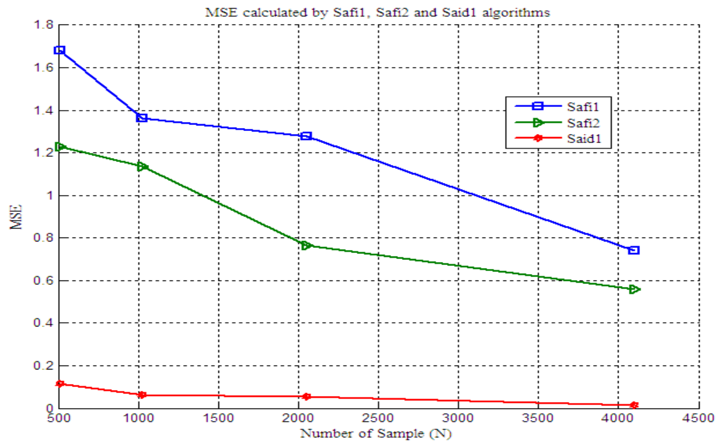

Figure 4. The Variation of the MSE using Safi1, Safi2 and Said1 Algorithms with Different Numbers of Samples N

From Table 1 we can conclude that for all numbers of samples (N = 512, 1024, 2048 and 4096), Said1algorithm gives satisfactory results so as to estimate the parameters of Proakis (B) channel if MSE value is very low, that is to say, Said1 has a good performance regarding to Safi1 and Safi2 algorithms.

6.2. Performance of Safi1, Safi2 and Said1 Algorithms in the Noise Case

From the previous section, we will compare the performance of the three presented algorithms (Safi1, Safi2 and Said1), with the Additive Gaussian Noise (AWGN). For this reason we select two values of Signal Noise Ratio (SNR=0 dB and SNR=16 dB).

Table 2. Parameters of Proakis (B) Channel in the Noisy case using the Three Algorithms with Number of Sample N = 1024

SNR	*Algorithms*	*h*(1)	*h*(2)	*h*(3)	*MSE*
0 dB	*Safi1* *Safi2* *Said1*	-0.1816 0.4383 0.4306	0.1849 0.6983 0.8051	0.2043 0.6086 0.4013	2.9375 0.2717 0.0783
16dB	*Safi1* *Safi2* *Said1*	-0.15650 0.4723 0.4040	0.2930 0.6833 0.8226	0.2445 0.4295 0.4001	2.4863 0.1922 0.0142
	Real Parameters	0.407	0.815	0.407	

In the similar way, we conclude that the third algorithm (Said1) gives a good estimation to channel's parameters which are very close to the real values, so Said1 algorithm is shown an outperformance than Safi1 and Safi2 algorithms.

6.3. Blind Identification Channel (Bran A)

In this section we are going to identify the amplitude and the phase of BRAN A channel [19], [20] using the number of samples N = 2048 and the SNR= 16 dB, to attest the performance of the Safi1, Safi2 and Said1 algorithms.

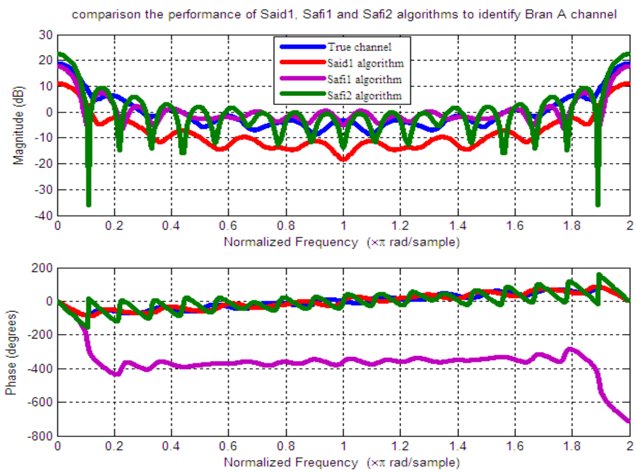

Figure 5. Identification of the Amplitude and the Phase BRAN A Channel's, using the Three Algorithms

The simulation result obtained (Figure 5) shows that both algorithms have very satisfactory results in terms of the amplitude of the impulse response compared to the real measure. Concerning the phase of impulse response is different from the real measure of Safi1, but is the same for Said1 and Safi2 algorithms, so Said1 algorithm gives a good identification to BRAN A channel.

6.4. Equalization of Channel by Said1, CMA and SKMAA Algorithms

In this part we attest the performance of Said1, CMA and SKMAA algorithms to equalize the output of channel, Using the different SNR values, the total number of data is T=1024 and the modulation QPSK or 4 QAM symbol sequence.

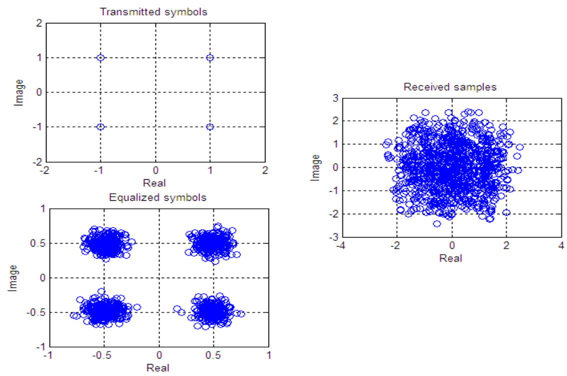

Figure 6. Channel Equalization by Said1 Algorithm, with the SNR=25 dB

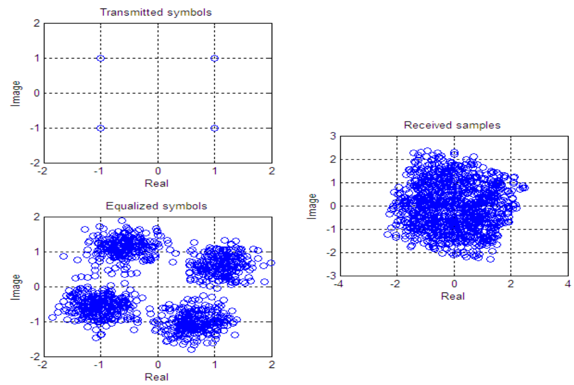

Figure 7. Channel Equalization by CMA Algorithm, with the SNR=25 dB

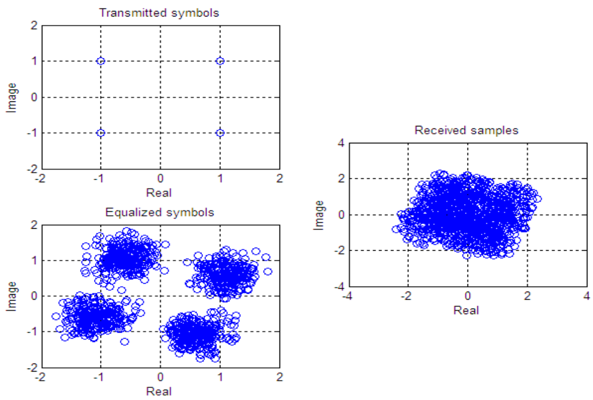

Figure 8. Channel Equalization by SKMAA Algorithm, with the SNR=25 dB

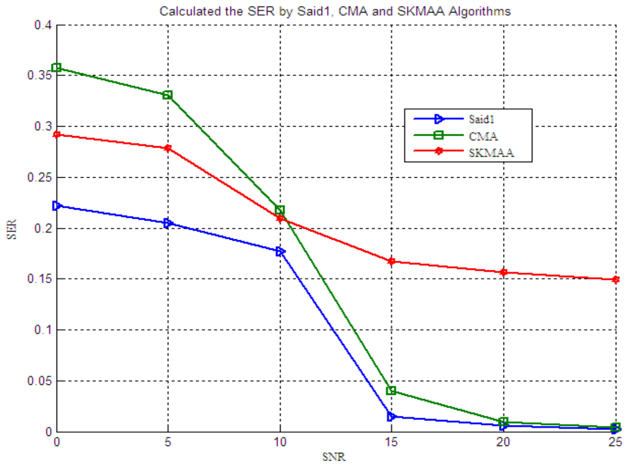

Figure 9. SER in Function the SNR Values using Said1, CMA and SKMAA Algorithms

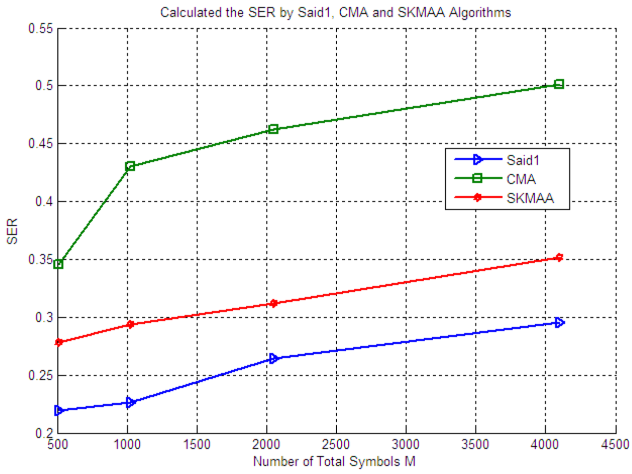

Figure 10. SER in Function the Number Total of Symbols M using Said1, CMA and SKMAA algorithms with SNR=6 dB

From the results of simulation obtained, we notice that the Said1 algorithm gives the good equalization of channel regarding to CMA and SKMAA algorithms. In addition, the SER is decreasing according to SNR values (Figure 9) and is increasing according to number total of symbols M (Figure 10) for the three algorithms (Said1, CMA and SKMAA).

7. Conclusion

This paper has presented three algorithms: Safi1, Safi2 and Said1 which are essential to estimate the channel's parameters (Prokis B) and to identify the radio mobile channels (Bran A). Furthmore this work investigated the problem of blind channel equalization by the following algorithms; Said1, the Constant Modulus Algorithm (CMA) and Sign Kurtosis Maximization Adaptive Algorithm (SKMAA). Simulation results show that the Proposed Algorithm (Said1) gives a good estimation to the parameters of channel in the noisy environment (SNR = 0dB), regarding to Safi1 and Safi2 algorithms. Concerning SKMAA algorithm performance is more higher compared to CMA algorithm, but Said1 stays the most potential algorithm in relation to other equalizers (CMA, SKMAA) for blind channel equalization.

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