iJRASET
International Journal For Research in
Applied Science and Engineering Technology

INTERNATIONAL JOURNAL FOR RESEARCH

IN APPLIED SCIENCE & ENGINEERING TECHNOLOGY

Volume: 5 **Issue:** VI **Month of publication:** June 2017

DOI:

Volume 5 Issue VI, June 2017
ISSN: 2321-9653
International Journal for Research in Applied Science & Engineering Technology (IJRASET)

Performance Analysis and Coding of Blind Adaptive Fractional – Space Constant Modulus Algorithm

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Abstract: in bandwidth-efficient digital transmission, adaptive algorithm is widely used in the digital signal processing like channel estimation, channel equalization, echo cancellation, high definition television (hdtv) set-top cable demodulators, non-invasive medical diagnosis, geophysical exploration and image enhancement and recognition and so on. One of the most important and prevalent adaptive algorithms is the lms algorithm. We present in this paper a multiple objective optimization approach to fast blind channel equalization. By investigating first the performance (mean-square error) of the standard fractionally spaced cma (constant modulus algorithm) equalizer in the presence of noise, we show that cma local minima exist near the minimum mean-square error (mmse) equalizers. Consequently, fractional spaced cma may converge to a local minimum corresponding to a poorly designed mmse receiver with considerable large mean-square error. The step size in the lms algorithm decides both the convergence speed and the residual error level, the highest speed of convergence and residual error level.

I. INTRODUCTION

Adaptive Blind equalization has the potential to improve the efficiency of communication systems by eliminating training signals Blind equalizers are used in micro-wave radio. They were realized in Very Large Scale Integration (VLSI) for various purposes. Blind processing applications are emerging in wireless communication technology. Blind methods are of great importance in digital signal communication systems as they allow channel equalization at the receiver without the use of training signals or additional bits. The topic of blind equalization of Linear Time Invariant (LTI) channels, both SIMO and MIMO, has drawn considerable attention over the past years and several algorithms have been developed. One of the practical problems in digital communications is Inter-Symbol Interference (ISI), which causes a given transmitted symbol to be distorted by other transmitted symbols. The ISI is imposed on the transmitted signal due to the band limiting effect of the practical channel and also due to the multipath effects (echo) of the channel [1]. One of the most commonly used techniques to counter channel distortion (ISI) is linear channel equalization. The equalizer is a linear filter that provides an approximate inverse of the channel response. Since it is common for the channel characteristics to be unknown or to change over time, the preferred embodiment of the equalizer is a structure that is adaptive in nature. Conventional equalization techniques employ a pre-assigned time slot (periodic for the time-varying situation) during which training signal, known in advance by the receiver, is transmitted. In the receiver, the equalizer coefficients are then changed or adapted by using some adaptive algorithm (e.g., LMS, RLS, etc.) so that the output of the equalizer closely matches the training sequence. However, inclusion of this training sequence with the transmitted information adds overhead and thus reduces the throughput of the system. Therefore, to reduce the system overhead, adaptation schemes are preferred that do not require training, i.e., blind adaptation schemes. In blind equalization (Godard, nov., 1980)[3], instead of using the training sequence, one or more properties of the transmitted signal are used to estimate the inverse of the channel. The challenge is achieving blind equalization using only a limited amount of data. A widely tested algorithm is the constant modulus algorithm (CMA). In the absence of noise, under the condition of the channel invertibility, the CMA converges globally for symbol-rate IIR equalizers and fractionally spaced FIR equalizers. It is shown in [11] that CMA is less affected by the ill conditioning of the channel. However, Ding *et. al.* showed that CMA may converge[2] to some local minimum for the symbol rate FIR equalizer. In the presence of noise, the analysis of convergence of CMA is difficult and little conclusive results are available. Another drawback of CMA is that its convergence rate may not be sufficient for fast fading channels. Another approach to the blind equalization is based on the blind channel estimation. Some of the recent eigen structure-based channel estimations require a relatively smaller data size comparing with higherorder statistical methods. However the asymptotic performance of these eigen structure-based schemes is limited by the condition of the

International Journal for Research in Applied Science & Engineering Technology (IJRASET)

channel [12, 13]. Specifically, the asymptotic normalized mean square error (ANMSE) is lower bounded by the condition number of the channel matrix. Unfortunately, frequency selective fading channels with long multipath delays.

II. CONSTANT MODULUS ALGORITHM

CMA[2],[5] has the assumptions that input to the channel is a modulated signal which has constant amplitude at every instant in time. The advantage of the blind equalization is the bandwidth is high because of there is no training of the pulses. In the conventional equalization process needs the training of the pulse. CM is used for QAM[7] signals where the amplitude of the modulated signal is not the same at every instant. The error e(n) is then determined by considering the nearest valid amplitude level of the modulated signal as the desired value[1].

Adaptive channel equalization without a training sequence is known as blind equalization. A baseband model with a channel impulse response, channel input, additive white Gaussian noise (AWGN), and equalizer input are denoted by c(n),s(n), w(n), and u(n) respectively. The data symbols transmitted s(n) , are assumed to consist of stationary independently and identically distributed (i.i.d.), real or complex non-Gaussian random variables.

The equalizer input,

$$u(n)=s(n)*c(n)+w(n) \qquad (1)$$

is then sent to a tap-delay-line blind equalizer with impulse response , intended to equalize the distortion caused by intersymbol interference (ISI) without a training signal.

The output of the blind equalizer

$$\begin{aligned} y(n) &= u(n)*f(n) \\ &= s(n)*h(n)+w(n)*f(n) \\ &= \textstyle\sum_i h(i)s(n-i) + \sum_i f(i)w(n-i) \qquad (2) \end{aligned}$$

It can be used to recover the data transmitted symbol s(n)

where h(n)=c(n)*f(n)

A. Cost Function Of Cma

The cost function of the constant modulus algorithm is[6]

$$JCMA\,(n)=E\{[\,|y(n)|^2 - R2\,]^2\} \qquad (3)$$

where

$$R2 = E\{|s(n)|^4\} / E\{|s(n)|^2\}$$

Depending on the cost function only the blind equalization was determined [2].

III. FRACTIONAL SPACED CMA - CONSTANT MODULUS ALGORITHM

In digital communication, equalizer was designed to compensate the channel distortions, through a process known as equalization. There are two types of equalization which are: 1) Trained equalization, 2) Blind (self-recovering) Equalization. Blind equalization finds important application in data communication systems. In data communications, digital signals are generated and transmitted by the sender through an analog channel to the receiver. Linear channel distortion as a result off limited channel bandwidth, multipath and fading is often the most serious distortion in digital communication system. Blind equalization improves system bandwidth efficient by avoiding the use of training sequence. The linear channel distortion, known as the Inter-symbol interference (ISI), can severely corrupt the transmitted signal and make it difficult for the receiver to directly recover the transmitted data. Channel equalization and identification has proven to be an effective means to compensate the linear distortion by removing much of the ISI.

A. Fractional–Spaced Cma

Fractional –spaced CMA[6] used to directly estimate equalizer *f* . It is similar to CMA. In fractional space it is global convergences.

$$\min J = E[(|f^H X(n)|^2 - R_z)^2] \qquad (4)$$

Update rule:

$$f_{n+1} = f_n - \mu E(|f^H X(n)|^2 - R_z)X(n)X^H(n)f_n \qquad (5)$$

1) Algorithm:

Volume 5 Issue VI, June 2017
ISSN: 2321-9653
International Journal for Research in Applied Science & Engineering Technology (IJRASET)

a) Construct the received sample space.

b) Construct the sample vector *X* (*n*) .

c) For n=1,2,…….

Update function (5)

Calculate instant error

d) Check SER.

B. Coding For Fractional –Spaced Cma

```
% Blind channel estimation/equalization
% adpative CMA method in Fractional space
%
%
T=1000;    % total number of data
dB=25;     % SNR in dB value

%%%%%%%%% Simulate the Received noisy Signal  %%%%%%%%%%%
N=5; % smoothing length N+1
Lh=5;  % channel length = Lh+1
Ap=4;  % number of subchannels or receive antennas
h=randn(Ap,Lh+1)+sqrt(-1)*randn(Ap,Lh+1);   % channel (complex)
for i=1:Ap, h(i,:)=h(i,:)/norm(h(i,:));    end       % normalize
s=round(rand(1,T))*2-1;  % QPSK or 4 QAM symbol sequence
s=s+sqrt(-1)*(round(rand(1,T))*2-1);
% generate received noisy signal
x=zeros(Ap,T);    % matrix to store samples from Ap antennas
SNR=zeros(1,Ap);
for i=1:Ap
   x(i,:)=filter(h(i,:),1,s);
   vn=randn(1,T)+sqrt(-1)*randn(1,T);   % AWGN noise (complex)
   vn=vn/norm(vn)*10^(-dB/20)*norm(x(i,:));  % adjust noise power
   SNR(i)=20*log10(norm(x(i,:))/norm(vn));   % Check SNR of the received samples
   x(i,:)=x(i,:)+vn;                        % received signal
end
SNR=SNR    % display and check SNR
%%%%%%%%%%%%%% adaptive equalizer estimation via CMA
Lp=T-N;   %% remove several first samples to avoid 0 or negative subscript
X=zeros((N+1)*Ap,Lp);  % sample vectors (each column is a sample vector)
for i=1:Lp
   for j=1:Ap
      X((j-1)*(N+1)+1:j*(N+1),i)=x(j, i+N:-1:i).';
   end
end
e=zeros(1,Lp);  % used to save instant error
f=zeros((N+1)*Ap,1); f(N*Ap/2)=1;    % initial condition
R2=2;                  % constant modulas of QPSK symbols
mu=0.001;      % parameter to adjust convergence and steady error
for i=1:Lp
  e(i)=abs(f'*X(:,i))^2-R2;               % instant error
  f=f-mu*2*e(i)*X(:,i)*X(:,i)'*f;     % update equalizer
```

Volume 5 Issue VI, June 2017
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```
  f(N*Ap/2)=1;
%   i_e=[i/10000 abs(e(i))]           % output information
end
  sb=f'*X;   % estimate symbols (perform equalization)
% calculate SER
H=zeros((N+1)*Ap,N+Lh+1);  temp=0;
for j=1:Ap
   for i=1:N+1, temp=temp+1; H(temp,i:i+Lh)=h(j,:); end  % channel matrix
end
fh=f'*H;    % composite channel+equalizer response should be delta-like
temp=find(abs(fh)==max(abs(fh)));   % find the max of the composite response
sb1=sb/(fh(temp));  % scale the output
sb1=sign(real(sb1))+sqrt(-1)*sign(imag(sb1));  % perform symbol detection
start=N+1-temp;  % general expression for the beginning matching point
sb2=sb1(10:length(sb1))-s(start+10:start+length(sb1));  % find error symbols
SER=length(find(sb2~=0))/length(sb2)   % calculate SER
if 1
   subplot(221),
   plot(s,'o');   % show the pattern of transmitted symbols
   grid,title('Transmitted symbols');  xlabel('Real'),ylabel('Image')
   axis([-2 2 -2 2])
      subplot(222),
   plot(x,'o');  % show the pattern of received samples
   grid, title('Received samples');  xlabel('Real'), ylabel('Image')
      subplot(223),
   plot(sb,'o');   % show the pattern of the equalized symbols
   grid, title('Equalized symbols'), xlabel('Real'), ylabel('Image')
   subplot(224),
   plot(abs(e));   % show the convergence
   grid, title('Convergence'), xlabel('n'), ylabel('Error e(n)')
end
```

IV. SIMULATION RESULTS

The results of FS-CMS have been simulated and verified. FS-CMS gives improved performance in noisy environment. Fig.1 depicts the transmitted symbols, Fig.2 shows the received symbols without equalization. Fig.3 shows the symbols received after equalization. This is explicitly the improved performance of FS-CMS under noisy conditions. Fig.4 depicts the convergence rate of FS-CMS

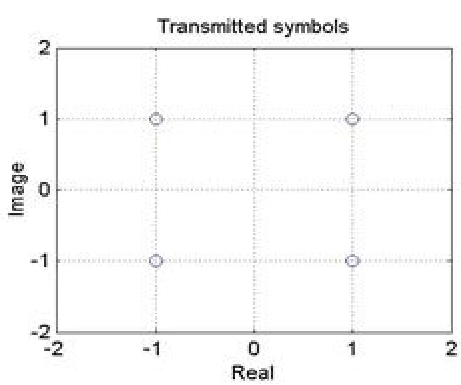

Fig1. Transmitted signal

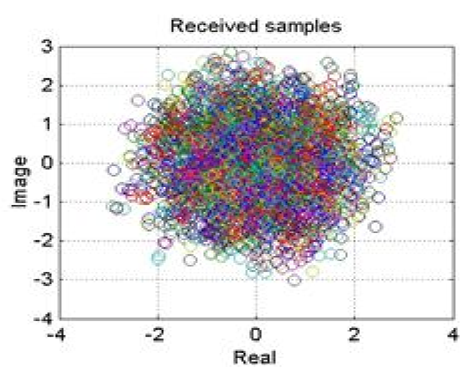

Fig2. Received signal

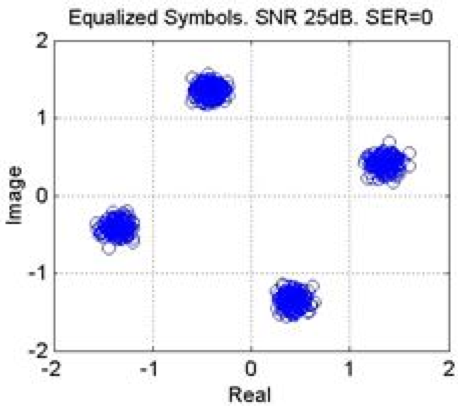

Fig3. Symbols after FS-CMA Equalization

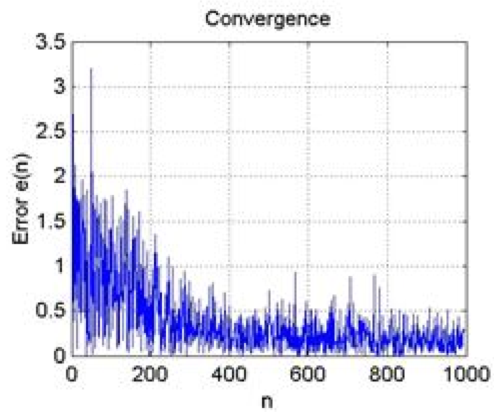

Fig.4. Convergence by FS-CMA

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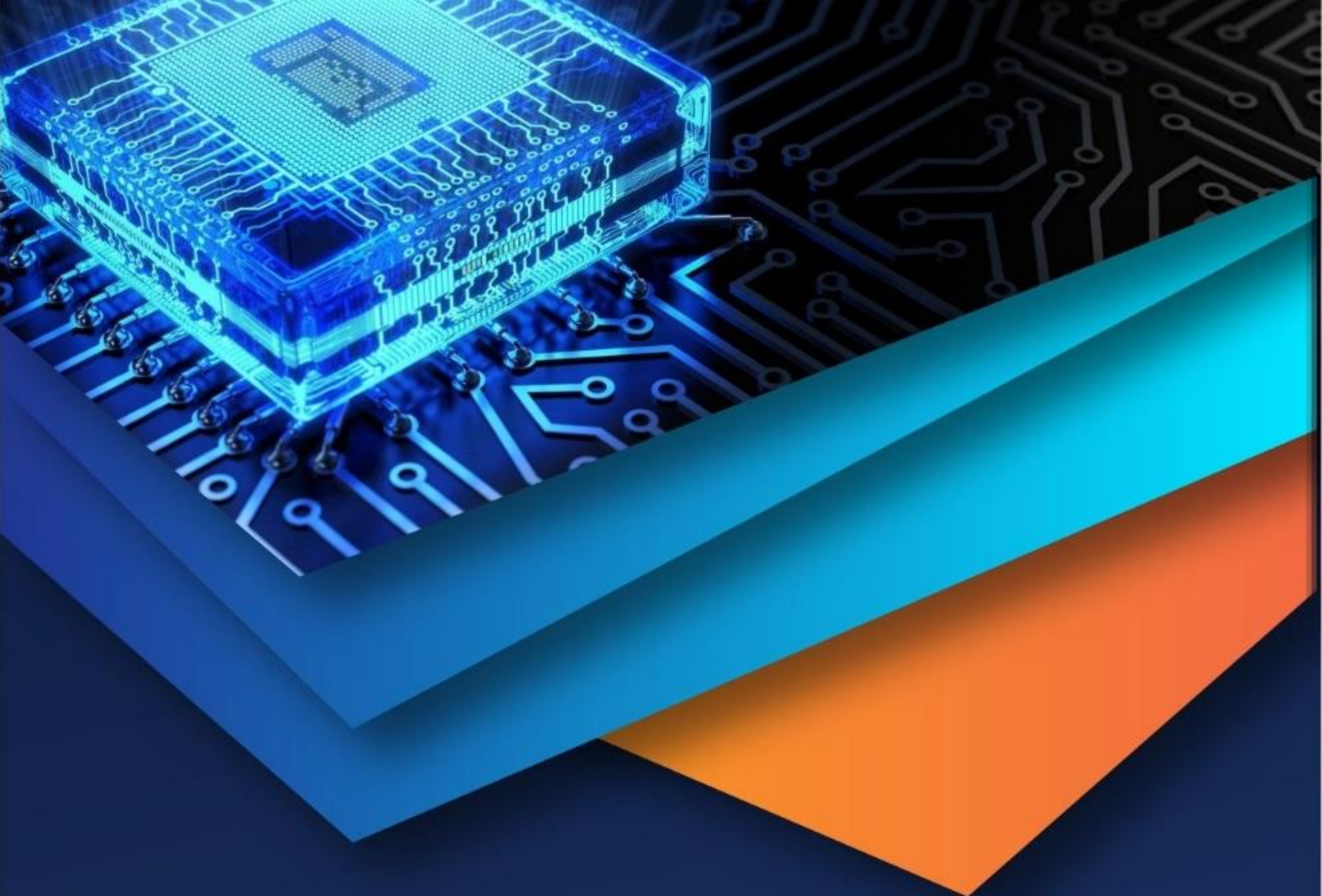

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