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INVESTIGATION OF PHASE NOISE ON THE PERFORMANCE OF LMS-RLS ADAPTIVE EQUALIZER

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ABSTRACT:- This paper investigates the effect of phase noise on equalization of communication channels using least mean square (LMS) and recursive least square (RLS) adaptive algorithms. The aim of the investigation is to mitigate inter-symbol interference (ISI) caused by the channel and to impose the bit error rate (BER) in the received signals.

The equalizerusestwobasicadaptivealgorithms: LMS algorithmand RLS algorithm. Without LMS-RLS equalizer,theBER ismorethan when the system modelincludesLMS-RLS equalizer as indicated in table (1) and table (2). Equalizer algorithm is analyzed using MATLAB v.9 Communication Block Set.

1. INTRODUCTION

The channel used to transmit the data distorts signals in both amplitude and phase causingISI. Other factors like terminal noise, impulse noise, cross talkand the nature of channel itself cause farther distortion to the received symbols. The recent digital transmission systems imposethe application of channel equalizers withlowcomplexity and low BER.

Adaptiveequalizers are unavoidable to satisfy these requirements. A channel equalizer isan important component of communication system and is used to mitigate the ISIintroduced by the channel.The equalizer depends upon the channel characteristics.Inwireless channel due the multi path fading, the channel characteristics change with time. Thus, it may be necessary for channel equalizer to track the time varying channel in orderto provide reasonable performance[1]. An adaptive equalizer isessentially a linearadaptive filter used to model the inverse transfer function of the channel. Two well-known adaptive algorithms are the LMS and RLS algorithms. Although the RLS algorithm has betterconvergencespeed thanthe

LMSalgorithm, it's complexity for hardwareimplementation can be very high. Actually, the LMS algorithm is widely adoptedin hardware implementation because of it's simplicity robustness [2]. The LMS algorithm executes quickly but converges slowly, and it's complexity grows linearly with the number ofweights. The RLS algorithmconverges quickly, but it's complexity grows with thesquare of the number of weights. The term equalization can be used to describe anysignal processing operation that minimizeISI. An equalizer within a receiver compensatesfor average range ofexpected channel amplitude and delay characteristics, thus, anequalizer attempts to mitigate ISI and improve the receiver's performance[2]. Equalizermust track the time varying and unknown properties of channel thus are called "adaptive equalizer" [3].The effect of phase noise in the equalized system is investigated in this paper to assess the robustness of the algorithms.

2.LMS AND RLS ALGORITHMS

2.1 LMS Algorithms

The LMS algorithm shown in Figure (1), consists of two basics procedures;*(i)* filter processwhich involves computing the output of linear filter response to the input signal and generating and estimation error by comparing this output with a desired response[4], and*(ii)* a adaptive process, which involvesthe automatics adjustment of the parameter of filteraccordance with the estimate error. The step-size parameter(μ) is critical to the performanceofLMS and determines how fast the algorithm convergence along the error performancedefined bythefunction $(1/2)e^2(n)$. As well-known, if the step-size is large the convergence rate of the LMS algorithm will be rapid if the step-size is small the convergence rate of theLMS algorithm will beslow. Many variable step-size LMS algorithms have been robust that improve on the performance of the LMS algorithm by using large step-size at the earlystages of the adaptive process and smallstep-size . High level measurement noise or input data tends to deteriorate the convergence performance and in order to overcome this problem, the step-size [6] iscontrolled by the squared autocorrelation of errors at adjacent interval. The weight vector equation:

$W(n+1) = W(n)+1/2\mu[-\nabla(E\{e^2(n)\})]$ ....(1)

where μ = step-size parameter, and$e^2(n)$ = MSE between output x(n), and gradient vector in the aboveweight.

$\nabla_w(E\{e^2\})= -2L+2 R*W(n)$…………………….(2)

where L and R matrices in real time.

2.2 RLS Algorithm

The RLS filter overcomes some practical limitation of the LMS filter by providing faster rate ofconvergenceand good performance. A rateof convergence is typically an order of magnitude faster than LMS filter. Due to the factthat RLS filter whites the input data by using the inverse correlation matrix of the data. The improvement is achievedat the experience of an increase in computational complexity of RLS filter [7]. The RLS algorithm has the same two procedures as LMSalgorithm, accept that it provides a tracking rate for fast channel, and RLS algorithm is known tohave the stability issues due to the covariance update formula p(n)[8], which is used for automaticsadjustment in accordance with estimation error[9]. The block diagram is the same of Figure (1).

3.SYSTEM DESCRIPTION AND SIMULATION RESULTS

The model employed in this work consists of random integer generator, QAM modulator, additive white Gaussian noise (AWGN)channel, phase noise, LMS equalizer, RLS equalizer, and QAM demodulator, is implemented using MATLAB block set as shown in Figure(2). Simulation is carried out by varying signal to noise ratio and component gain of channel for the RLS and LMS algorithms individually and both. The output is observed in the form of BER, number of errors and the number of bits processed. The equalizer block provides option for the selection of the RLS and LMS algorithms for simulation. The phase noise block applies noise to a complex, base band signal as follows: *(i)* generate AWGN and filters it with a digital filter*(ii)* Add the resulting noise to the angle component of the input signal. Thephase noise block generates phase noise over the entire spectral window, from (0Hz to $\pm$fs/2), where fsrepresentthesampling frequency. Theblock generates a phase noise with 1/f characteristics over the entire frequency range. The LMS linear equalizer uses a linear equalizer and LMS algorithm to equalize a linearly modulated base band signal through a dispersive channel. During the simulation the block uses the LMS algorithm to update the weights, once per symbol if the number ofsamples per symbol parameter is (1).Then the block implements a symbol-spaced equalizer, otherwise, the block implements a fractionally spaced equalizer. The errorrate calculation block compares input data from a transmitter with input data from a receiver. It calculate the error rate as a running statistics, by dividing the total number of unequal pairs of data elements by the total number of input data element from one source[4]. When the equalizer system operates with frequency noise having power spectral density from (0 dBc/Hz to -200 dBc/Hz) and frequency offset($\Delta f = 0$Hz), the

BER is found to be $8.991*10^{-5}$ when SNR = 12dB. If the equalizer system operates with frequency noise having power spectral density from (0 dBc/Hz to -100 dBc/Hz) and ($\Delta f > 0$Hz), the BER will have different values as shown in Table (1). In contrast, without LMS equalizer, the BERis equal to (0.4561) when the phase noise (Φ) varies from (0dBc/Hz to -100dBc/Hz) and ($\Delta f = 0$Hz) when SNR =12dB. For ($\Delta f > 0$Hz) and phase noise (Φ) varies from (0dBc/Hzto -100dBc/Hz), the BERincreases dramatically as shown in Table (2).

4. CONCLUSIONS

In this study the performance for communications system using channel equalizer to mitigate ISI and low BER by using LMS and RLS algorithms has been investigate. The results shown the effect of phase noise and frequency offset on the magnitude of BER. In the absence LMS equalizer the BER will be larger than when the system model includes LMS-RLS equalizer.

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Table(1) : BER versus phase noise and frequency offset with LMS-RLSequalizer.

Φ (dBc/Hz)	BER			
	Δf = 0 Hz	Δf = 200 Hz	Δf = 500 Hz	Δf = 1000 Hz
0	0.00008991	0.4995	0.4998	0.5004
-20	0.00008991	0.5035	0.5012	0.5016
-40	0.00008991	0.008462	0.07136	0.1886
-60	0.00008991	0.0000999	0.0001199	0.0001499
-100	0.00008991	0.00008991	0.00007992	0.00007992
-200	0.00008991	0.00008991	0.00008991	0.00008991

Table(2): BER versus phase noise and frequency offset without LMS-RLSequalizer.

Φ (dBc/Hz)	BER			
	Δf = 0 Hz	Δf = 200 Hz	Δf = 500 Hz	Δf = 1000 Hz
0	0.4561	0.4974	0.4988	0.497
-20	0.4561	0.502	0.4999	0.5004
-40	0.4561	0.4461	0.4515	0.4879
-60	0.4561	0.456	0.456	0.4557
-100	0.4561	0.4561	0.4561	0.4561
-200	0.4561	0.4561	0.4561	0.4561

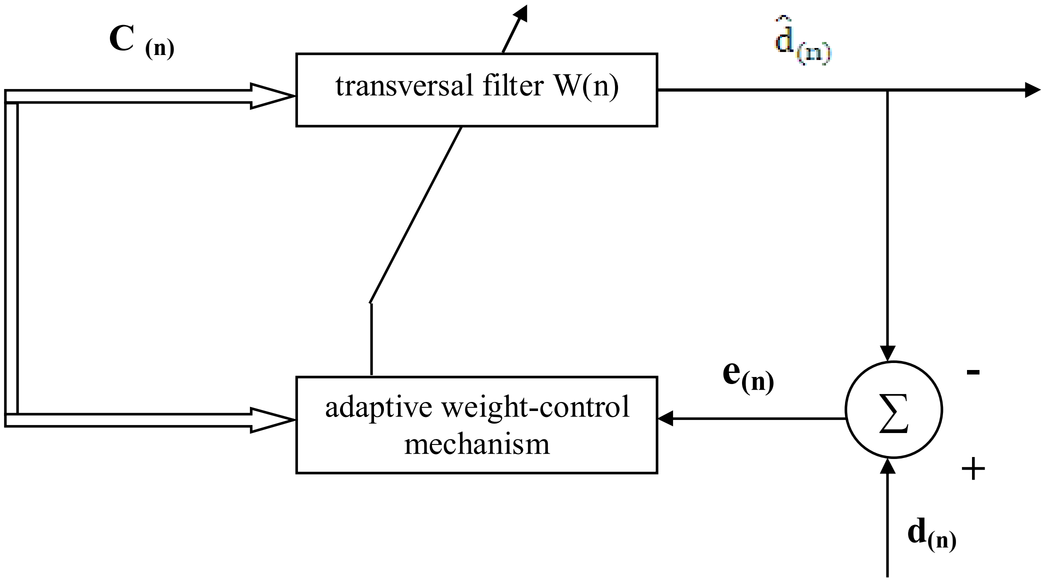

Fig.(1) : Block diagram of adaptive transversal filter employing LMS algorithm.

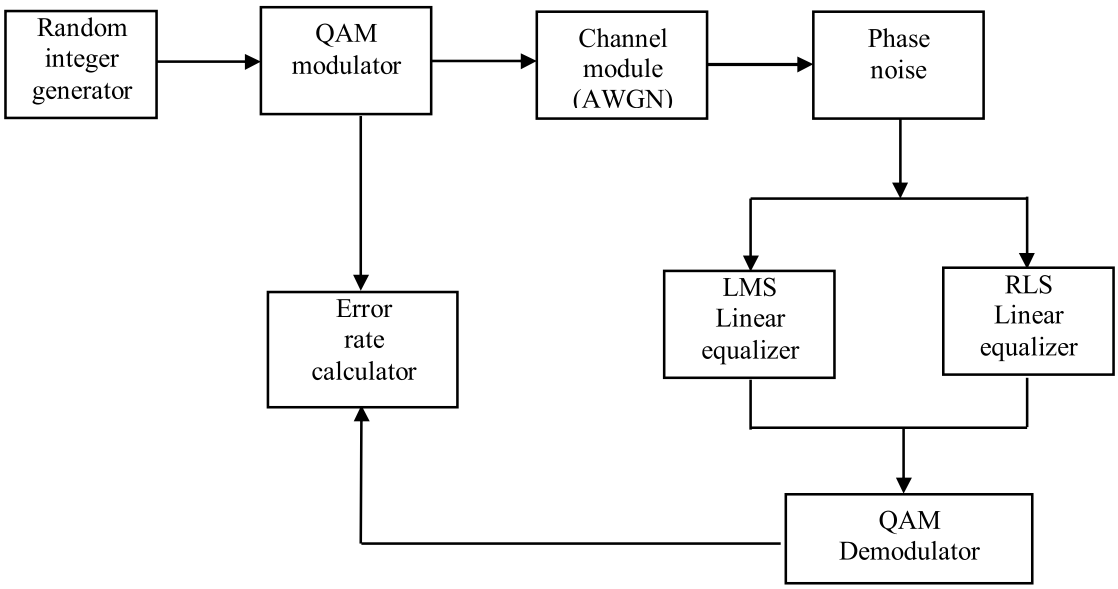

Fig. (2) : block diagram,Stucture of system module.

Figures(3) and (4) show the variation of BER with Δf when SNR=12 dB andphase noise = -30dBc/Hz and -40 dBc/Hz, respectively.The results indicate that BER increases with Δf.

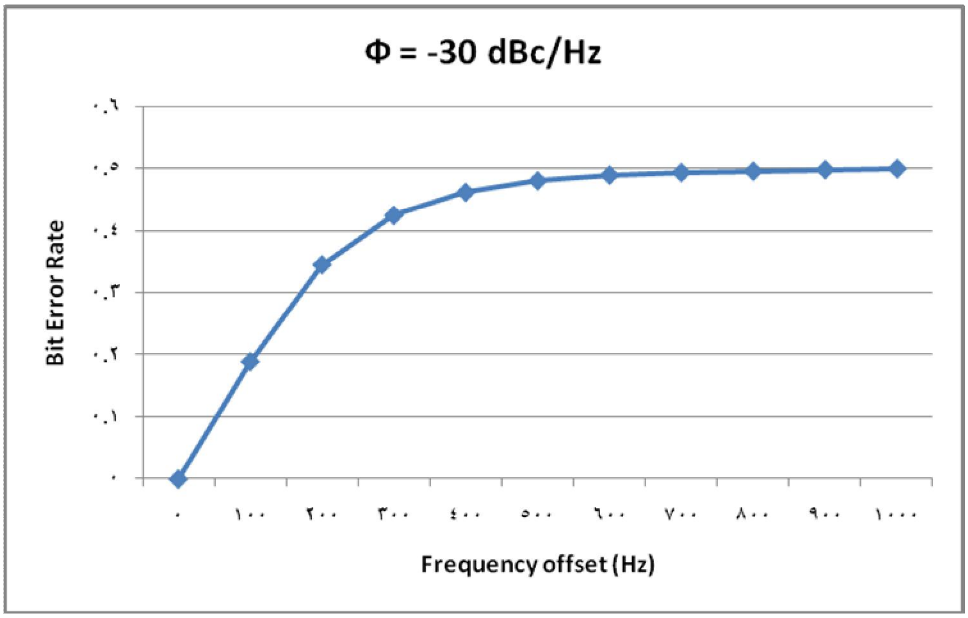

Fig.(3): BER versus frequency offsetwhen SNR = 12dB and Φ = -30 dBc/Hz.

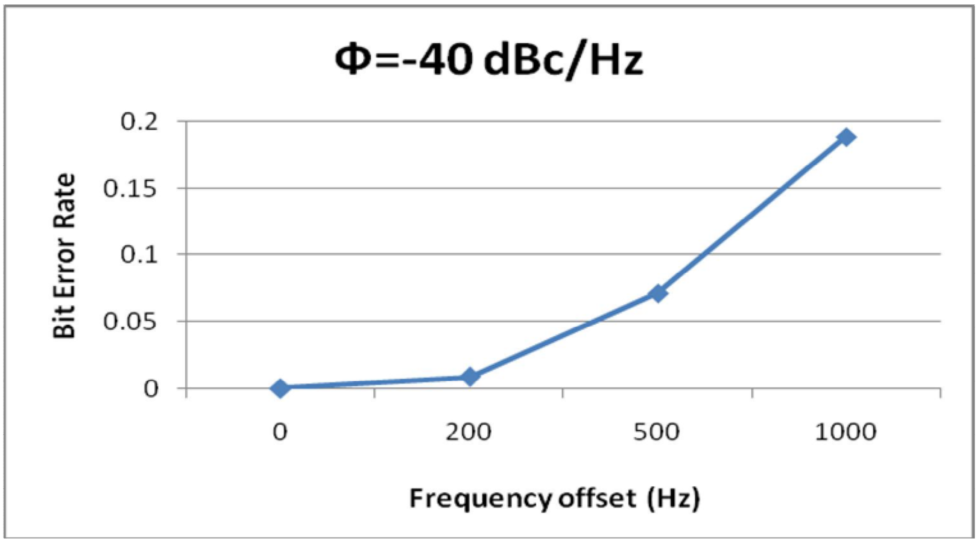

Fig. (4): BER versus frequency offset when SNR = 12dBand Φ = -40 dBc/Hz.

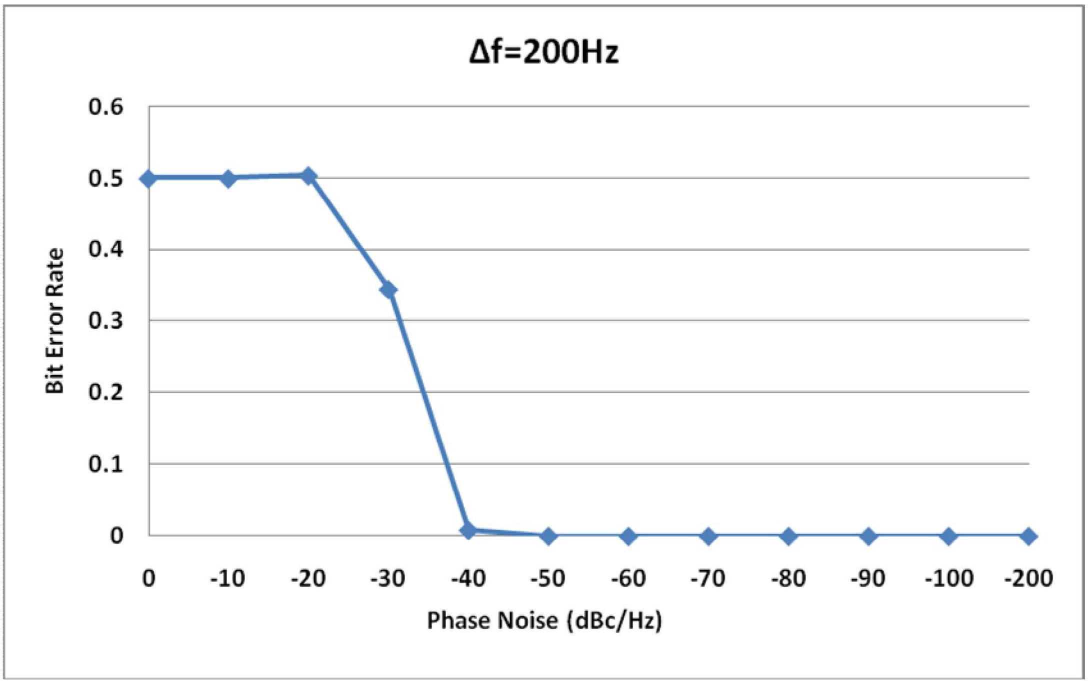

Fig.(5): BER versus phase noise when SNR = 12dB and Δf =200Hz.

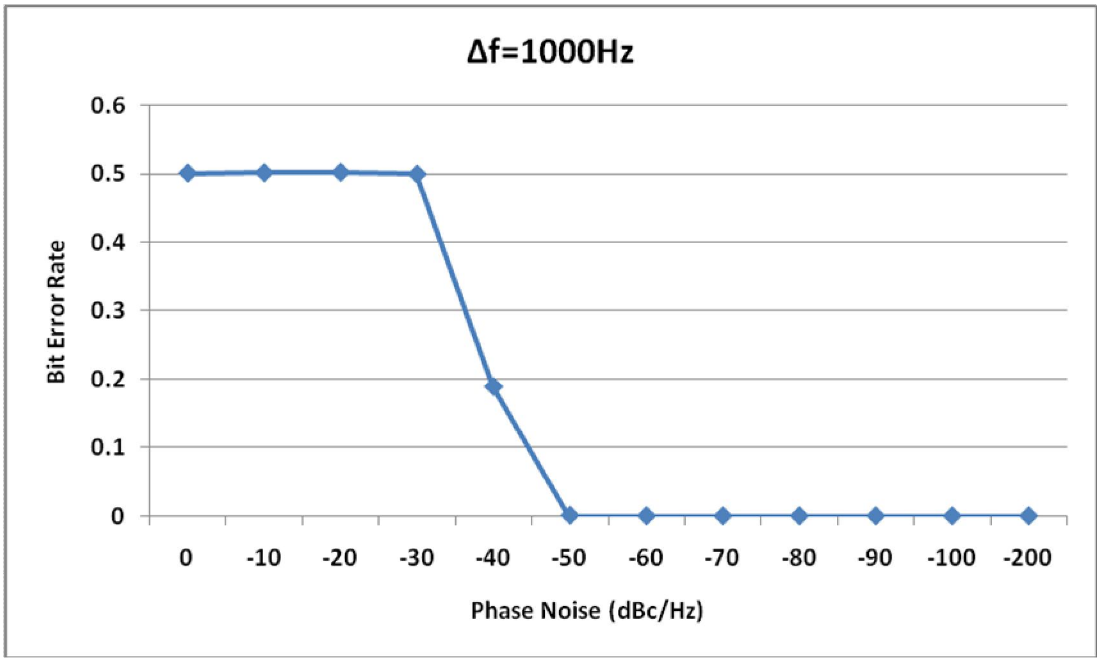

Fig. (6): BER versus phase noise when SNR = 12dB and Δf =1000 Hz.

تحليل تأثير الضوضاء على مسارات قنوات الاتصال التي تستعمل خوارزميات (LMS-RLS)

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الخلاصة

ان البحث يتحرى تحليل تأثير الضوضاء على مسارات قنوات الاتصال التي تستعمل خوارزميات تكيفية، اقل مربع متوسط (LMS) واقل مربع تكراري (RLS) ، ان هدف الاستقصاء هو تسكين او تقليل التداخل الحاصل بين الاشارات المستلمة بسبب تاثير قنوات الاتصال ويستعمل المعادل نوعان من الخوارزميات الاول (LMS) والثاني (RLS) ، في حالة عدم استعمال المعادل تكون نسبة الخطأ (BER) اكبر مما هو عليه في حالة عدم وجود المعادل كما مبين في الجدول رقم (١) و الجدول رقم (٢)، تم استعمال برنامج (MATLAB v.9) لتحليل خوارزميات المعادل.