opción

89

Opción, Año 35,No.89 (2019): 2337-2365 ISSN
1012-1587/ISSNe: 2477-9385

Optimum Design and Implementation of Adaptive Channel Equalization using HDL Coder

Mohammed Hussein Ali
Al-Turath University College
Noora H. Sherif
Al-Turath University College
noora.hani.na
Ghufran Saady Abd-almuhsen
Madenat Alelem University College

Abstrac

The LMS algorithm is the most widely used in wireless applications such as equalization and adaptive filtering in imaging, communications, etc. The initialization of the LMS equalizer results from a pilot based channel estimation. This technique is achieved by using algorithms which adapt the parameters of Finite Impulse Response (FIR) equalizer to decrease the clutter and Inter symbol Interference (ISI) to give an accurate received signal. This paper presents the implementation of the LMS algorithm on Field Programmable Gate Arrays (F P G A). Construction of adaptive equalizer that is built on multiplier and adder is to achieve the Multiple Access Channel (MAC) process of the Finite Impulse Response (FIR). The parameters of the adaptive equalizer are modified off-hand by the LMS method based on insertion information's. The proposed construction of this method designed using MatlabSimulink (R2015a) to treat with parallel constructer. The designed converted to VHDL coding pattern, as will as a VHDL test bench using Simulink HDL Coder tool to realize hardware directly from Simulink design.

KeyWords: Adaptive Equalizer, Least Mean Square (LMS) algorithm, field programmable gate array (FPGA), Finite Impulse Response (FIR) filter.

Diseño e implementación óptimos de ecualización adaptativa de canales utilizando HDL Coder

Resumen

El algoritmo LMS es el más utilizado en aplicaciones inalámbricas como ecualización y filtrado adaptativo en imágenes, comunicaciones, etc. La inicialización del ecualizador LMS resulta de una estimación de canal basada en piloto. Esta técnica se logra mediante el uso de algoritmos que adaptan los parámetros del ecualizador de respuesta de impulso finito (FIR) para disminuir el desorden y la interferencia entre símbolos (ISI) para dar una señal recibida precisa. Este artículo presenta la implementación del algoritmo LMS en matrices de puertas programables de campo (F P G A). La construcción del ecualizador adaptativo que se basa en el multiplicador y el sumador es lograr el proceso de Canal de acceso múltiple (MAC) de la Respuesta de impulso finito (FIR). Los parámetros del ecualizador adaptativo se modifican por el método LMS basado en la información de inserción. La construcción propuesta de este método diseñado usando MatlabSimulink (R2015a) para tratar con constructor paralelo. El diseño se convirtió al patrón de codificación VHDL, al igual que un banco de pruebas VHDL utilizando la herramienta Simulink HDL Coder para realizar hardware directamente desde el diseño Simulink.
Palabras clave: ecualizador adaptativo, algoritmo de mínimo cuadrado medio (LMS), matriz de compuerta programable de campo (FPGA), filtro de respuesta de impulso finito (FIR).

1. Introduction

Several-track effect founds in wireless transfer data connection. Signal through various channels of different delays, received at the one time, leading to Interfacing between symbols. Regrettably most of the time. The digital transfer of information is attached by A phenomenon defined as Inter symbol Interference (ISI).

Transfer information's carrier and peripheral positions (buildings, etc.), produce them spread environmental variations with time. That is, ISI changes reasoned by many-track influences and time, and ISI increases at the data transmitted during the channel is dispersive, in which each pulse that is received is affected by somewhat by neighbor pulses and depend on which interference happens in the transmitted signals. The equalizer is composed of an equalization model and adaptive algorithms. Adaptive algorithms is a technique used to resolve overlapping symbols. Equalization adjustment is used to make the channel attend to an unknown time varying channel, so it requires a special kind of algorithm to upgrade the filter parameters for channel

tracking the alterations. Parameters such as: symbol period, propagation delay across multiple paths, Doppler propagation, time of coherence / bandwidth, variable time or fixed property, channel gain etc. Play a pivotal role in the execute a wireless transmissions as summarized in [1].
The paper includes the optimum design and implementation of adaptive filter that employing LMS algorithm.. The target devices are designed and implemented on FPGA. FPGA plays an important function and supply several benefits like prototyping, speed of transfer data, adaptive, flexibility of structure, rapid execution of calculations and low cost. a group of pipelined devoted processors, is designed as a system of linear variation with integer variables. We demonstrate, how the code-size efficient synthesis on FPGAs can be solved with well-known mathematical programming.

To perform equalization there are various adaptive algorithms can be used[2]:

Least mean square, (LMS)

Signed LMS, including sign LMS, signed regressor LMS, and sign - sign LMS

Normalized LMS

Variable-step - size LMS

Recursive least squares (RLS)

Constant modulus algorithm (CMA)

2. LMS adaptive equalization algorithm

Equalization process is a removal process of ISI and clutter influences from the information's, and is the existed at the end of receiver. The major obstacles to realizing an enhancement digital transfer rates with desired accuracy is intersymbol interference. ISI trouble can be solved using channel equalization [3].The target is equalizer that is designed to make the impulse response of the channel and equalizer are gathered is as close to -d^(-γ) (delay symbol)may be done. Figure 1 demonstrate a block diagram of such a channel. Often the channel coefficients are not defined beforehand and more differ over time, which used in implementations significantly. And then, it is It is very important to employ a equalizer that is adaptive, which proceed means to track path features.

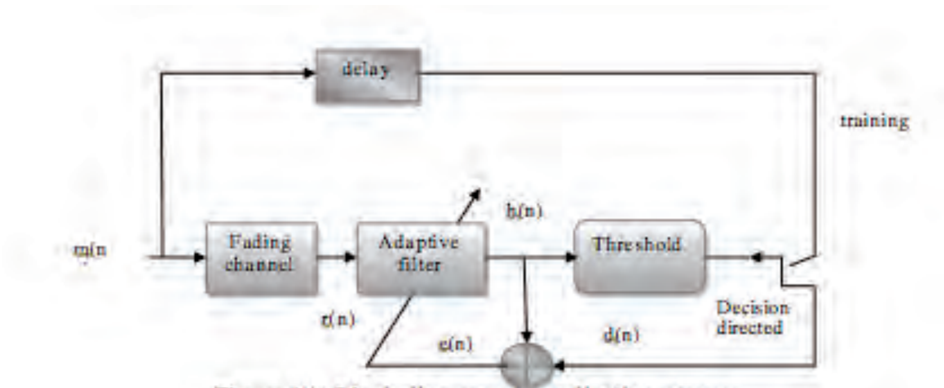


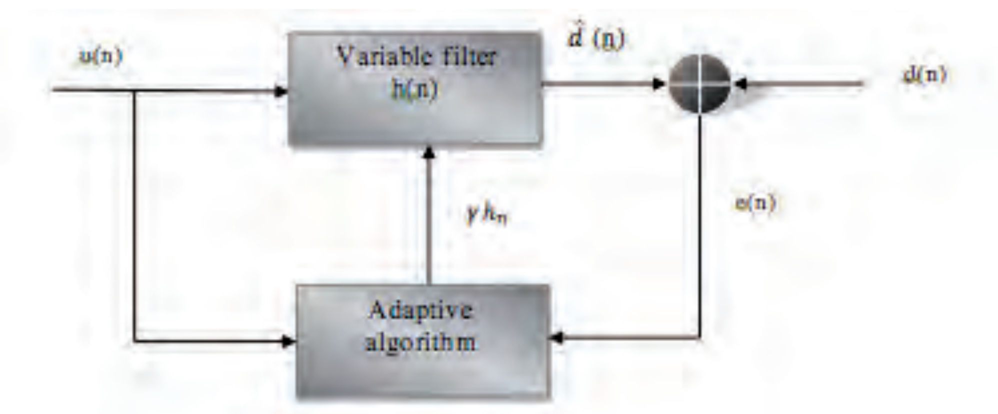
Figure (1): Block diagram of equalization process.

knowledge and skills in a way that changes their ideas and behaviors in academic work. In other words, development is now one of the most important systems that must be available in schools. Dealing with and implementing them within schools (Abdel Aziz, Safaa and Salama Abdel Azim, 2007, 336)
As the teacher is the key to the educational process and the pioneer of the society on which he relies on the upbringing of his sons, the strong and correct birth, the question of preparation and qualification has occupied the minds of educators and specialists. The teacher is not only a communicator of knowledge, but also educates the generations in terms of mental, moral and physical education. To reality by way of correct behavior (Zair et al., 2011: 57)

Practical education is a pivotal center in teacher education programs. It is regarded as the correct practical method and the proper method that enhances the student's ability to apply knowledge, concepts and theoretical principles. Among the important objectives that practical education seeks to achieve, which positively influence the preparation and training of students - The apprentice is provided with good professional skills and the teaching skills that qualify him to be an effective teacher with his students in the classroom.
And the responsibility of preparing the students applied in the faculties of education requires follow-up not only in theoretical and practical lectures, but beyond to follow them during the period of application and supervision by the supervisors supervisors and school administrations and teachers cooperating in the schools of application. This calendar and follow-up by the previous three categories Is an essential element of the educational and training process received by the student applied in the period of preparation within the college to verify the extent of practice and employment of the principles and theoretical concepts of the scientific

and educational decisions that he received during his years of study.
The importance of research is thus determined by the importance of:
1. Applicant in the Faculty of Education as a teacher of future generations.
2. Evaluation of the program of practical education in general and the importance of evaluating the student's performance applied in the teaching of mathematics in particular.

3. The views and opinions of the academic supervisor, the collaborating teacher and the school principal in evaluating the performance skills as they are the most individuals in contact with the students applied during the application period.
4. Performance skills (teaching skills) as the operational framework for each theoretical study received by the student applied in the preparation period.
5. Encourage school administrations and cooperating teachers to participate effectively in evaluating the practical education program in general and the performan



The estimation of required signal is done by convolved the input message (data) with the impulse response.
The estimation signal can be demonstrate as
d (n) = h(n) * r(n)……………………………………..(4)
And input symbols is
R(n) = [r(n), r(n-1),r(n-2),……,r(n-k)]T…………………..(5)
R(n) denotes the input data vector and ·T denotes transpose of the data vector..
For each instant the coefficients of variable filter will be modified
hn+1= hn + γh_n ……………………………………(6)
Where γh_n is a correction coefficient that generated by adaptive algorithm.

Algorithm that is adaptive uses h_n to upgrades the coefficients of the equalizer.

Therefore, LMS techniques looking for the optimum design of equalizer weights (based on the Minimum Mean square Error MMSE criterion). This operation several times are repeated fastly, so the equalizer tries to converge, and much than one method (like gradient or steepest decent algorithms) must be used in ordered to decreased the incorrect results . The adaptive equalizer block up the filter weights so that the results(error signal) extent to acceptable value or in the state of training sequence are to be sent [4].The operation modes of adaptive equalizer are training mode and tracking mode.
The technique of LMS method is consider as a type of adaptive equalizer utilized to simulate a filter which is desired, by calculate the parameters of filter that contact to creating the LMS of the error (result from the difference between the desired and the actual signal). It is type of a gradient descent procedure, which is that equalizer only upgrade based on difference between the desired and the actual signal at the current time. Tapped delay line is used to build LMS filter structure which is responsible for performing the filtering method which comprise ,calculating the output d (n- d) of a linear prediction equalizer in response to the data input and producing an estimation error by taking the difference between the output with a desired response as shown: [5].
$e(n) = d(n) - y(n)$……………………….(7)
y (n) is output of filter and demand response at time n.
while the second process of LMS algorithm is adaptive process as mentioned previously which includes the modification automatics the coefficient of the filter based on the prediction error.
$h(n+1) = h(n) + \Delta u(n)*e(n)$ ………….(8)
And Δ is the step size, and the predicate tape weight vector is (n+1) at time (n+1).So the adaptive filter require (n+1) multiplications and need |(n+1) additions to upgrade the filter weights.
When these both process(filtering and adaptive processes) are working with each other produce the so called feed back loop.

3. Simulink Model of Adaptive Equalizer

Design of Adaptive equalizer can be done using the communications blocks Toolbox which contents Simulink blocks, System objects, and that provide equalization capabilities can be classified as Adaptive or Maximum-Likelihood Sequence Estimation

Figure 3 shows the SIMULINK model of adaptive equalizer

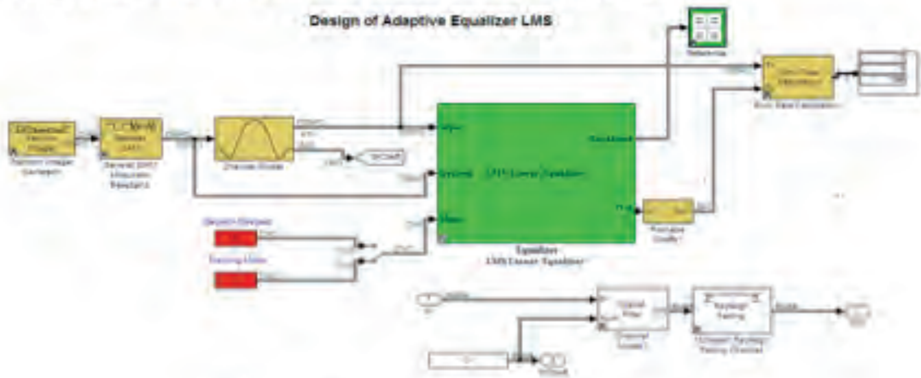


Figure 3: Simulink Model Adaptive Equalizer

Figure 3: Simulink Model Adaptive Equalizer

Structure of Simulink model comprises of block of generate random integer, MQAM modulator, symbol converter, , and a raised cosine shaping filter at the end of transmitting side, The type of channel that used is Rayleigh fading., adaptive LMS algorithm module, block for converting the integer value to symbols, and the end of model connected an error calculator, all the blocks designed using MATLAB communications tool. A stream of bits was transmitted by the generator of random integer block r[n] to simulate a message. The r(n) stream of bits was transform into integer data from 0 to M-1 and this operation is done first, then the modulator carry out its function.

By using delay tapes the required response can be obtained after a Appropriate delay which was execute to the adaptive LMS algorithm in the form of a training series Which plays a role in running filter coefficients by Weighting equations of the upgrade LMS algorithm.

The data bits stream set from the equalizer then mapped return from convertor integer to bits and vice versa, and when reach this step can be calculated error rate of the all emulation, an error calculator block was jointed between the transmitter block and the receiver block which compares input data that transmitting with output data from the equalized.

4.Result of system model

The results(error rate) of this model come from divide random pairs of symbols data by the sequence of data input generated from the integer random block, and this result demonstrated in Figure 4.

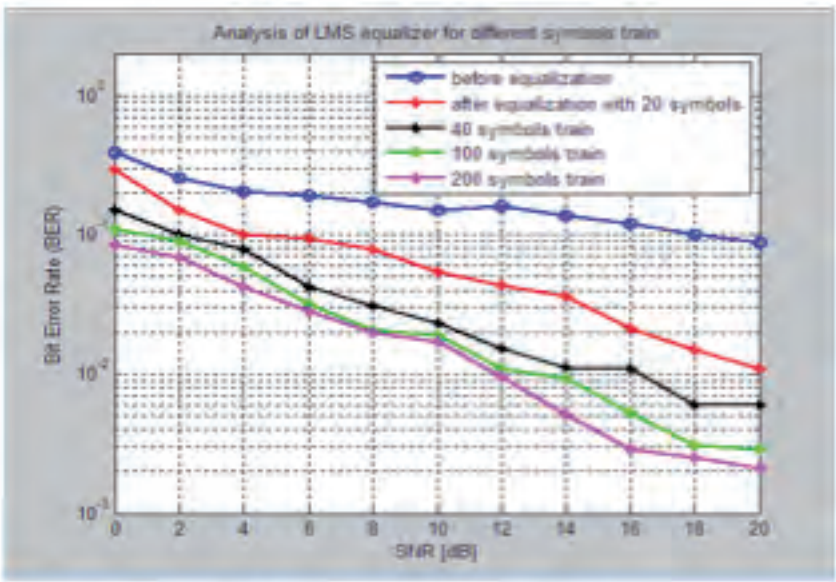


Figure 4: BER for Adaptive LMS Algorithm.

The simulation results shown in Figure 4 demonstrate that performance of adaptive LMS algorithm. Simulated is done in terms of BER and number of errors as a function of differ amount of SNR (dB) with 16 QAM signaling platform, each frame include 2000 symbols(each symbol content four bits) generated by integer block message, over Rayleigh fading channel with doppler shift(fd) equal to = 50 Hz and the other parameters used for algorithm simulations are step size = 0.01, number of filter taps = 6, reference taps= 4, leakage factor = 1. When compare the results that obtained it is clear that performance of the system became better with used adaptive equalizer, and BER for LMS equalizer was improved with the training length is increased from 20 to 100 symbols generated by sequence blocks. Furthermore, it is clear that when training length increased above 100 symbols give rise to reduce efficiency of band width due to high latency, so in this case the BER will be slight improved. Therefore, the distortion that result from ISI effects will be decreased with Adaptive LMS equalizer and limit training sequence.

5. FPGA Implementation results of Adaptive LMS Equalizer

To implement LMS equalizer on FPGA can use HDL Coder which is high level design tool and faster method which create VHDL code from Simulink blocks and State flow finite state machines, as well as Simulink HDL coder supplies interfaces to collect manually-written HDL codes, HDL mimicry blocks and RAM blocks in its environment[6].

The adaptive LMS equalizer has been done using Matlab Simulink and tested, also synthesis as well as simulated done using Altera Quartus II simulation environment targeting Cyclone III FPGA. It supplies a entire design medium for designing equalizer on a programmable platform. It displays a very substantial library of modules that can be used to build many types of handling units can be used. The designed LMS Equalizer is modular, which gives the planner to check and analysis the different modules separately. Figure (5) demonstrates the block diagram of register transfer level (RTL) of internal structure of the LMS Equalizer. RTL glimmer is used in hardware description languages (HDLs) like VHDL to produce high-level submitting of a system, from which lower-level submitting and eventually active wiring will be obtained, and logic optimization can be obtained by using synthesis tools. Figure (6) shows a schematic file to assign input and output pins for the LMS equalizer system. Finally ModelSim Altera 6.1g, used and can test the data input with out of all LMS Equalizer and result is shown in Figure (7). The source signal in the simulation can be utilized to compare the result of output signal with input data, the (ce_reult) appears as clock enable. The maximum operating frequency of FPGA implementation is 100MHz and master clock frequency is used 110MHz.

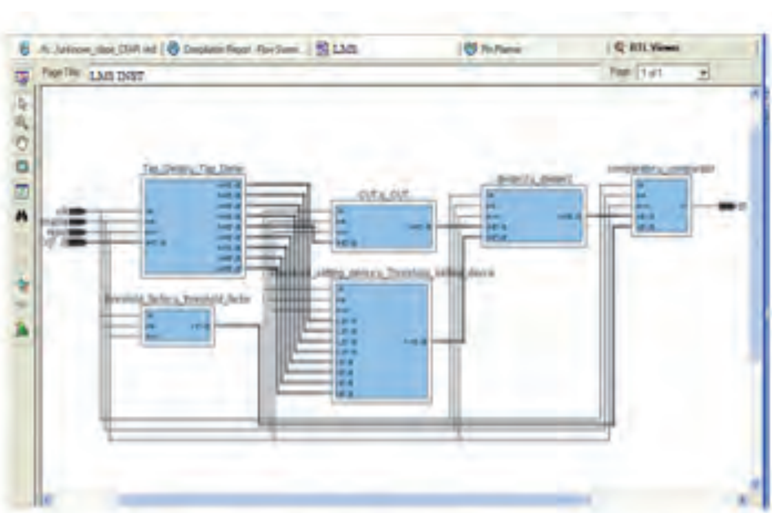


Figure 5: Register Transfer Level of Adaptive Equalizer.

Figure 5: Register Transfer Level of Adaptive Equalizer.

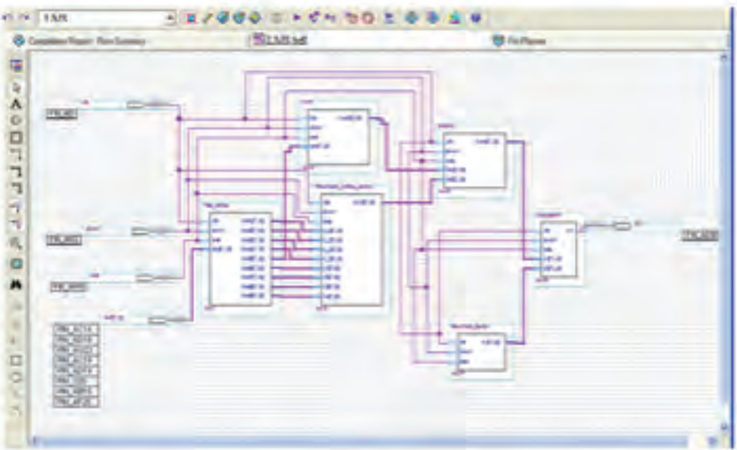


Figure 6: Top level design of the Adaptive LMS Equalizer on FPGA Platform.

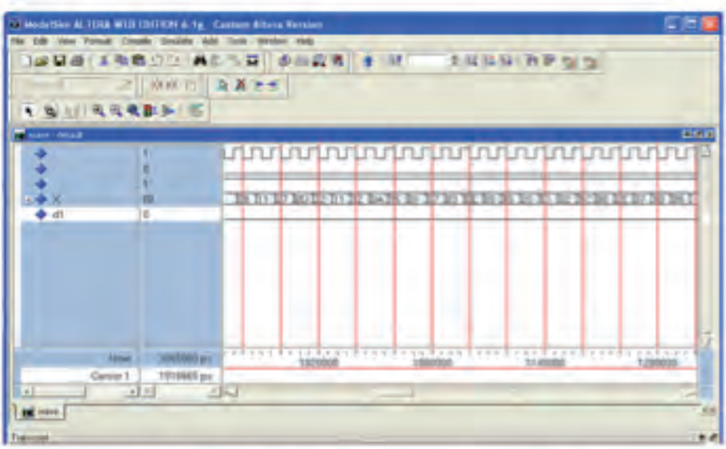


Figure 7: Waveform Simulation of Adaptive LMS Equalizer.

The input clocks that content in the Cyclone III board: 125 MHz, 50 MHz and can add clocks which is consider as external clock by the user. So, the implemented speed must be restricted, when it is implemented on Cyclone III. The synthesis reports for the LMS Equalizer can be summarized as in the Table 1

Table 1: Compilation Summery

The LMS Equalizer Core Synthesis Summary for CycloneIII (EP3C12F7801C7) device			
Resources	Utilized	Available	Percentage
Total logic elements	9018	119188	0.7
Total registers	7533	119188	0.6
Total memory bits	6550	3981332	1.6
DSP 9 bit elements	233	577	4

5.Conclusion

Adaptive LMS equalizer in this paper was designed and implemented using Matlab Simulink and FPGA platform, and the response of adaptive filter has been analyzed. Therefore, can be concluded from the simulation results that system has relatively perfect performance in terms BER, when LMS algorithm with FIR filter are used and adapted, where the effects of inter symbol interference result from Time-dispersive channels is minimized . While, the defect of LMS algorithm is that the converge is slow gradually. The implementation of the LMS algorithm is accomplished via field programmable gate array (FPGA) design through create connection link between simulink design and VHDL by using HDL coder tools which simplest and active procedure for control users. The speed, capacity consumed, and complexity are clear in the compilation report, and complexity can be improved by using multiplex multiplier, so the multipliers that used in the implementations can be reduced.

References

[1] WANG Tian-lei, "Developments in Adaptive Equalization Algorithm Research", Journal of Wuyi University [J],2009, 02(23),pp37-42.

[2] John G. Proakis, "Digital Signal Processing Principles, Algorithms and Applications", Pearson Prentice Hall, fourth Edition, page No. 909-911.

[3] Adinoyi, S. Al-Semari, A. Zerquine, "Decision feedback equalization of coded I-Q QPSK in mobile radio environments," Electron. Lett. vol. 35, No1, pp. 13-14, Jan. 1999

[4] Wang Junfeng and Zhang Bo "Design of Adaptive Equalizer Based on Variable Step LMS Algorithm". Proceedings of the Third International Sym

posium on Computer Science and Computational Technology (ISCSCT '10) Jiaozuo,2010, P. R. China, 256-258.
[5] S.Haykins, "Analog and Digital Communications" , Prentice Hall, 1996.
[6] P.K. Dash, S. Shekhar" Design and Implementation of Single OrderLMS based Adaptive System Identification on Altera based Cyclone II FPGA"IEEE Conference, April 2014.

UNIVERSIDAD
DEL ZULIA

opción
Revista de Ciencias Humanas y Sociales

Año 35, N° 89, (2019)

Esta revista fue editada en formato digital por el personal de la Oficina de Publicaciones Científicas de la Facultad Experimental de Ciencias, Universidad del Zulia.
Maracaibo - Venezuela

www.luz.edu.ve   www.serbi.luz.edu.ve

produccioncientifica.luz.edu.ve