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System Identification using Adaptive Filters

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Abstract*: *Identification of system is one of the major applications of an adaptive filters, mainly Least Mean Square (LMS) algorithm, because of its ease in calculations, the ability to withstand or overcome any conditions. The unknown System can be a FIR or an IIR filter. Same input is fed into both undefined system (which is unknown to us) and the adaptive filter, their outputs will be subtracted and the error subtracted signal will be given to adaptive filter. The adaptive filter is modified until the system which is unknown and the adaptive filter becomes relatively equal. System identification defines the type and functionality of the system. By utilizing the weights, the output of the system for any input can be predicted.

Keywords*: *System Identification, adaptive filter, Least Mean Square algorithm, NLMS algorithm, RLS algorithm.

I. INTRODUCTION

Identification of a system is the form of designing dynamic device from experimental data. The main goal is to obtain the digital model and estimate the weights (Transfer Function Coefficients) of given unknown circuit using Adaptive Filter System Identification approach. An Adaptive Filter is a digital filter, which has a function that can be controlled. Varying parameters will be adjusted using an optimized adaptive algorithm. An Adaptive Filtering technique is successfully applied in modern antennas, equalization channel problems, interference cancellation, echo cancellation and many others. This technique can be used to detect the system of undefined systems that will be varying using Signal Processing in real time. There are many structures for adaptive filtering (i.e., LMS, NLMS, RLS). LMS algorithm is simple and can be easily applied, whereas RLS algorithm is complex for computation and cost is high. But LMS algorithm takes longer time to converge, RLS takes lesser time and is faster. Convergence factor decides the accuracy of the weights. Convergence Factor is denoted by µ and if µ is very large, the algorithm will not coincide. If µ is very small, the algorithm will coincides very slow and it will be very difficult to identify the changed conditions.

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Hence the selection of proper value of the convergence factor plays an important in the convergence time period, it should neither be too small nor too large. In LMS algorithm adaption is based on the gradient-based approach that updates filter weights to the optimum filter weights.
In RLS algorithm adaption is based on the recursive approach, where the filter coefficient are found which will reduce linear weighted square function based on input signal. NLMS algorithm is very much identical to LMS algorithm, only step size is normalized. This is used to overcome the effect of convergence rate, gradient noise amplification problem and change the step size between two adjacent coefficients of the filter will also change. The main goal of using an adaptive filter for identifying the undefined system is to obtain a linear definite model which will imitate the best fit to an unknown undefined system, i.e. determination of an impulse response(h[k]), of the unknown undefined system. The impulse response is useful since it gives a precise calculated output of undefined systems for all signal input, the output obtained is convolution of signal input and impulse response.

II. RELATED WORK

Irina Dorean et.al. [1], presents identification of a system using stochastic gradient method, that is also referred as Least Mean Square algorithm. These filters are implemented using digital signal processing (DSP), for increasing performance these are implemented using ASICs or FPGA. The main logic here is taking gradient descent for estimating a time varying signal. This will find min value and by way of taking increasing steps in negative direction of gradient. It is done in order to reduce error signal value. Basically, using these two equations the unknown system is identified using LMS algorithm.

$$e[n] = d[n] - y[n] \tag{1}$$

$$c[n+1] = c[n] + \mu\, e[n] x[n] \tag{2}$$

µ is the main parameter which decides the efficiency of the adaptive output signal. We should make sure to keep it to desired value so that it will converge properly. The main reason for using LMS algorithm is that it is comparatively very simple to implement in both hardware as well software as it is computationally simple and efficiently uses memory.
Sajjad Ahmed Ghauri et.al. [2], compares the three commonly used adaptive filtering algorithms in System identification is they are Least mean square (LMS), recursive least square (RLS) and Normalized least mean square (NLMS) algorithms. LMS is computationally simple than the other two, NLMS is its normalized form and RLS is a complex but efficient algorithm.

The LMS adaptive filter updating equations are similar to the above paper, in NLMS the major challenge in finding the appropriate value of the step size is overcome by normalizing the input and its updating equation is as shown,

$$C[n+1] = c[n] + ((\mu/||x(n)||^2))*e(n)*x(n) \quad (3)$$

The RLS is a recursive algorithm which has good convergence but computationally complex moreover, it requires predefined conditions and information to update the estimation.

III. TRANSFER FUNCTION

The transfer function denotes relationship between the signal output of the control system and signal input, for all probable input values. TF is normally utilized in analysing systems similar to single input-output filters. The word is frequently utilized in reference with LTI systems. Fig. 2 represents a basic LTI system.

Fig.1 .Representation of basic LTI system

In a Laplace Transform, R(s) represents input and C(s) represents output ,so Transfer Function can be represented as G(s) given by:

$$G(s) = \frac{C(s)}{R(s)} => R(s).G(s) = C(s) \qquad (1)$$

Hence, we can express Transfer function as the ratio of the Laplace transform of output variable to Laplace transform of input variable as shown in (1) by imagining all initial conditions as zero.

IV. METHODOLOGY

Fig. 3 illustrates the working of the complete identification of system. The unknown undefined system to be recognized is a finite impulse response and infinite impulse response filter. The input signal is a white noise which is Gaussian having zero mean and unit variance. This signal will be further feed to low pass filter (LPF) where the higher frequency signals will be attenuated, the cut-off frequency is decided on the basis of the sampling frequency of the ADC. The filtered input signal is given to the unknown system and to the adaptive filter via ADC (microcontroller).

The procedure for adaptive filtering algorithms is accomplished by using a microprocessor where the adaptive filtering code is dumped. Both the input as well as the output signal from the unknown system is made to hold simultaneously using sample and hold circuit. The simultaneous operation is achieved by giving same gate pulse to both the sample and hold circuits used. Further, this will be sent through an ADC where sampled analog signal will be converted to digital. This conversion is required since we are making use of microcontroller where the adaptive filter present is a digital system. After the conversion the adaptive filtering takes place. The output from the unknown system and adaptive filter are compared, if any difference is detected then an error signal will be generated. This error signal is provided as a feedback to the adaptive filter. The adaptive filter varies itself depending on the error signal, this process repeats until the error signal is zero indicating that the adaptive filter is now similar to the unknown system. Therefore, the transform function of the unknown system is obtained. The final output will be displayed on the monitor screen. Let's understand the need of each block in detail.

A. White noise Generator

White noise is a random signal which ensures an equal density of different frequencies. White noise has a constant power spectral density. White noise is useful to find the impulse response of an electronic/electrical circuit. In discrete time, white noise remains as a discrete signal whose samples are considered as sequence of uncorrelated random variables signal having zero mean with a finite variance. Fig. 2 shows the white noise circuit.

B. Low Pass filter and Clamper

According to Nyquist criteria for sampling the repetitive signal will be corrected and reconstructed, so that the frequency of sampling should be greater than twice the maximum frequency, which is designated to sampling. Since the sampling frequency of our ADC is around 15 KHz, no component should be present in the input signal with a frequency greater than 7.5 KHz. Therefore, the white signal needs to pass through a low pass filter with a cut-off frequency of 7.5 KHz.

Since the input signal consists of both positive and negative signal, but the ADC cannot convert the negative signal. Hence, the signal needs to be clamped positively such that it contains only positive components.

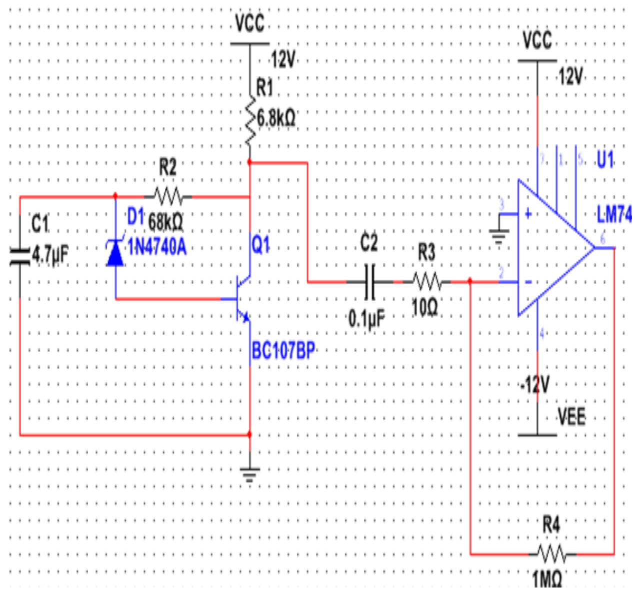

Fig.2 . White noise Generator Circuit

C. Sample and Hold

Sampling and Hold circuit receives samples from the input signal which is analog and it will hold for certain amount of time. The output obtained is a sampled form of the input signal. For adaptive filtering algorithm, input signal and its corresponding output is required simultaneously.

Hence, both the input and output signals are sampled and held during computations. To ensure that the output for that particular input is held without any error, both the sample and hold circuits should be triggered simultaneously. Fig. 4 shows the circuit of sample and hold used.

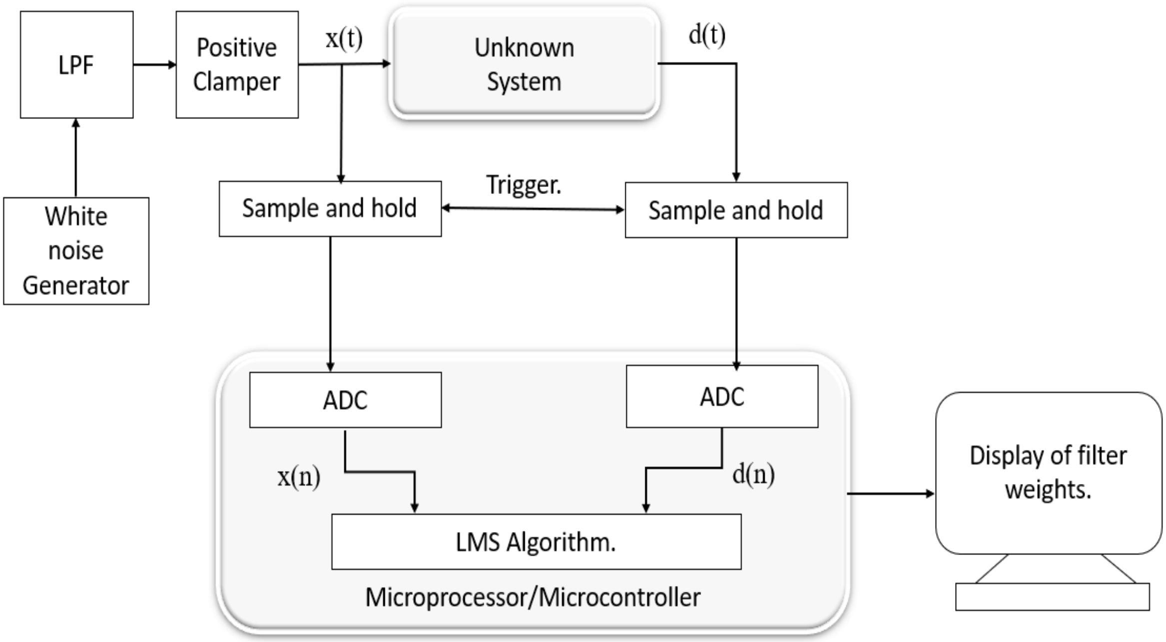

Fig.3 .Block Diagram of Identification Methodolog

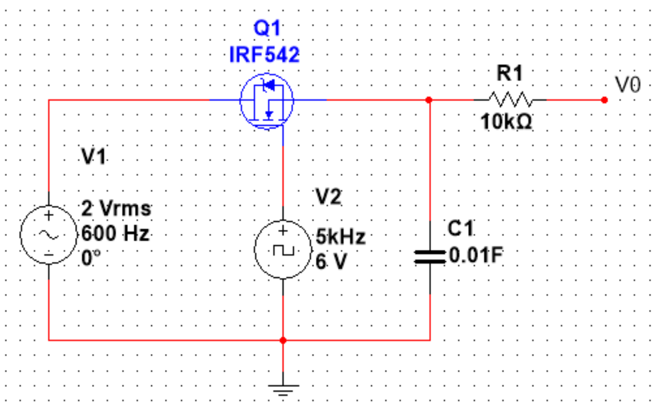

Fig.4 . Basic Circuit diagram of Sample and Hold

The main components of sample and hold circuit consist of a IRF542 transistor, a capacitor to store and hold the electric charge. IRF542 is utilized as an element for switching. Through its drain terminal input voltage is applied and over the gate terminal the voltage control is applied. During positive signal of voltage control the MOSFET is switched ON and MOSFET will be changed to OFF when the voltage control becomes zero this acts as the open switch. Capacitor will be charged to its peak value when MOSFET acts as a closed switch and the analogue signal is fed to capacitor through the drain terminal. Charging of the capacitor stops when the MOSFET switch is opened. Capacitor will experience high impedance due to the resistor R1. This causes the capacitor to hold for certain amount of time. This period can be stated as holding period. The period in which input voltages are sampled is referred as sampling period.

D. Microcontroller Configuration

Microcontroller is the heart of this system. It performs multiple functions such as analog to digital conversion (ADC), error signal calculation and realization of LMS adaptive algorithm.

The Microcontroller used is ATMEGA16A which has a 10-bit ADC with 8 channels and 1 MHz internal clock frequency. This clock frequency is insufficient; hence a 16 MHz external clock is used to speed up the ADC.

The microcontroller used is Atmega16A which is a 40 pin IC. The output signal from the sample and hold is specified as input to unknown undefined circuit as well as to pin40 of the IC, which is also the input pin for ADC channel-0, here analog input will be converted to digital form. Fig.5 shows the microcontroller interconnections.

NEX AVR USB ISP STK500V2 is used to dump the code into Atmega16a. NEX AVR USB ISP STK500V2 is a high-speed USB powered STK500V2 compatible In-system USB powered programmer for AVR family of microcontrollers.

Written code is dumped into the Atmega16a IC from the programmer by means of a tool known as Avrdude. Avrdude will be utilized efficiently through the command line for reading or writing memory chips (flash, fuse bits, eeprom, lock bits, and signature bytes) otherwise through an interactive (terminal) mode.

Using AVRDUDE in interactive mode helps in adapting distinct bytes of eeprom, determining memory contents, programming fuse/lock bits, etc. The complete chip memory in the file can be programmed through command line.

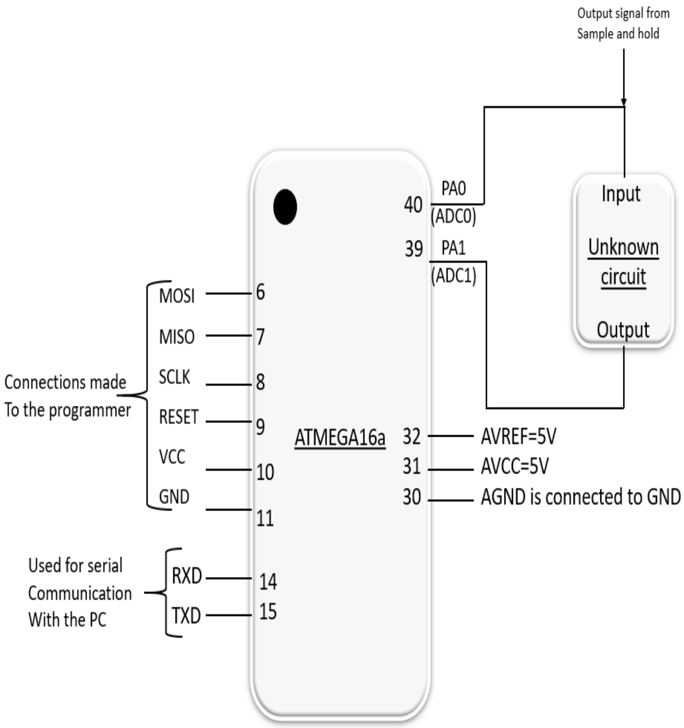

Fig.5 .Interconnection of ATMEGA16a

E. Implementation of LMS Algorithm

The flowchart of LMS algorithm implemented using Atmega16a is shown in the Fig.6 which is programmed using Atmel studio software.

The steps involved in the flowchart are

- Read two values, one from the input given to the unknown system into the ADC channel0 further stores it into x[i] and another from output of the unknown system into the ADC channel1 and store it in the variable d.
- Make A[i]=x[i] and give a[i] as the input to adaptive filter.
- Calculate the adaptive filter output 'y' using formula in (2)

$$y = y + (A[i] * W[i]) \quad (2)$$

- Calculate error using (3) and update the coefficients of the filter using (4).

$$e[n] = d[n] - y[n] \quad (3)$$

$$W[n+1] = W[k] + (\mu * x[n] * e[n]) \quad (4)$$

- Check if error is less than 0.001, if yes display the coefficients of the unknown undefined system and stops the method else repeat the similar process until the condition is satisfied.

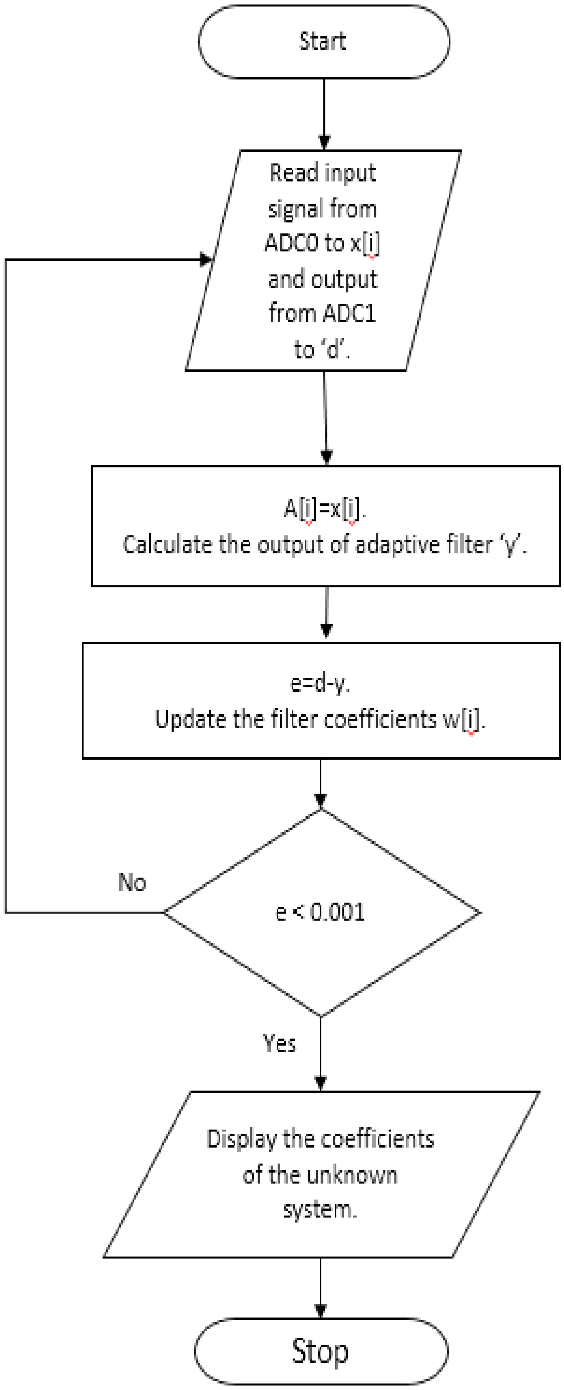

Fig.6 .Flowchart of LMS Algorithm

F. Displaying of Coefficients

Once we calculate the coefficients of the unknown system these values are sent to the computer using USB to TTL converter. It has 5 pins which are +5v, GND, RXD, TXD and +3v. This obtained data can be printed on the screen using a tool named Putty. This is an emulator, which provides free, open terminal source with a serial console and network file transfer application.

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V. RESULT AND DISCUSSION

We have tested with simple FIR circuit i.e. having only resistive circuit. Two resistor having same value is connected in series and voltage drop across resistor will be reduced to half. This half value will be the filter coefficient (weights). Below shows the figure of simple FIR resistive circuit.

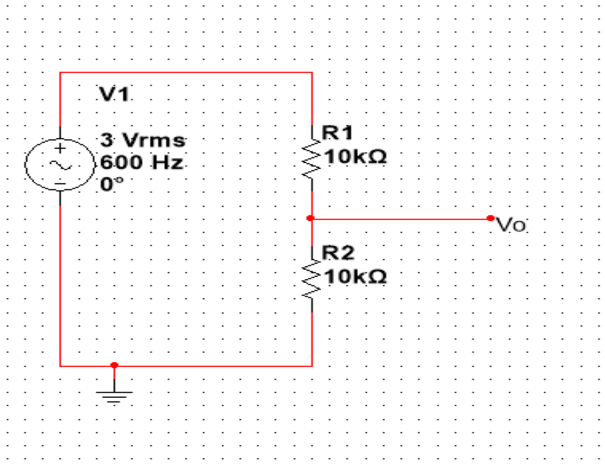

Fig 7: Series Resistive Circuit with same resistor value

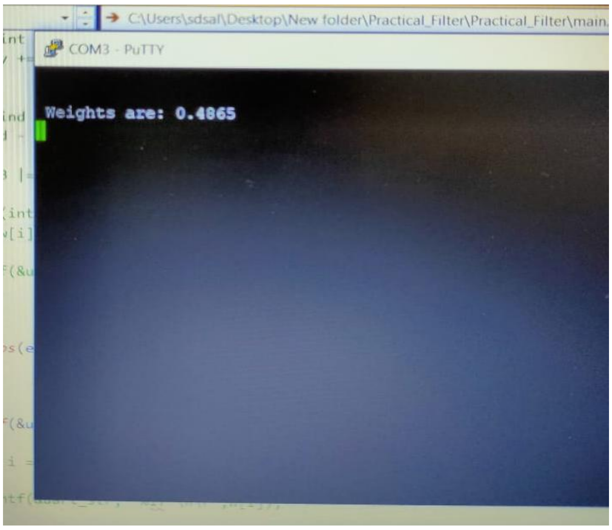

Fig 8: Result of Series same Resistive Circuit

The output is read on puTTy terminal using TTL converter. Obtained weight value is approximately equal to 0.5 as shown in Fig 8.

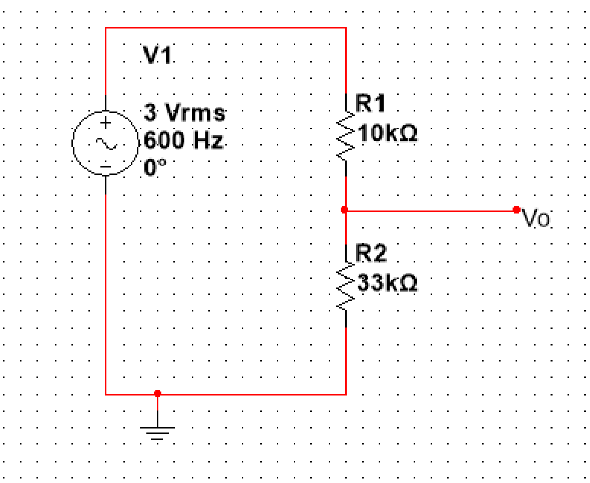

Fig 8: Series Resistive Circuit with different resistor value

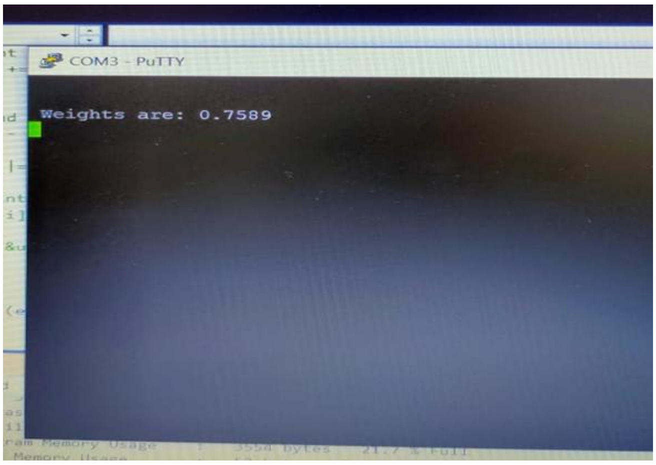

Fig 9: Result of Series Different Resistive Circuit

We have also tested for different resistor circuit and verified, which is shown in Fig 8. LMS adaptive filtering algorithm is implemented but it is not most accurate. In order to overcome this different adaptive techniques of filtering can be realized. NLMS algorithm in which the input is normalized, LMS-GA which uses a genetic search approach, LMS with varying step size and other such algorithms have been developed. RLS is another algorithm, which is computationally complex but the estimate is more accurate to the unknown system.

VI. CONCLUSION

The Adaptive algorithm is adopted here to learn the parameters of digital FIR/IIR filters that is the weights (Transfer function) of an unknown system. There are a lot of different adaptive techniques of filtering that can be realized in order to estimate the unknown function with each of them having their own advantages and disadvantages. The simplest is the LMS algorithm, it will be very much easier for applying this in hardware and software because of its simple way to compute any functions and also effective memory usage. For smaller input vectors LMS algorithm is very accurate, however for large input accuracy and performance is very hard to predict since it requires many computations with lesser step-size. In order to overcome this problem other algorithms such as NLMS, RLS, LMS-GA can be used but this is very complex and of high cost whereas it gives more accurate results. Hence the choice of algorithm can be made based on the application, for example for an application involving the estimate to be highly accurate with cost and computationally complexity not being an issue RLS algorithm can be used, but for applications where approximate result is sufficient keeping the complexity simple then LMS algorithm is preferred. Therefore, for laboratory application where the accuracy is not an issue and simplicity is also required LMS algorithm is used.

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