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DESIGN A HIGH EFFICIENT ADAPTIVE FILTER USING DISTRIBUTED ARITHMETIC

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ABSTRACT: In this paper the design of high efficient adaptive filter using distributed arithmetic is implemented. The adaptive filter has high throughput whose filter coefficients changes during runtime. The main advantage of distributed arithmetic is its high computational efficiency. Distributed arithmetic is a popular technique for implementing the computations of sum and product. Initially, normal input data and impulse data is given as input to data queue. From data queue the values are arranged in particular format and applied to adaptive filter technique. The adaptive filter technique will initialize the coefficient and save it in the register. Now adder and multiplier operations are performed based on distributed arithmetic operation. To save this outcome of the system accumulator is used. Hence this technique will improve the efficiency in effective way. From results it can observe that it gives effective output in terms of delay and area.

KEY WORDS: Analog Filter, Digital Filters, Adaptive Filter, Distributed Arithmetic, Accumulator, Data Queue, Adder, Multiplier, Register, Digital Signal Processing (DSP).

I. INTRODUCTION

Each filter in DSP has its own characteristics such as low-pass and high-pass have different type of frequency responses. The parameters like gain, stop-band attenuation, roll-off and oscillations in the responses are different for each filter. All these parameters do not match perfectly with the ideal response characteristics like infinite attenuation in stop-band, faster roll-off and unity pass band gain [1]. In order to obtain ideal characteristics filters optimized using approximation functions in designing of analog linear filter. These approximation functions use statistical methods to optimize the transfer function of the filter being designed.

The remarkable characteristics of digital filters in their performance lead to widespread use of them in digital signal processing units. The two main functions performed by filters in DSP are signal separation and restoration. If a signal mixed up with interference from other signals or noise, signal separation would be better choice [2]. For instance, a device recording heart beat of a fetus in the womb. The actual signal interfered by the heartbeat or air inhalation of the mother. In this case, a filter is employed to separate original signals from interfered signals. Therefore, they can be processed separately.

Sometimes signals may lose or distorted due to unusual reasons. In this case, a filter is used to restore the lost signals. Some examples of signal distortion are sounds recorded from a voice recorder of poor standard, blurring of images captured due to imperfect focusing of lens or shaking. The signals distorted in these cases would have been improved using signal restoration filters. Any one of the filter types either analog or digital filters can address this problem [3].

There are many ways for designing of digital filters. Each filter design is suitable for particular application in time-domain or in frequency domain. Filters for the time-domain applications are specially designed to conserve the signal shape because the information is encoded in the signal by the source. Therefore, filters in this domain are employed to preserve the shape of waveforms like signal restoration, suppressing of DC components and smoothing.

Unlike time-domain, the shape of the signal is not significant in frequency domain. Because the signals in the frequency domain contain periodic waveforms, the phase, frequency and amplitude of the signals holds the information. Therefore, filters in this domain are employed to allow certain band of frequencies which holds necessary information. Signal separation is main objective of frequency domain filters [4].

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Apart from the filters in two domains other filters are used known as custom filters. The functioning of custom filter is different from the remaining two. This type of custom filters designed to remove the unnecessary convolution that means to perform deconvolution. Adaptive filter consists of processor, which can be programmed with specific function. The function of the Adaptive filter can be modified by altering program kept inside the memory of a processor, without changing the hardware. Digital filters can be designed, tested and realized effortlessly by using usual computer. The characteristics of analog filters changes with the variations to temperatures. Whereas digital filters are not subjected to changes with time and temperature. Therefore, digital filters stable compared to analog filters.

During previous years, digitals filters are only able process low-frequency signals. Whereas analog filters is best choice for high frequency signals. As the digital technology has been developing without leaps and bounds the frequency range of digital filters increased and can operate in RF frequency range. Unlike limited functionality of analog filters on signals digital filters have special features can process signals in several ways. It makes Adaptive filter adaptable to variations in the properties of the signal.

Digital filters can achieve hard targets of filter characteristics with the help composite arrangements of filters such as series or parallel. These designs are much easier and compressed rather than analog filters.

II. OVERVIEW OF FILTERS

Analog filter:

In analog filters the filtering characteristics are achieved using some electronic elements like capacitors, resistors and operational amplifiers. This kind of analog filter is employed in noise suppression, graphic equalizers, video enhancement and wireless network systems. An analog filter to perform specific function can be implemented using some well-known organized methods. Analog filters are employed for signals, which are continuously varying quantities (ex: voice signals from a recorder, current or voltage outputs from transducers)

Digital Filter:

In digital filters the filtering characteristics are achieved by performing mathematical operations on the sampled values of digital signal. It employs a conventional processor or specific signal-processing chip. In DSP processors initially the analog signal is changed into digital format using ADC (analog to digital converter). The continuous signal is sampled and then quantized to make a digital signal.

The signal produced consists of consecutive samples indicating different input signals values in the form of binary numbers. The next stage processor in Adaptive filter performs various mathematical operations on the transformed digital signal. These operations may include multiplication by some constants and adding the resultant product values. The resultant signal also consists of modified sampled values, which undergo digital to analog conversion by DAC. Instead of continuous signals like voltage or current, Adaptive filter processes the signals. Which are present in the form of samples indicating binary numbers. The figure (1) illustrates the steps included in digital filtering system.

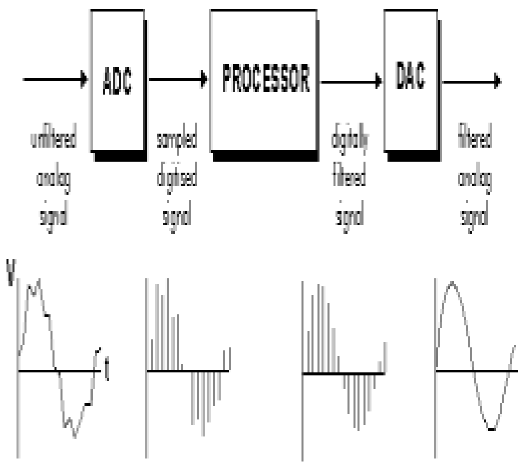

Fig. 1: DIGITAL FILTER SYSTEM

Analog and digital filters are contrast in their characteristics like performance and operating range. Analog filters are active in wide range of frequencies whereas digital filters are limited to small range. However, the

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performance of digital filters is unbeatable and cannot be obtained using analog filters. So, we should make right choice suitable for specific applications. The choice makes extrodinary variation in addressing problems in filtering. In analog filters we must focus on drift in characteristics of resistors and capacitors. Similarly, in digital filters limitations of signal ranges must be focused, as the Adaptive filter can provide most excellent performance.

Conventionally, in the digital signal processors the filters operate on signals which are present in time domain as the signals are sampled in time-domain with constant time durations. There are many ways to sample a signal. One of them is sampling in space (Example: Acquiring an image from array of sensors with a gap of small distances attached to aircraft wing). The time and space are often used domains in sampling of a signal. The time domain indicates the signal samples taken at regular intervals of time. Other domain frequently used is frequency domain. The number of samples for one second shown in frequency domain.

Time domain description of filter:

The equation (1) depicted below represent first order differential equation of a filter. If more number of input samples included in the output equation, the order of the filter would gets increased. The output of a Adaptive filter is expressed here as a weighted sum of earlier and its present inputs.

$$y(m) = ay(m-1) + x(m) \text{.........(1)}$$

In the above equation y(m), x(m) represents output, input of a filter respectively. Where 'a' is a co-efficient of the filter.

Impulse response:

The impulse response of filter is defined as output of the filter for the given impulse input. A equation shown below (2) discrete time impulse response of a filter for the impulse input when m=0, 1, 2, so on.

$$y(m) = a^m \quad \text{where } m=0, 1, 2, \ldots \ (2)$$

It is considered that y(-1) = 0.The impulse response is highly essential for analyzing a filter. The reasons for this are the signals in the digital signal processing are given in the form discrete samples the output of the impulse input is also comprise of responses of the chain impulses in the input.

The impulse signal consists of a series of impulses at all frequencies with equal magnitude. Therefore, impulse response contains filter response at all frequencies.

The FFT of impulse response gives frequency response of the filter in the same way the IDFT of frequency response gives impulse response.

III. APPROXIMATE SUM-OF-PRODUCTS DESIGNS BASED ON DISTRIBUTED ARITHMETIC

Because of the adaptability of the degree of parallelism in the appropriated math structure, the region speed tradeoff can be balanced. Appropriated number juggling is somewhat sequential activity that figures the internal result of two vectors in equal. It requires no increase and it has a proficient system to play out the SOP activity. Approximate SOP (ASOP) model dependent on truncation.

There are three two input 16-bit adders, one 3-input 16-bit adder, 16 look up tables with 8 cases as shown from figure (2). Accumulator consists of 16 elements. In this approximate model K and N values plays important role. The value of K is 3 and N is reduced into m bits at the LSB part of ak and bk for k = 1, 2, and 3 which are truncated. m = 8, 6, and 4 bits are implemented. For this implementation, three two-input 16 − m bit adders, one three-input 16 − m bit adder, 16 − m lookup tables with eight cases, and final accumulator with 16−m elements are required.

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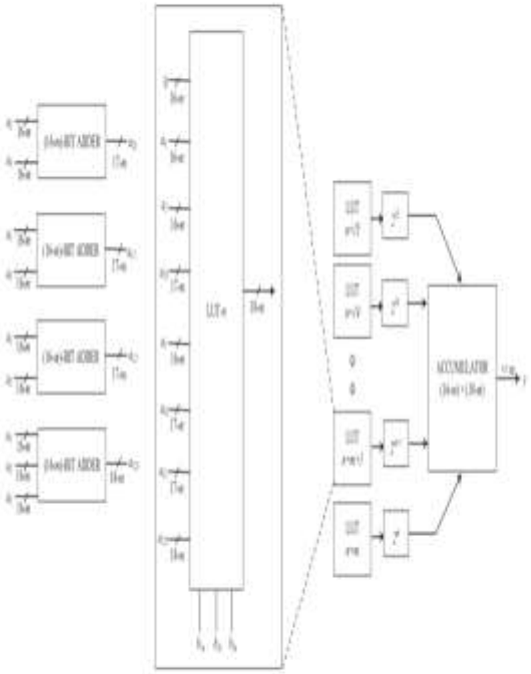

Fig. 2: APPROXIMATE LOOKUP TABLE AND CORRESPONDING ASOP (ASOP1) STRUCTURE FOR K = 3 AND N = 16

This considerably reduces the hardware utilization at all the levels. The approximate model with reduced elements is shown in Fig. 2. It should be noted that in ASOP1, the number of input bits to the adders is reduced, which further reduces the complexity of accumulator ($16 - m \times 18 - m$).

IV. ADAPTIVE FILTER USING DISTRIBUTED ARITHMETIC

The below figure (3) shows the block diagram of adaptive filter using distributed arithmetic. Initially, normal input data and impulse data is given as input to data queue. Data queue will arrange the input data in particular format. Both coefficient and register plays important role in adaptive filtering technique. Adder and multiplier will perform its operation based on distributed arithmetic unit. Accumulator will save the output which is coming from adder and multiplier block.

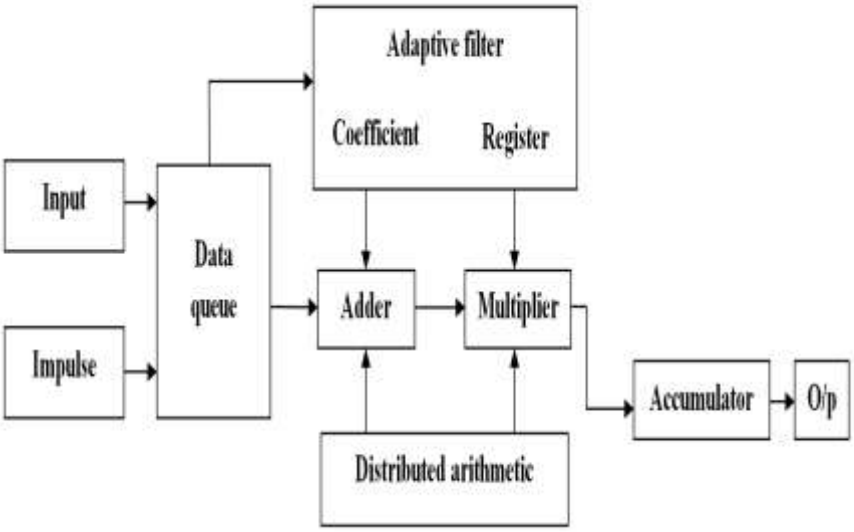

Fig. 3: BLOCK DIAGRAM OF ADAPTIVE FILTER USING DISTRIBUTED ARITHMETIC

Initially, the input data and impulse data are followed by the data queue block. The main intent of data queue is to maintain the data in sequence manner. This data is modified by the addition of entities at one end of sequence and removal of entities from other end sequence.

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The Register is another sort of successive rationale circuit that can be utilized for the capacity or the exchange of twofold information. This successive gadget stacks the information present on its data sources and afterward Shift s or "Shift s" it to its yield once every clock cycle, thus the name Shift Register.
A register fundamentally comprises of a few single piece "D-Type Data Latches", one for every information bit, either a rationale "0" or a "1", associated together in a sequential sort daisy-chain course of action with the goal that the yield from one information lock turns into the contribution of the following hook, etc.
An accumulator is a register for short-term, intermediate storage of arithmetic and logic data in a computer's CPU (central processing unit). The most elementary use for an accumulator is adding a sequence of numbers.

The term ‘adaptive’ refers to acquiring one’s behavior. A filter is said to be adaptive filter, if it has the capability to adjust according to the characteristics of the input signal. Adaptive filters are significant in various applications of DSP like noise suppression, termination of echoes in phones, signal enhancement in medical diagnosis and control systems. Adaptive filters are feasible to employ for the signals which are changing continuously. When a signal is changing its characteristics continuously the noise also varies with the time and the amount interference by the noise in the varying signal is cannot be determined accurately. The effective suppression of noise takes place only if the filter has ability to adjust to the signal. Consider a frequency spectrum of a signal which has strong noise interference. In this case filtering by using a conventional filter will not conserve the shape of the signal.

V. RESULTS

The below figure (4) RTL Schematic of proposed system.

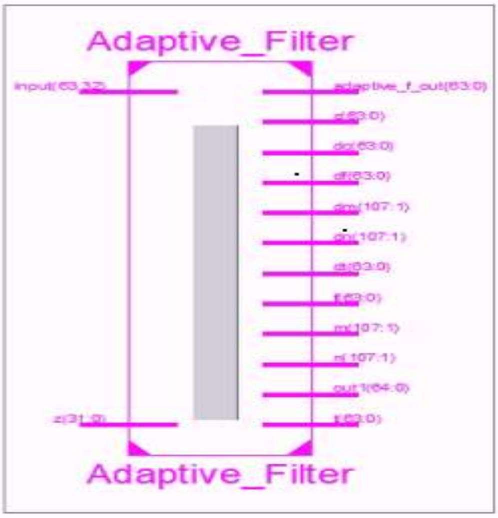

Fig. 4: RTL SCHEMATIC OF ADAPTIVE FILTER USING DISTRIBUTED ARITHMETIC

The below figure (5) shows the Technology schematic of proposed system. Technology schematic is the combination of Look up table, truth table, equation and K-Map.

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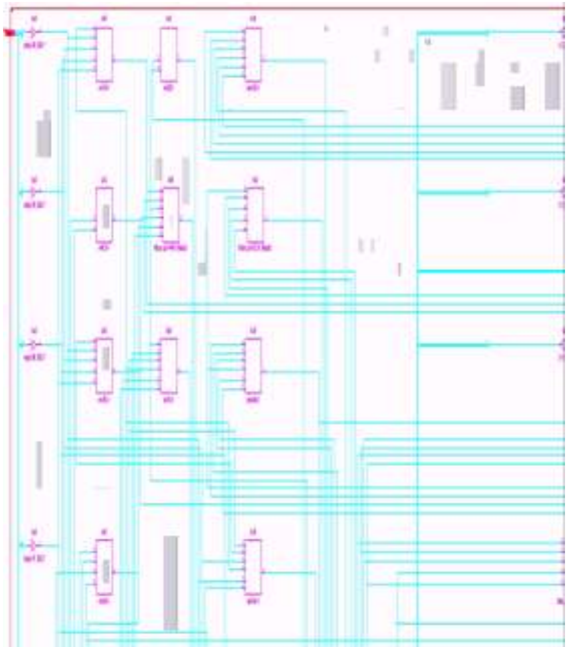

Fig. 5: TECHNOLOGY SCHEMATIC OF ADAPTIVE FILTER USING DISTRIBUTED ARITHMETIC

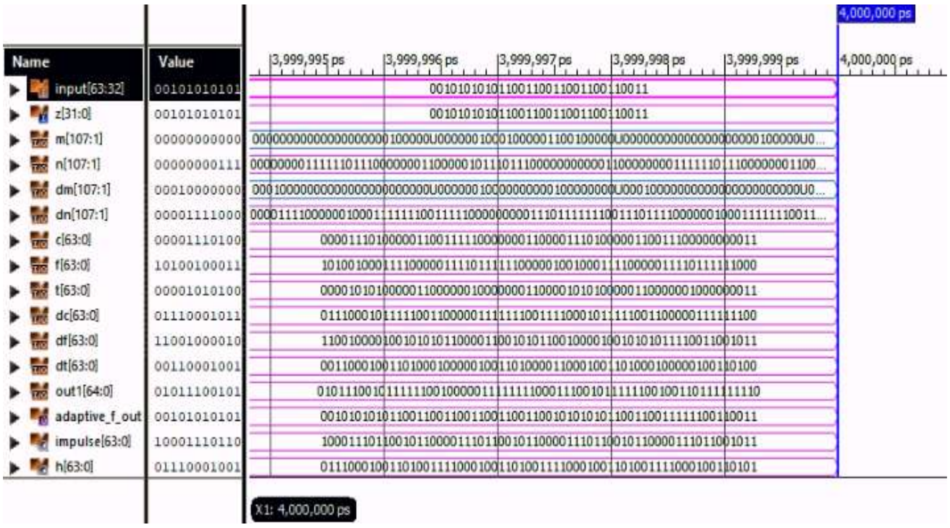

Fig. 6: OUTPUT WAVEFORM OF ADAPTIVE FILTER USING DISTRIBUTED ARITHMETIC

The below figure (7) shows the delay representation of proposed system. in this total delay, logic delay and route delay are given.

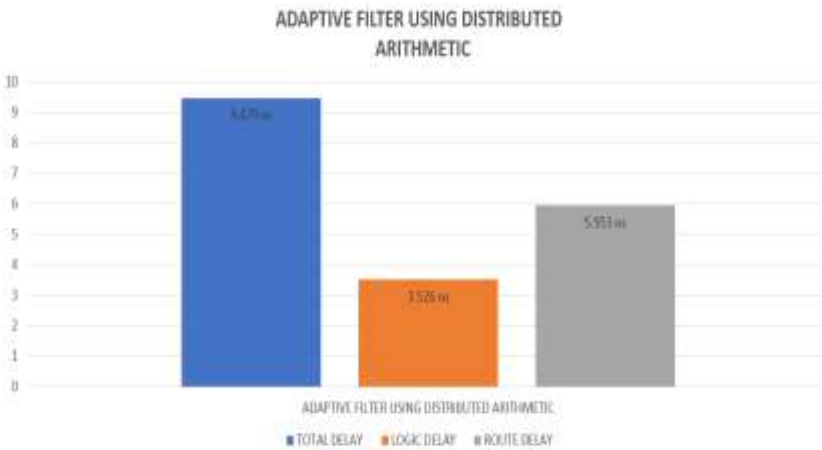

Fig. 7: DELAY REPRESENTATION OF PROPOSED SYSTEM

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VI. CONCLUSION

Hence in this paper the design of high efficient adaptive filter using distributed arithmetic was implemented. Distributed arithmetic operation provides high computational efficiency in this system. Data queue plays very important role in entire system. The adaptive filter technique will initialize the coefficient and save it in the register. To save this outcome of the system accumulator is used. Hence this technique will improve the efficiency in effective way. From results it can observe that it gives effective output in terms of delay and area.

VII. REFERENCES

[1]. Suganthi Venkatachalam and Seok-Bum Ko, "Approximate Sum-of-Products Designs Based on Distributed Arithmetic", 1063-8210 © 2018.

[2]. Atul A. Chandekar , Mahesh Pawar , "Delay and Power Optimized Adaptive Filter using Distributed Arithmetic", 978-1-5090-5686-6/17/$31.00 ©2017 IEEE.

[3]. Mphd Tasleem Khan, Shaik Rafi Ahamed, "Low Complexity and Critical Path based VLSI Architecture for LMS Adaptive Filter using Distributed Arithmetic", 2017 30th International Conference on VLSI Design and 2017 16th International Conference on Embedded Systems.

[4]. C S Vinitha, R K Sharma, "Memory- Based VLSI Architectures for Digital Filters: A Survey", 978-1-5090-5384-1/16/$31.00 ©2016 IEEE.

[5]. Subrahmanyam Mula, Vinay Chakravarthi Gogineni, and Anindya Sundar Dhar, "Robust Proportionate Adaptive Filter Architectures Under Impulsive Noise", 1063-8210 © 2014 IEEE.

[6]. Vakulabharanam Ramakrishna, Tipparti Anil Kumar, "Low Power VLSI Implementation of Adaptive Noise Canceller Based on Least Mean Square Algorithm", 2013 4th International Conference on Intelligent Systems, Modelling and Simulation.

[7]. K.Deergha Rao, Ch..Gangadhar, "VLSI Realization of Adaptive Equalizers of SIMO FIR Second Order Volterra Channels", 1-4244-0387-1/06/$20.00 c 2006 IEEE.

[8]. Ming-Hwa Sheu, Ho-En Liao, Shih-Tsung Kim, Ming-Der Shieh, "A Novel Adaptive Algorithm and VLSI Design for Frequency Detection in Noisy Environment Based on Adaptive ITR Filter", 0-7803-6685-9/01/$10.0002001 EEE.

[9]. Alejandro Diaz-Sanchez and Jaime Ramirez-Angulo, "A Novel VLSI Sampled-Data Adaptive Filter", 0-7803-3583-X/97 $10.00 01997 IEEE.

[10]. Alejandro Dim-Sanchez'" and Jaime Ramirez-Angulon, "VLSI Voltage Mode FIR Analog Adaptive Filters", 0-7803-3636-4/97 $10.00 0 1997 IEEE.

[11]. Jeng-Kuang Hwang and Ching-Hsien Chang , "VLSI Chip Design of a CORDIC-Based Adaptive Lattice Filter", 0-7803-2612-1195 $4.00 0 1995 IEEE.

[12]. Seong-Mo Park, "A VLSI Architecture For Adaptive Signal Processing" CH2998-3D 1Kx)o-OO83$01.000 1991 Lefe.

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