FAST TWO DIMENSIONAL DIGITAL FILTER DESIGN BASED ON FAST FOURIER TRANSFORM

MUZHIR SHABAN AL-ANI
[2]University of Human Development, College of Science and Technology, Department of Information Technology, Sulaimani, KRG, Iraq
E-mail: muzhir.al-ani@uhd.edu.iq

ABSTRACT

Digital signal processing (one dimension or two dimension) is one of the most powerful field that will shape 21st century science, engineering and technology. The field of digital filter design becomes an important issue for their wide range of applications. Consequently two dimensional (2D) filter design plays is an important field of processing that have direct application on digital image processing. The main objective of this work is to design an efficient fast 2D digital filter, in which can be implemented via fast Fourier transform. The motivation of this approach is concentrated on the efficient high speed digital filter design. To achieve this target the digital 2D digital filter is implemented to avoid the raised problems. The implemented approach gives an accurate design to select filter type and size according to their application. Good performance in filtering operation and processing time are achieved in testing the implemented approach. The processing time gain is increasing rapidly according to the increasing in the filter size comparing with the direct convolution.

Keywords: *2D Filters, Filter Design, Real Time Filters, FFT, and 2D Convolution.*

1. INTRODUCTION

Existing four main types of filters; low pass filter (LPF), high pass filter (HPF), band pass filter (BPF) and band stop filter (PSF), on the other hand LPF is the fundamental filter type in which it is possible to generate all other types from LPF [1,2]. So most of the filter design concentrated on LPF design that is the core of all filter design. Either one dimensional (1D) or two dimensional (2D) filters, these filters can be analog or digital depends on the form of the data [3,4].

2D digital filter has wide range of application and the most important one in concerned with image processing [5]. 2D digital filter has attracted much attention in the field of digital image processing [6]. These filters are useful in seismic data processing, graphic equalization, astronomy, biomedical image processing, biomedical imaging, biometrics, pattern recognition, object recognition, remote sensing, geographical images, etc. [7,8].

There are two main categories corresponding to the design of digital filters; non recursive filter or finite impulse response (FIR) and recursive filter or infinite impulse response (IIR), in addition each of these two categories have many sub divisions [9]. Also filter implementation can be performed in time domain (direct way) and in frequency domain (indirect way). It is a big challenge to select the filter type that guided to be implemented, this work will be concentrated on 2D FIR type in frequency domain [10].

Many algorithms are considered as filter design, these algorithms are either dealing with time domain or frequency domain representations [11]. Some algorithms go to specific narrow band filter design with good gain [12]. As a real time implementation is needed, it is better to introduce a hardware aspects of digital filter in order to minimize computational operations [13]. Special hardware pipelined filter is implemented based on structure [14,15]. Optimizing filter design is an efficient technique to be achieved [16,17]. An advanced discrete wavelet transform filter bank structure based on low complexity is implemented to achieve high performance [18].

One important thing of 2D digital filters and their applications on images is how to achieve the big amount of operations at a real time [19]. Any algorithm can be realized via hardware or software implementation [20]. The compensation between hardware and software is an excellent solution to

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

reach an acceptable real time operation, in addition to reliable system [21,22].

2. RESEARCH OBJECTIVE

Recently digital filters play an important role in many applications, including two dimensional applications. For this reason digital filters leading to large number of papers have been published on this subject. The aim of this work was to study the two dimensional digital filters applied on images using fast Fourier Transform. Then this work try to design an efficient approach to study and design two dimensional digital filter via fast Fourier transform.

3. FILTER PERFORMANCE

Finite impulse response filter design starts with its specifications in either time domain or frequency domain [23]. In the time domain, the design aims to generate the impulse response [24]. In the frequency domain, the design aims to generate spectrum response as shown in figure 1 in which illustrate the low pass filter [25]. Important parameters are pass band frequency and stop band frequency ωp and ωs, the transition band Δω=ωs−ωp, stopband, cutoff frequency ωc=(ωp+ωc)/2, pass band ripple δp and stop band ripple δs. The two ripples are assumed to be equal [26]. Usually for ideal filter, the pass band frequency magnitude is normalized to be one and the stop band to zero, and the frequency response of the designed filter oscillates between one and zero [27].

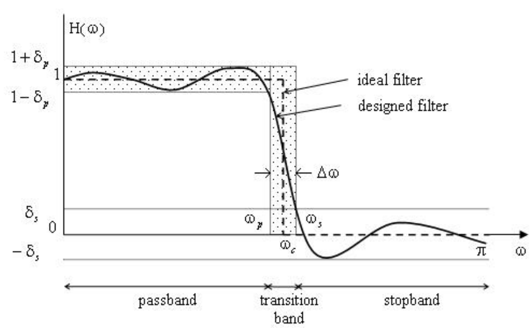


Figure1: The Ideal Filter Design

4. RELATED WORK

Many works are published related to the field of two dimensional digital filter, and below concentrated on some of these works:

Yu Wang, et al. (2011) implemented a multi-objective scalable algorithm by local search that is specifically proposed for a multi-objective optimization problem. To evaluate the efficiency of this approach, a comparison of experimental results with conventional methods are done including four typical cases of IIR filter design. The experimental results show that the proposed method can effectively improve the linear phase response of the designed filter and can obtain a lower order filter. In addition, it reaches them with computational costs comparatively lower than the traditional methods [28].

Mohammad Shams Esfand Abadi, Sahar Nikbakht Aali (2014), designed the one-dimensional adaptive filtering algorithms to 2D structures and establishes the new adaptive 2D filters. This algorithm reduces the computational complexity in 2D adaptive filtering applications. Next, they presented a unified approach to the establishment and analysis of the quadratic average family performance of the 2D adaptive filtering algorithm. This analysis is based on energy conservation arguments and does not need to assume a Gaussian or white distribution for the design. The implemented approach emonstrated good performance by the results of multiple selections in the identification of the 2D system and the 2D adaptive noise cancellation for the restoration of images [29].

Andrzej Handkiewicz et al. (2014) developed an expert tools that are the basis of the method of synthesis of non-reciprocal multiport circuits without losses composed of spinners and capacitors. The algorithms are written in C ++ and the tools form a friendly environment to automate the design of filters, filter pairs and filter banks. It is possible to design in this environment not only classic structures like Butterworth, Chebyshev and elliptical filters. The nonreciprocal prototype circuit makes it possible to design more complex filters, including the cross sections necessary to improve the characteristics of the filtering process [30].

Jun-Zheng Jiang et al. (2015) derived the sufficient state for a perfect reconstruction from using the poly phase decomposition of the analysis and synthesis filters and the rapid structure of the filter bank is implemented. Then, the perfect reconstruction condition in the frequency domain is transformed into a set of quadratic equations with respect to the filter prototype, which is used to formulate the design problem in an unrestricted optimization problem. An efficient iterative

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

algorithm is proposed to solve the problem. Numerical examples are included to verify the validity of the perfect reconstruction condition and the effectiveness of the design method [31].

Abhijit Chandra, Sudipta Chattopadhyay (2016) designed an effective digital hardware filter that has attracted attention which needs to be addressed through various useful means. A popular method has been to encode the derivation coefficients of this filter in the form of the sum of the powers of two signatures and, consequently, the multiplication operation is replaced by simple addition and change. This approach presented a detailed review of the basic design approaches for the synthesis of a high quality finite impulse response filter. In addition, traditional and heuristic search algorithms have been properly incorporated and organized [32].

Akhilesh Gotmare et al. (2017) presented a comprehensive review of the use of structural research for the identification of systems and the design of digital filters. This work focused on the identification of different systems using adaptive filters with infinite impulse response and Hammerstein models, as well as the estimation of chaotic systems. In addition to presenting a comprehensive review of the various swarm computing schemes and evolution for system identification as well as digital filter design, the paper also looks for the servo as a quick reference for some popular evolutionary computational algorithms [33].

T. Bindima, Elizabeth Elias (2017) proposed multi-objective optimization algorithm of the artificial bee colony with a complete search space to find the optimal coefficients of the Farrow sub-filter. In addition, a new low-complexity implementation approach for finite precision of variable digital filters is also proposed using a minimal spanning tree approach. The minimum spanning tree approach deploys the shift inclusive differential coefficients and the various common sub-expression elimination to optimize the multiple constant multiplications involved in filter fabrication. The hardware system is implemented to avoid complexity and power consumption [34].

Judhisthir Dash et al. (2017) designed a linear digital dual-band multifunctional filter (LPDBF) is designed to provide a hybrid meta-heuristic technique called the Hybrid Differential Hybrid Algorithm. These filters are required in different modern digital systems specific for the simultaneous processing of signals in two or three different channels. The proposed approach is an efficient evolutionary hybrid technique and is modeled taking into account both the benefits of optimizing differential evolution of improvement and firefly techniques. The general search capability of the improved differential evolution technique is enhanced by improved firefly movement. The performance of the proposed method of linear phase double-band design phase is contrasted with few popular methods of optimal design [35].

According to these related works, it is clear that these works are concentrated on filter types, filter design and filter performance as well as the accuracy of the implemented method and algorithm. The proposed approach try to design and implement an efficient two dimensional digital filter applied on images. The proposed approach differ from the literatures as a real design and implementation using the processor and FFT. In addition it is supported by real applications.

5. METHOGOLOGY

The methodology of this approach is concentrated on fast Fourier transform and filter design that are explained in detailed in the following items.

5.1 Implemented Approach

2D filter design depends on two main aspects; the design of the filter impulse response and the design of the applied window. The impulse response of the filter depends on the application and on what is the function of the filter. The applied window depends on its size and its type depends on the truncation width that fixed to be applied. The main steps of the implemented approach are demonstrated below (figure 2):

- Received inputs that required input image, input window, and input filter response.
- Multiply input window with the input filter response to generate the truncated impulse response.
- Convolution of the truncated impulse response with the input image to get the filtered image.

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

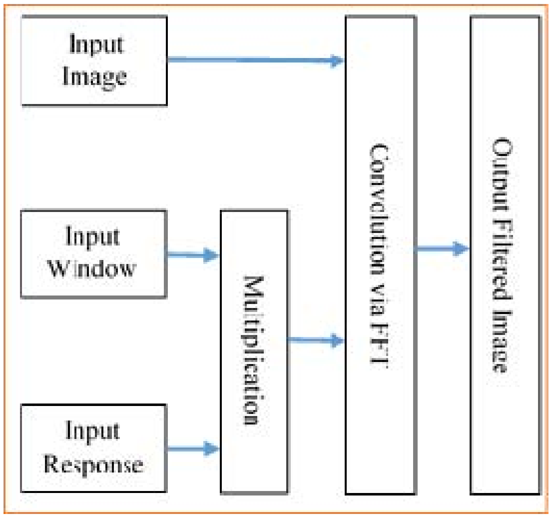


Figure 2: Steps Of The Implemented Approach

5.2 Fast Fourier Transform

The implemented approach concentrated on the filter design applied in frequency domain, so the suitable way is to use FFT in the fast form. Cooley and Tukey ate the first two researchers that implemented FFT at 1967 and their approach was a principle way to minimize number of operations. As in image processing and filtering it is required high number of operations so the best way is to adapt fast method in this approach.

Forward and inverse of one dimensional discrete Fourier transform (1D DFT) are performed by the following two equations:

$$X(k) = \sum_{n=0}^{N-1} x(n)\, e^{-j2\pi nk/N} \qquad (1)$$

$$x(n) = \frac{1}{N}\sum_{n=0}^{N-1} X(k)\, e^{j2\pi nk/N} \qquad (2)$$

Where N represents the number of values, n and k represent the input and output index.
In order to explain the repetition of the values let us take 8 input values, so the multiplication of the index (n*k) is demonstrated in table 1. This table indicates that there are 64 multiplication values and these values can be explained as below:

- All values of the first row is zeros.
- All values of the first column is zero.
- There are repetition of similarity equal to half of remain values.
- There are repetition of sine and cosine similarity as shown in table 2 and table 3.

If the number of input values is N, then index n varied from 0 to N-1 and index k varied from 0 to N, then number of index multiplication is equal to N*N. So the general equation can be reconstructed as below:

- No. of values of the first row = N.
- No. of values of the first column = N.
- Half No. of values = ½ *N*N.
- Similarity of Sine and Cosine leading to half of half.

So the final equation can be rewrite as below:

$$Multiplications = \frac{1}{2}\left(\frac{1}{2} N * N - 2 * N\right) \quad \ldots (3)$$

$$Multiplications = \left(\frac{1}{4} N * N - N\right) \quad \ldots (4)$$

$$Multiplications = \frac{1}{4} N\,(N - 1) \quad \ldots (5)$$

The procedure starts by generating a strength of sine and cosine values, then these values are arranged in vector to be read as needed. So in this way all the calculations are cancelled and the only read operation is required that take minimum time.

Table 1 Multiplication Of Index (N, K)

i*j	0	1	2	3	4	5	6	7
0	0	0	0	0	0	0	0	0
1	0	1	2	3	4	5	6	7
2	0	2	4	6	8	10	12	14
3	0	3	6	9	12	15	18	21
4	0	4	8	12	16	20	24	28
5	0	5	10	15	20	25	30	35
6	0	6	12	18	24	30	36	42
7	0	7	14	21	28	35	42	49

Table 2 Sine Values Of Multiplication Of Index (N, K)

0.7071	1.0000	0.7071	0.0000	-0.7071	-1.0000	-0.7071	-0.0000
1.0000	0.0000	-1.0000	-0.0000	1.0000	0.0000	-1.0000	-0.0000
0.7071	-1.0000	0.7071	0.0000	-0.7071	1.0000	-0.7071	-0.0000
0.0000	-0.0000	0.0000	-0.0000	0.0000	-0.0000	0.0000	-0.0000
-0.7071	1.0000	-0.7071	0.0000	0.7071	-1.0000	0.7071	-0.0000
-1.0000	0.0000	1.0000	-0.0000	-1.0000	0.0000	1.0000	-0.0000
-0.7071	-1.0000	-0.7071	0.0000	0.7071	1.0000	0.7071	-0.0000
-0.0000	-0.0000	-0.0000	-0.0000	-0.0000	-0.0000	-0.0000	-0.0000

Journal of Theoretical and Applied Information Technology
15th December 2017. Vol.95. No 23

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

Table 3 Cosine Values Of Multiplication Index (N, K)

0.7071	0.0000	-0.7071	-1.0000	-0.7071	-0.0000	0.7071	1.0000
0.0000	-1.0000	-0.0000	1.0000	0.0000	-1.0000	-0.0000	1.0000
-0.7071	-0.0000	0.7071	-1.0000	0.7071	0.0000	-0.7071	1.0000
-1.0000	1.0000	-1.0000	1.0000	-1.0000	1.0000	-1.0000	1.0000
-0.7071	0.0000	0.7071	-1.0000	0.7071	-0.0000	-0.7071	1.0000
-0.0000	-1.0000	0.0000	1.0000	-0.0000	-1.0000	-0.0000	1.0000
0.7071	-0.0000	-0.7071	-1.0000	-0.7071	-0.0000	0.7071	1.0000
1.0000	1.0000	1.0000	1.0000	1.0000	1.0000	1.0000	1.0000

FFT procedure is implemented via the many steps (figure 3): input image, organize image, generate sine and cosine values, organize vectors, apply FFT then choose the indicated values of sine and cosine.

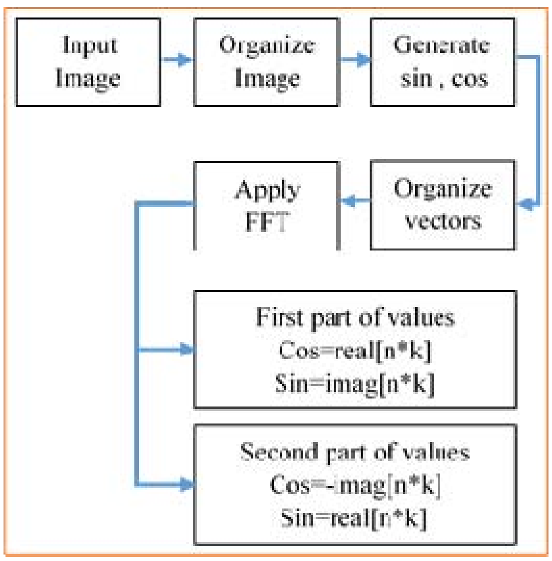


Figure 3: Steps Of FFT Architecture

5.3 Fast Fourier Transform

The design of the filter passes into a specified procedure to generate the desired impulse response. This design is concentrated on FIR in which the first design deals with LPF. When the impulse response is calculated according to the required characteristics, then for LPF is performed to pass certain low band and stop certain high band. The transition band is selected perfectly to resizing the coefficients of the filter which represents the performance of the filter. The performance of the filter characteristics is related to upper and lower tolerance.

The implementation of selected 2D response can be achieved via separable 1D functions:

$$h(n1,n2) = h1(n1)\ h2(n2) \qquad (6)$$

where $h1(n1) = 0$ outside the interval $0 \le n1 \le N1 - 1$

where $h2(n2) = 0$ outside the interval $0 \le n2 \le N2 - 1$

So the frequency response of equation (6) will be as below:

$$H(n1,n2) = H1(n1)\ H2(n2) \qquad (7)$$

The implementation of selected 2D window can be achieved via separable 1D windows:

$$w(n1,n2) = w1(n1)\ w2(n2) \qquad (8)$$

So the frequency response of equation (8) will be as below:

$$W(n1,n2) = W1(n1)\ W2(n2) \qquad (9)$$

The implemented of 2D filter can be performed as below:

$$y(n1,n2) = \sum_{k1=0}^{N1-1}\sum_{k2=0}^{N2-1} x(k1,k2)\ h(n1-k1, n2-k2) \qquad (10)$$

$$y(n1,n2) = \sum_{k1=0}^{N1-1} h1(n1-k1) \sum_{k2=0}^{N2-1} x(k1,k2)\ h2(n2-k2) \qquad (11)$$

The implemented of 2D filter can be performed via 1D filter:

$$f(k1,n2) = \sum_{k2=0}^{N2-1} x(k1,k2)\ h2(n2-k2) \qquad (12)$$

So the final filter implementation can be performed as below:

$$y(n1,n2) = \sum_{k1=0}^{N1-1} h1(k1,k2)\ f(k1-n2) \qquad (13)$$

The 2D FIR filter implementation can be FFT can be performed as below:

$$H(k1,k2) = FFT\,[h(n1,n2)] \qquad (14)$$

$$X(k1,k2) = FFT\,[x(n1,n2)] \qquad (15)$$

$$Y(k1,k2) = X(n1,n2)\ H(k1,k2) \qquad (16)$$

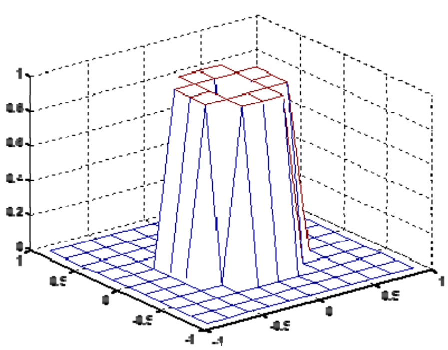
$$y(n1,n2) = IFFT\,[Y(k1,k2)] \qquad (17)$$

6. RESULTS AND DISCUSION

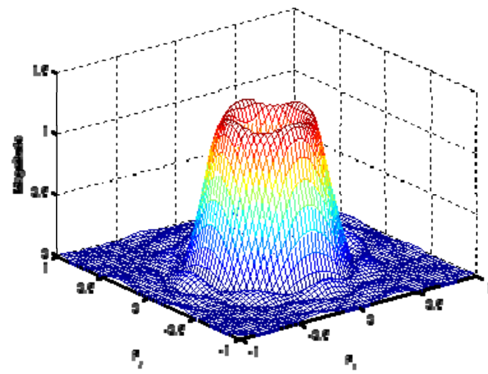
In the above design the overall system is implemented to generate window, impulse response and the testing of this design to a certain image. To achieve high performance, the implementation started from 11*11 filter size of low pass filter as in

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

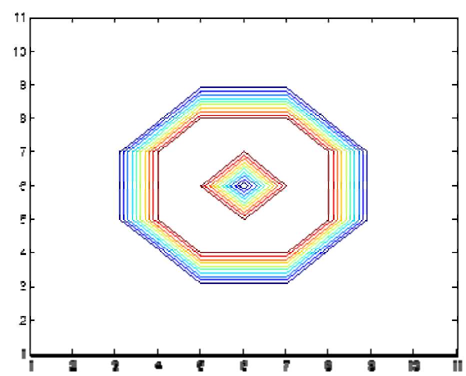
figure 4, this figure demonstrates ideal Kaiser widow, filter response, and their contour in a, b, c and d respectively. The implemented procedure started from filter size of (11*11) up to (101*101) and minimize this work, the demonstrated figures shown only up to (51*51) filter size. Figures 4, 5, 6, 7 and 8 demonstrated the filter design of sizes (11*11), (21*21), (31*31), (41*41) and (51*51) respectively. At these figures it is clear that the precise of filter will be increase according to increasing in the filter size that means according to increasing in the filter coefficients. On the other hand the inclination of the filter response is decreased so the transition width of the filter response is decreased. Figure (9) shows one of the high resolution image that used in this work, this image is of size 3096*4128 that have 12780288 pixels. This figure shows three parts a, b and c that are the original image, image after applying low pass filter, and then the contour of the image. Table 4 demonstrated a brief of the variation in number of operations and the required processing time according to the filter size at a fixed image size. In this table it is clear that there is a big amount of processing time gain is achieved by applying this approach and this gain is increased rapidly with the increasing of filter size. The processing time gain started from 1.76 at filter size (11*11) and reached 57.24 at filter size (101*101).



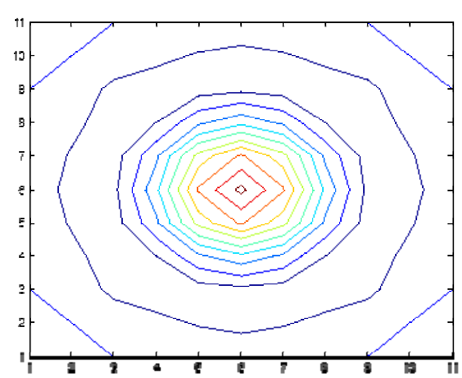
(A) Ideal Kaiser Window



(B) Filter Response



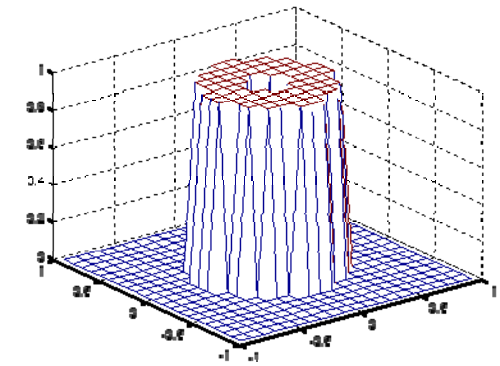
(C) Contour Of Kaiser Window



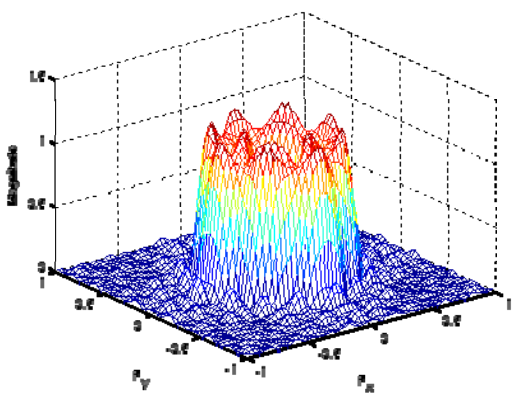
(D) Contour Of Filter Response

Figure (4) 11 Points Filter

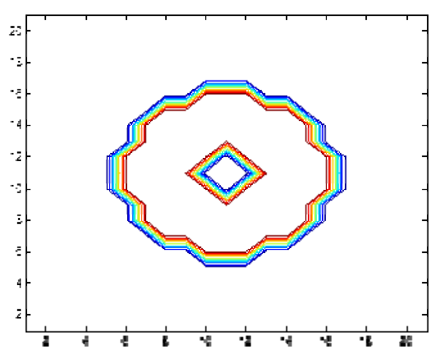
ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195



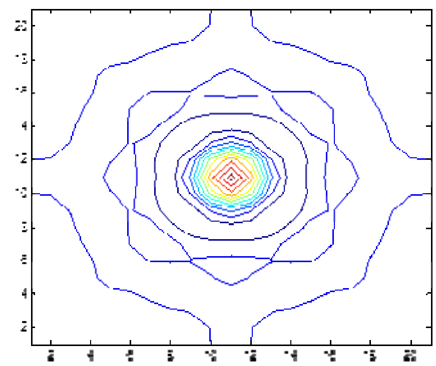
(A) Ideal Kaiser Window



(B) Filter Response

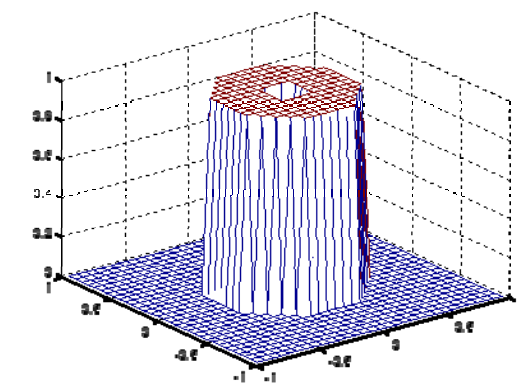


(C) Contour Of Kaiser Window

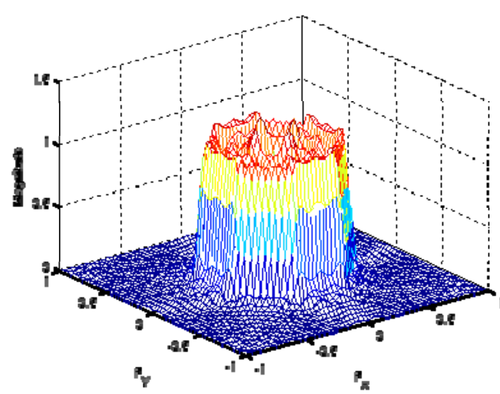


(D) Contour Of Filter Response

Figure (5) 21 Points Filter Design

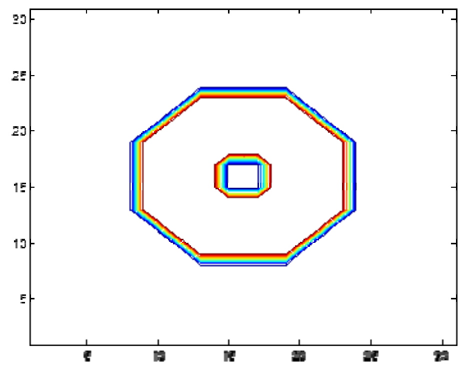


(A) Ideal Kaiser Window

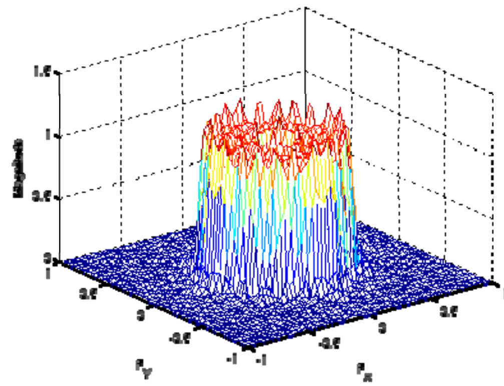


(B) Filter Response

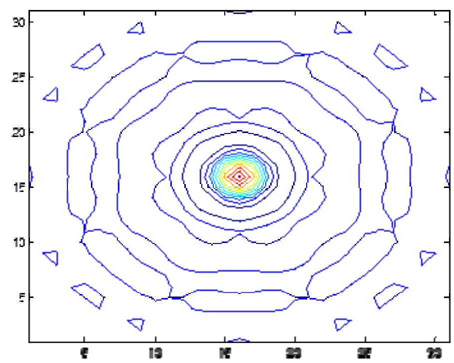
ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195



(C) Contour Of Kaiser Window

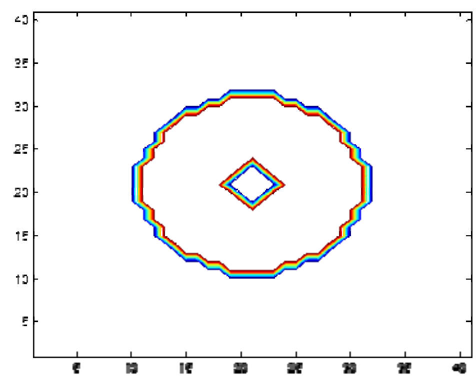


(B) Filter Response

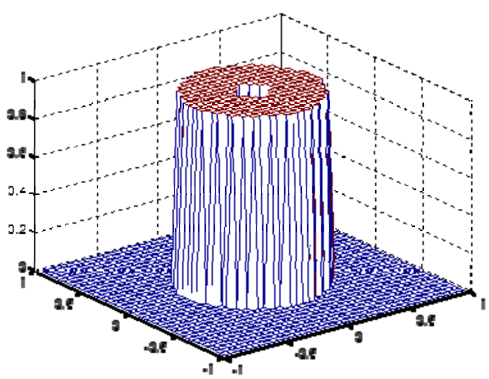


(D) Contour Of Filter Response

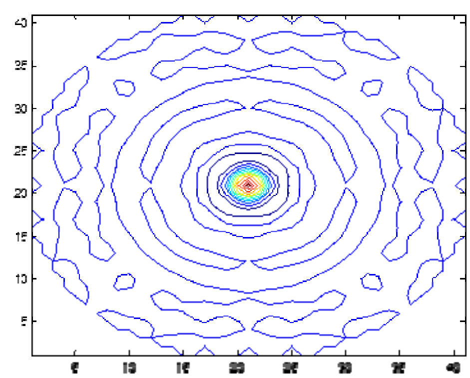
Figure (6) 31 Points Filter Design



(C) Contour Of Kaiser Window



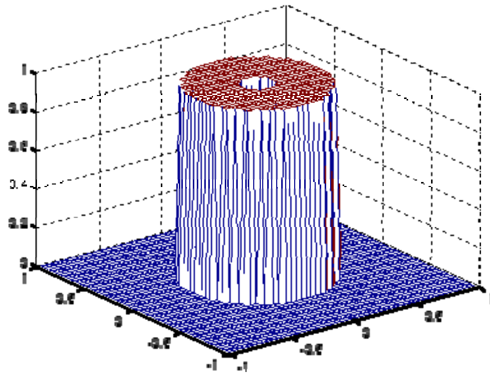
(A) Ideal Kaiser Window



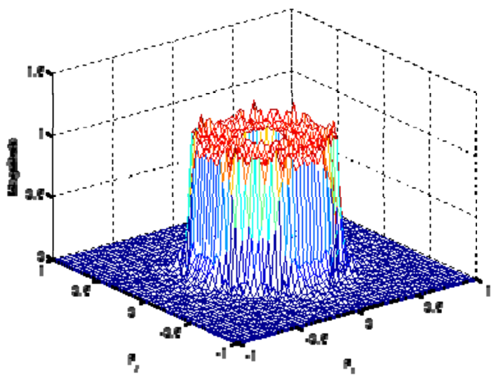
(D) Contour Of Filter Response

Figure (7) 41 Points Filter Design

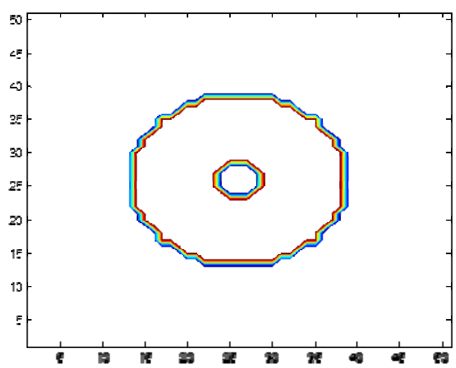
ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195



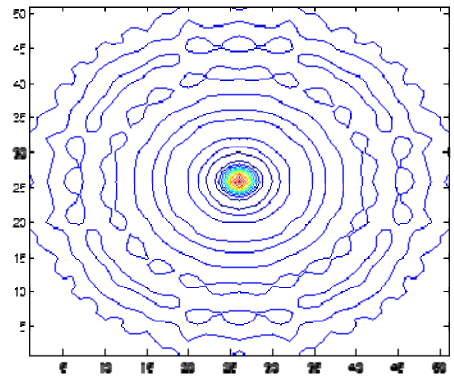
(A) Kaiser Window



(B) Filter Response



(C) Contour Of Kaiser Window

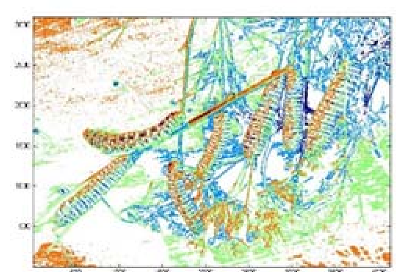


(D) Contour Of Filter Response

Figure (8) 51 Points Filter Design

(A) Original Image

(B) Filtered Image



(C) Contour Of Image

Figure (9) The Processing Of The Original Image

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

Table 4 Required Time Of The Implemented Approach

Image size	No. of pixels	Filter size	No. of operations via direct convolution
3096*4128	12780288	11*11	1.5464e+009
3096*4128	12780288	21*21	5.6361e+009
3096*4128	12780288	31*31	1.2282e+010
3096*4128	12780288	41*41	2.1484e+010
3096*4128	12780288	51*51	3.3242e+010
3096*4128	12780288	61*61	4.7555e+010
3096*4128	12780288	71*71	6.4425e+010
3096*4128	12780288	81*81	8.3851e+010
3096*4128	12780288	91*91	1.0583e+011
3096*4128	12780288	101*101	1.3037e+011
Image size	Time of direct convolution (seconds)	Time of implemented filter (seconds)	Gain
3096*4128	1.237521	0.701846	1.7632
3096*4128	2.397182	0.704784	3.4013
3096*4128	7.028692	0.707784	9.9306
3096*4128	8.463574	0.709784	11.924
3096*4128	11.146043	0.712574	15.642
3096*4128	15.802542	0.715574	22.084
3096*4128	21.182930	0.717574	29.520
3096*4128	27.211916	0.720574	37.764
3096*4128	33.399896	0.723574	46.159
3096*4128	41.646353	0.727574	57.240

7. CONCLUSIONS

Digital filter design plays an important part for their wide range of applications. Many methods and algorithms are implemented to achieve certain system performance. The implemented approach of digital filter based on based FFT applied in frequency domain. It is clear that filter design depends on many factors such as accuracy, details, transition band … etc. This approach is applied on various images in order to achieve an adequate performance of the system. The obtained result indicated that this approach gives good results of filtering in addition that achieve processing time. In addition the obtained gain in processing time is increased according to the increasing of filter size up to 57 times at a filter size of (101*101) that leads to good performance. The limitations may arise with the implementation of small size filter according to the speed factor. So in this case it is better to realize this approach via direct convolution.

REFRENCES:

[1] Z. J. Zhang, P. L. Shui, and T. Su, “Efficient design of high-complexity cosine modulated filter banks using 2Mth band conditions,” IEEE Transactions on Signal Processing, vol. 56, no. 11, pp. 5414-5426, 2008.

[2] X. P. Gao, B. D. Li, and F. Xiao, “Lattice Structure for Generalized-Support Multidimensional Linear Phase Perfect Reconstruction Filter Bank,” IEEE Trans. Image Process., vol. 22, no. 12, pp. 4853-4864, 2013.

[3] F. Zhou, J. Z. Jiang, and P. L. Shui, “ Fast design of 2D fully oversampled DFT modulated filter bank using Toeplitz-block Toeplitz matrix inversion,” Signal Process., vol. 111, pp. 194-198, 2015.

[4] J. Z. Jiang, and P. L. Shui, “Design of 2D oversampled linear phase DFT modulated filter banks via modified Newton's method,” Signal Processing, vol. 92, no. 6, pp. 1411-1421, 2012.

[5] P. L. Shui, “Image denoising using 2-D separable oversampled DFT modulated filter banks”, IET Image Processing, vol. 3, no. 3, pp. 163-173, 2009.

[6] P. L. Shui, and J. Z. Jiang, “Two-dimensional 2× Oversampled DFT Modulated Filter Banks and Critically Sampled Modified DFT Modulated Filter Banks,” IEEE Transactions on Signal Processing, vol. 58, no. 11, pp. 5597-5611, 2010.

[7] J. Z. Jiang, and P. L. Shui, “Design of 2D linear phase DFT modulated filter banks using bi-iterative second-order cone program,” Signal Processing, vol. 90, no. 12, pp. 3065-3077, 2010.

[8] Muzhir Shaban Al-Ani and Shokhan Mahmoud H, “Medical Image Enhancement Based on an Efficient Approach for Adaptive an Isotropic Diffusion”, International Journal of Advances in Engineering & Technology, May 2013, Vol. 6, Issue 3, pp. 1424-1430.

[9] Muzhir Shaban Mohammed Al-Ani, Munther n. Al-Tikriti and Kais Abdul-Nafi, "A Real Time Digital Signal Processing System", Journal of Engineering and Development, University of Al-Mustansiriya, Vol.2, No.3, Iraq, September 1998.

[10] Muzhir Shaban Al-Ani, “Hardware Implementation of a Real Time Image Compression”, IOSR Journal of Computer Engineering (IOSR-JCE), Volume 19, Issue 3, Ver. V (May - June 2017), PP 06-13.

ISSN: 1992-8645 www.jatit.org E-ISSN: 1817-3195

[11] Muzhir Shaban Mohammed Al-Ani, S. Lorenzo L. Nozal and B. Rui, "Fast Image Filtering Implementation", Second International Conference on Document Analysis and Recognition (ICDAR'93), 20-22 October 1993, Tsukuba Science City, Japan. In cooperation with the IEEE Computer Society and IGS.

[12] B. Rui, S. Lorenzo, L. Nozal and Muzhir Shaban Mohammed Al-Ani, "Digital Signal Processor Accelerator Board for Image Processing on VME Bus Based System", Machine Vision Application, Architectures, and System Integration 17-18 November 1992, Boston, Manssachusetts, USA, PP 85-90.

[13] Qeethara Al-Shayea and Muzhir Al-Ani, “Efficient Window Approach of FIR Filter Design (MSK2)”, IJCSNS International Journal of Computer Science and Network Security, Vol.16 No.2, February 2016.

[14] Muzhir Shaban Mohammed Al-Ani and Santiago Lorenzo, "System Implementation of LUT FFT", 6th Mediterranean Electrotechnical Conference (MELECON'91), 22-24 May, 1991, Yugoslavia.

[15] L. Nozal, S. Lorenzo, B. Rui and Muzhir Shaban Mohammed Al-Ani, “Pipe-Line Programmable Logic Device (PLD) A New Solution for Image Processing", SICE'91 Conference, 17-19 July, 1991, Yonezawa, Japan, PP 1097-1100.

[16] L. Nozal, S. Lorenzo, B. Rui and Muzhir Shaban Mohammed Al-Ani, "A New Vision System Programable Logic Devices Digital Signal Processor Architecture (PLD+DSP)", International Conference on Industrial Electronics Control and Instrumentation (IECON'91)", 28 October – 1 November, 1991, Japan, PP 2014-2018.

[17] L. Nozal, S. Lorenzo, B. Rui and Muzhir Shaban Mohammed Al-Ani, "Real Time and Low Cost Image Processing Architecture Based on Programmable Logic Device (PLD)", Intelligent for Mechanical System, Proceedings IROS '91, 3-5 November, 1991, Osaka, Japan, PP 279-284.

[18] Muzhir Shaban Mohammed Al-Ani, S. Lorenzo, and L. Nozal, "Fast 2D Convolution Filter Based on LUT FFT", IEEE International Symposium on Industrial Electronics, 25-27 May, 1992, Chain, PP 446-449.

[19] Muzhir Shaban Mohammed Al-Ani, S. Lorenzo L. Nozal and B. Rui, "FFT LUT for Image Processing", The International Society for Optical Engineering (SPIE'92), Section of Algorithms Technique and Active Vision, 15-20 November 1992, USA, PP 121-129.

[20] L. Nozal, S. Lorenzo, B. Rui and Muzhir Shaban Mohammed Al-Ani, "Hardware Structure + Digital Signal Processing on Real Time", International Conference on Industrial Electronics, Control, Instrumentation and Automation, 9-13 November 1992, San Diego, California, USA, PP 1397-1402.

[21] Muzhir Shaban Mohammed Al-Ani, "A Real Time Image Processing System Architecture (EPLD + DSP), Journal of Engineering and Development, University of Al-Mustansiriya, Iraq, November 1998.

[22] Ghazi Ibrahem Raho, Ali Jbaeer Dawood and Muzhir Shaban Al-Ani, “Real Time Fast Algorithm of 2D DWT Based DSP Technology”, International Journal of Application or Innovation in Engineering & Management, Volume 2, Issue 10, October 2013.

[23] V. Rajaravivarma, P.K. Rajan and H.C. Reddy, “Design of multidimensional FIR digital filters using symmetrical decomposition technique,” IEEE Transactions on Signal Processing, vol. 42, no. 1, pp. 164–174, Jan. ‘94.

[24] H.C. Reddy, P.K. Rajan, G.S. Moschytz and A.R. Stubberud, “Study of various symmetries in the frequency response of two-dimensional delta operator formulated discrete-time systems,” Proc. 1996 IEEE-ISCAS, vol. 2, pp. 344–347, May 1996.

[25] H.C. Reddy, I.H. Khoo, P.K. Rajan, “Symmetry in the frequency response of two-dimensional delta operator formulated discrete-time systems,” Proc. the 1997 ECCTD, vol. 3, pp. 1118–1123, Aug 1997.

[26] I.H. Khoo, H.C. Reddy and P.K. Rajan, “Delta operator based 2-D filter design using symmetry constraints,” Proc. 2001 IEEE-ISCAS, vol. 2, pp.781–784, May 2001.

[27] I.H. Khoo, H.C. Reddy and P.K. Rajan, “Delta operator based 2-D filters: Symmetry, stability, and design,” Proc. 2003 IEEE-ISCAS, May 2003.

[28] Yu Wang, Bin Li , Yunbi Chen, “Digital IIR filter design using multi-objective optimization evolutionary algorithm”, Applied Soft Computing 11 (2011) 1851–1857.

[29] Mohammad Shams Esfand Abadi, Sahar Nikbakht Aali, “The novel two-dimensional adaptive filter algorithms with the performance analysis”, Signal Processing103 (2014) 348–366.

[30] Andrzej Handkiewicz, Piotr Katarzyn ski, Szymon Szczesny, Mariusz Naumowicz, Michał Melosik, Paweł Sniatała, Marek Kropidłowski "Design automation of a lossless multiport network and its application to image filtering", Expert Systems with Applications 41 (2014) 2211–2221.

[31] Jun-Zheng Jiang, Fang Zhou, Peng-Lang Shui, Shan Ouyang, "Theory and design of two-dimensional DFT modulated filter bank with arbitrary modulation and decimation matrices", Digital Signal Processing, Volume 44, September 2015, Pages 123-130.

[32] Abhijit Chandra, Sudipta Chattopadhyay, "Design of hardware efficient FIR filter: A review of the state-of-the-art approaches", Engineering Science and Technology, an International Journal, Volume 19, Issue 1, March 2016, Pages 212-226.

[33] Akhilesh Gotmare, Sankha Subhra Bhattacharjee, Rohan Patidar, Nithin V. George, "Swarm and evolutionary computing algorithms for system identification and filter design: A comprehensive review", Swarm and Evolutionary Computation, Volume 32, February 2017, Pages 68-84.

[34] T. Bindima, Elizabeth Elias, "A novel design and implementation technique for low complexity variable digital filters using multi-objective artificial bee colony optimization and a minimal spanning tree approach", Engineering Applications of Artificial Intelligence, 59 (2017) 133–147.

[35] Judhisthir Dash, Bivas Dam, Rajkishore Swain, "Design of multipurpose digital FIR double-band filter using hybrid firefly differential evolution algorithm", Applied Soft Computing, Volume 59, October 2017, Pages 529-545.