

Digital Filter Design of Infinite Impulse Response (IIR) Infrasound to Detect Fetal Heart Rate

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Abstract— One way of checking on a pregnant woman is inspection using ultrasonic wave Doppler technique, this check is done by putting Doppler ultrasonic waves over the abdomen to capture the heart beat of the fetus with the addition of the gel material, there sebaga in pregnant women who are allergic to the gel material. Alternatively we can use the technique in frassonik waves. With a digital design II Chebyshev band pass filter with a frequency range of 2-3Hz, the heart rate of pregnant women with fetal heart beat can be distinguished. With the help of software MatLab and Xilinx will be resolved in the manufacture of digital filter terser but. The simulation results with the software quite satisfactory.

Keywords— *heartbeat, infrasound.*

I. INTRODUCTION

1.1 Background Research

Since the beginning of pregnancy, it is advisable to conduct regular control to the nearest obstetrician. During the inspection control is highly dependent on the doctor or clinic in question, but in general here is given the type of examinations performed and it is necessary to ensure the health of mother and baby during pregnancy to child birth. One of these checks is the examination with Doppler ultrasound, this check is done by putting Doppler ultrasonic waves over the abdomen to capture the heart beat of the fetus with the addition of the gel material, there sebaga in pregnant women who are allergic to the gel material.

1.2 Some researchers History:

[G.S. Dawes, G.H.A. Visser, J.D.S. Goodman, C.W.G. Redman: 1981] ultrasonic sensors are used, compared to ultrasound-based technique with electro cardiography A direct and noted that the short-term index calculation error(STV) variability(calculated as the beat-to-beat changes) caused by the approach of ultrasoundreaches100%.

[G.W. Lawson, G.S. Dawes, C.W.G. Redman: 1982] uses ultrasonic sensors to do the same, but for other types of fetal monitor, download the error log many of the calculations (200%). In both cases the signal is recorded using a fetal monitor from generation before hand, which implement peak detection method to identify fetal heart rate. In further experiments, using a fetal monitor with the new generation of the auto correlation function.

[G.W. Lawson, R. Belcher, G.S. Dawes, C.W.G. Redman : 1983] gained a lot of errors is low on TV ratings: -35%. A minus sign means that the pen-ultrasonic approach decreased short-term variability in relation to the FECG.

[J. Wrobel, J. Jezewski, Dawid Roj, Tomasz P., Robert C., Adam : 2010]Estimated instantaneous reliability of the fetal heart rate (FHR) is influenced by autocorrelation techniques commonly used in the channel Doppler ultrasound fetal monitor today. Has noted greatly reduce the short-term variability, which is quite surprising because as has been shown previously, determine fetal monitor FHR signal with satisfactory accuracy in relation to electro cardiography direct reference to the fetus. The purpose of this study was to identify the source kesala-han caused by a commonly used approach. Results may be possible to developed the method for the correction of FHR variability describes the quantitative index for certain types of fetal monitor, which relies on the proposed cancellation of the constant error correction component, which has been assigned to an average properties of the autocorrelation function. Although the random error component remaining after correction is still not very satisfactory considering the instantaneous values, significanting katan Pen-track assessment of the reliability of the measurement of fetal heart rate variability in global cases confirmed one hour

Fetal heart beat sound (relatively weak), then actually using a sensitive microphone, the fetal heart can be detected by strengthening the signal generated by the microphone. The problem is no fetal heart beat in addition there is also another's heart beat that results in a stronger heart beat intensity. The frequency of fetal heart rate is higher than the heart beat of the

mother (fetal heart rate is 120-160beats per minute (about2-3Hz) while the heart of the mother around 80-90 beats per minute), then by technical filtering, then the heart beat the mother can be filtered in order to obtain only the fetal heartbeat.

Therefore, through this study, researchers will propose to design a digital filter (band pass filter) using approaches in frasound (2-3 Hz) are not widely used to monitor fetal heart beat rate, using Simulink and XiLink FPGA tool.

II. LITERATURE REVIEW

A filter is a circuit that is designed to skip a frequency band ter-course while weakening all signals outside this band. The digital filter is an implementation of the algo-Ritma mathematics into the hardware and / or software that operates on an input digital signal for the purpose of filtering is achieved. Jarin-gan-network regular filter is active or passive. Tissues of passive filters contain only prisoners, inductor, and capacitor alone. Active filters menggunakan transistor or op-amp plus tahanan, inductors, and capacitors. But ja-rang inductors used in filters are active, because uku-rannya large and expensive and could have Kompo-nen-component in a large persistence. With Filter we can filter out the frequency we want, there are several types of filters. As we know, based on the frequency that is passed, in general filter terbagi into 4 yak-ni Low Pass Filter; High Pass Filter; Band Pass Filter; and Band Stop Filter. IIR filters have infinite impulse response filter where the output is a function of input current conditions, previous input and output at an earlier time (causal), better known as recursive filters that involve pros-ice feedback and feed forward. For this study, researchers used a type of IIR digital filter. IIR filter is one type of digital filters are widely used in applications of Digital Signal Processing (DSP). IIR stands for Infinite Impulse Response. Why is it called the re-sponge infinite impulse (infinite)? Due to the feedback in the filter, if you are me-enter an impulse (ie a signal '1' followed by many signals '0'), then on his output will continue to oscillate for their feedback, although in practice it will be lost on someday. Advantages include IIR filter-kan is a need for fewer coefficient for frequency response-uensi steep so as to reduce the amount of computation time. In general, the transfer function of the digital filter IIR (Infinite Impulse Response) Order N [Smith, 1995] are as follows:

$$H(z) = \frac{B(z)}{A(z)} = \frac{b_0 + b_1 Z^{-1} + \dots + b_N Z^{-N}}{1 + a_1 Z^{-1} + \dots + a_N Z^{-N}} \quad (1)$$

The equation is different is expressed as the equation:

$$Y_k = \sum_{n=0}^L b_n x_{k-n} - \sum_{n=1}^L a_n y_{k-n} \quad (2)$$

Input-output equation IIR filter is as follows:

$$\begin{aligned} y(n) = & a_0 x(n) + a_1 x(n-1) + \dots \\ & + a_{M-1} x(n-(M-1)) \\ & - b_1 y(n-1) - b_2 y(n-2) - \dots \\ & - b_N y(n-(n-N)) \end{aligned} \quad (3)$$

Based IIR filter design techniques can be categorized in several design techniques. While the filters FIR (Finite Impulse Response) basically have two methods of design, namely the window (the window) and pitajamak the transition band (multiband with transition bands). IIR digital filter design techniques are based on bilinear transformation from analog transfer function prototype. Ana-log transfer function is typically one of the type of transfer function: But-terworth, Chebyshev Type 1, Type 2 Chebyshev and Elliptic (or Cauer). The difference between these filter types can be explained by looking at the lower pass filter (lowpass) analogous to those described below [Antoni-ou, 1993].

- Butterworth transfer function has escaped under maximum flat magnitude response and magnitude response berku-rang horizontally with air-tarnbah frequency.
- The transfer function passes under Chebyshev Type I had a magnitude responsive to the ri-ak equilibrium on tape passes and a magnitude response is reduced horizontally with increasing frequency in the outer tape passes.

- c. The transfer function passes under Chebyshev Type II has a magnitude response reduced horizontally on tape passes with increasing frequency and magnitude response ripple equilibrium pad prevent a tape.
- d. The transfer function passes under Elliptic mempu-nyai ripple magnitude response of equilibrium on the second tape, ribbon or ribbon prevent escapes.

2.1 Advantages of Digital Filters: 1

1. The working area is wide (can be frequency for low and high) with a sampling frequency menga-tour. It can even be used in areas Very Low Frequency, where the analog filter is difficult to apply in this frequency area, so it can be used for applications elektroni-ka biomedical.
2. Can provide phase response is absolutely linear.
3. Because the digital filter implementations can use the programmable pro-cessor, the frequency response can be changed automatically.
4. Some of the input signal (several chan-nel) can be filtered by a digital filter without doubling the hardware for each channel.
5. With the development of technology and fab-rikasi VLSI chip, power consumption and smaller digital filter can reduce the cost of production.
6. Advantages include IIR filter requires fewer coefficients for steep frequency response so as to reduce the amount of computation time.
7. In the Order IIR filter low frequency response that can produce sharp / steep (almost close to ideal).
8. The digital filter is very stable against the effects of temperature and time.
9. The digital filter can handle signals with low fre-keunsi accurate.

2.2 Digital Filter Chebyshev

Chebyshev filter is a part of the analog or digital type filter that has a sharper roll-off and more ripple in the passband (type I) or stopband ripple (type II) than the Butterworth filter. Chebyshev filter has the property to minimize the error between the ideal and the actual characteristics of the filter for the filter range, but the ripple in the passband. This type of filter is named in honor of a scientist named Pafnuty Chebyshev, due to their mathematical characteristics derived from the Chebyshev polynomials. Due to the passband ripple inherent in Chebyshev filter, a filter that has a smooth response in the passband but a more irregular response in the stopband are preferred for some applications [Williams, 2006]. To find or search order (order) n on Chebyshev, use the following equation

$$n = \frac{\log \left[\frac{10^{\alpha_{min}/10} - 1}{10^{\alpha_{max}/10} - 1} \right]}{2 \log \left(\frac{f_{stop}}{f_p} \right)} \quad (4)$$

$$\epsilon = \sqrt{10^{\alpha_{max}/10} - 1} \quad (5)$$

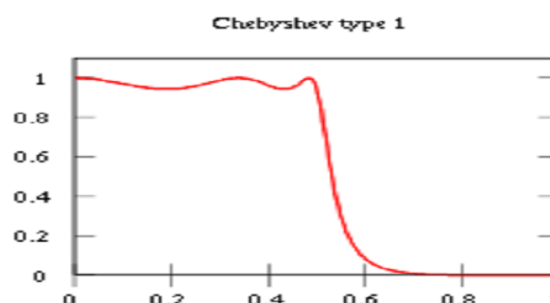


FIGURE. 1. CHEBYSHEV TYPE I LOWPASS FILTER [IFEACHOR, 2021]

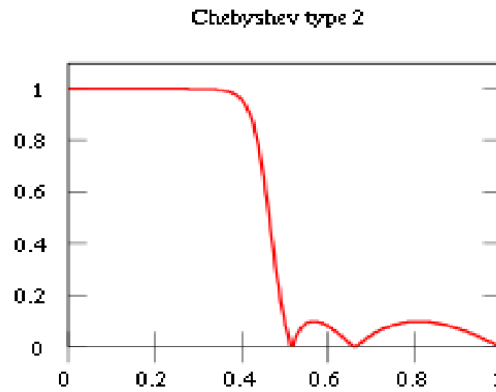


FIGURE. 2. CHEBYSHEV TYPE II LOWPASS FILTER [IFEACHOR, 202]

2.3 Form of Direct Form

The formulas can be cited in the direct form of mathematical equations as below:

$$H(z) = \frac{Y(z)}{X(z)} = \frac{(a_0 + a_1 z^{-1} + a_2 z^{-2} + \dots + a_k z^{-k})}{(1 + b_1 z^{-1} + b_2 z^{-2} + \dots + b_k z^{-k})}$$

$$H(z) = \frac{\sum_{i=0}^k a_i z^{-i}}{1 + \sum_{j=1}^k b_j z^{-j}}$$

2.4 Direct Form I

Direct approaches to the realization of the IIR filter Direct Form I, where the difference equation is evaluated directly. This form is practical for small filter, but may not be efficient and not practical (numerically stable) to design intricate or complex [If each or, 2002]. In general, this takes the form of the element to delay (delay) 2N (for signal input and output signals) for the filter of order N.

Direct Form I to its realization can directly use equations quoted as below:

$$y(n) = A = \sum_{i=0}^k a_i(xn - i) - \sum_{j=1}^k a_j(yn - j) \quad (6)$$

From this equation, we can construct a diagram as shown in Figure 3 and Figure 4

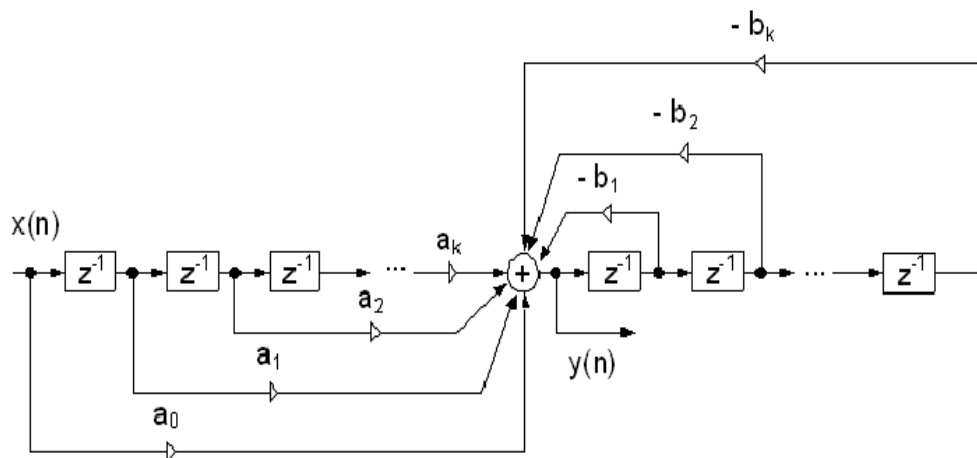


FIGURE.3.REALIZATIONOF THE DIRECT FORM I

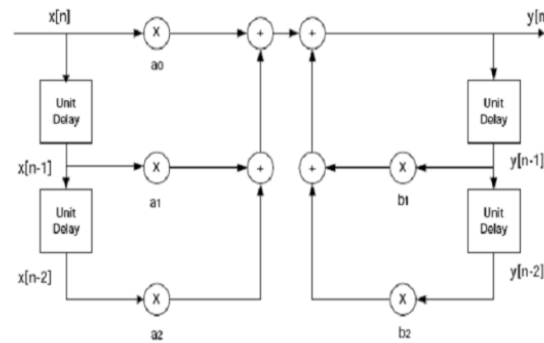


FIGURE. 4. SCHEMATIC DIAGRAM OF AN IIR FILTER

2.5 Direct Form II

Alternatively Direct Form II only requires N delay units, where N is the order of the filter, could potentially have as many as half a Direct Form I. This structure is obtained by reversing the order of the numerator and denominator of parts Direct Form I, because the two systems are actually linear, and commutative apply for the property. Kemudian, one will see that there are two columns of delay (delay) ie z^{-1} utilizing the center nets, and these can be combined because there were a lot of repeats. The disadvantage is that Direct Form II increases the chances of arithmetic (calculation) over-flow filter or a high Q resonance [Ruggiero, 1982]. It has been proven that Q increases, the noise around these two forms of topology increases indefinitely. This is because, conceptually first signal passes through the filter all poles (which can usually improve gain at the resonance frequency) before generating saturation, then pass through a filter that is all zero (which often weakens as many as half of all poles bracing). Realizations of Direct Form II can be written in mathematical terms:

$$Y(z) = H(z) \cdot X(z)$$

$$H(z) = \frac{N(z)}{D(z)}$$

$$Y(z) = \frac{N(z)}{D(z)} \cdot X(z) = \frac{X(z)}{D(z)} \cdot N(z)$$

$$W(z) = \frac{X(z)}{D(z)}$$

$$Y(z) = N(z) \cdot W(z)$$

$$W(z) = \frac{X(z)}{D(z)} = \frac{X(z)}{(1 + b_1 z^{-1} + \dots + b_k z^{-k})} \quad (7)$$

$$(1 + b_1 z^{-1} + \dots + b_k z^{-k}) \cdot W(z) = X(z)$$

$$w(n) + b_1(w(n-1) + \dots + b_k(w(n-k))) = x(n)$$

$$w(n) = x(n) - b_1(w(n-1) + \dots + b_k(w(n-k)))$$

$$w(n) = x(n) - \sum_{j=1}^k b_j(w(n-j))$$

$$Y(z) = N(z) \cdot W(z)$$

$$Y(z) = (a_0 + a_1 z^{-1} + \dots + a_k z^{-k}) \cdot W(z)$$

$$y(n) = a_0(w(n)) + a_1(w(n-1)) + \dots + a_k(w(n-k))$$

$$y(n) = \sum_{i=0}^k a_i(w^n - i)$$

From the equation above, we can construct a diagram as in Figure6 below:

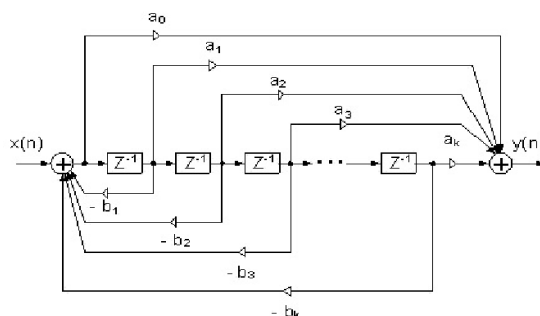


FIGURE5. REALIZATION OF THE DIRECT FORM II

2.6 Research Objectives

The specific objectives of this program are:

1. Creating a digital filter design type of Band Pass Filters IIR Chebyshev II order 4 so that it can be implemented as an on the FPGA board.
2. Realizing a digital infrasonic filter that can later be used for development, detection of fetal heart beat in the form of SoC (System on Chip).
3. Program MATLAB is used to determine the coefficient digital filter design.
4. Program Xilinx is used for the simulation of digital filter design MatLab and to match the results.

2.7 Research Methods

2.7.1 Digital Filter Design

This section discusses the design, designing band pass Chebyshev digital filter by performing mathematical calculations. To be implemented in a XilinxIse14.2 with VHDL is a hardware description language standard for VHSIC. Hardware description language used in electronic design automation to describe digital and mixed-signal systems such as the field programmable gate array and integrated circuits. Block diagram of the overall system can be described as below:

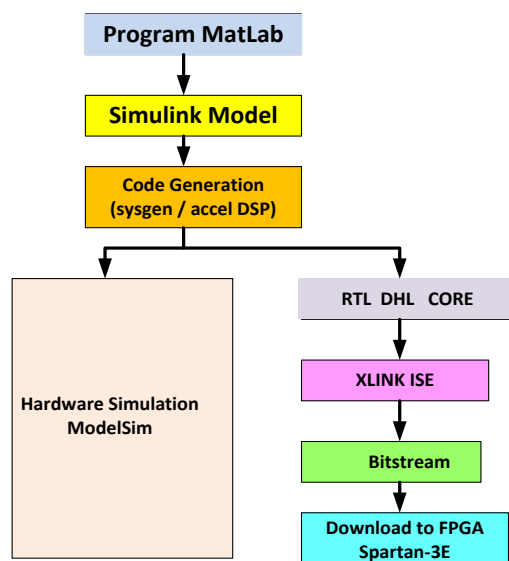


FIGURE 6. BLOCK DIAGRAM GLOBAL RESEARCH

The location is on the pole Butterworth filter and the Chebyshev polynomial coefficients appropriate and have been obtained and tabulated by Weinberg (see Table 1).

An abbreviated set of coefficients polynomial for Butterworth and Chebyshev functions presented here, some form of polynomial factors have been calculated and tabulated. Polynomial given in the foundation needed to build a low-pass transfer function. The general form of the transfer functions as in the table below:

TABLE1.
THE COEFFICIENT OF LOW-PASS FILTER POLYNOMIAL DENOMINATOR (DENOMINATOR POLYNOMIALS)

Order	B0	B1	B2	B3	B4	B5
BUTTERWORTH						
1	1	1				
2	1	1.414	1			
3	1	2	2	1		
4	1	2.613	3.414	2.613	1	
5	1	3.236	5.236	5.236	3.236	1
CHEBYSHEV 0.5 DB RIPPLE						
1	2.863	1				
2	1.516	1.425	1			
3	0.716	1.535	1.253	1		
4	0.379	1.025	1.717	1.197	1	
5	0.179	0.753	1.309	1.937	1.174	1
CHEBYSHEV 1 DB RIPPLE						
1	1.965	1				
2	1.103	1.098	1			
3	0.491	1.238	0.988	1		
4	0.276	0.743	1.454	0.953	1	
5	0.123	0.581	0.974	1.689	0.937	1
CHEBYSHEV 2 DB RIPPLE						
1	1.308	1				
2	0.823	0.804	1			
3	0.327	1.022	0.738	1		
4	0.206	0.517	1.256	0.716	1	
5	0.082	0.459	0.693	1.499	0.706	1
CHEBYSHEV 3 DB RIPPLE						
1	1.002	1				
2	0.708	0.645	1			
3	0.251	0.928	0.597	1		
4	0.177	0.405	1.169	0.582	1	
5	0.063	0.408	0.549	1.415	0.574	1

$$G(s) = \frac{A_0}{(B_0 + B_1s + B_1s^2 \dots + B_ks^k)} \quad (8)$$

The coefficient of the denominator of the equation (8) are tabulated in Table to function for the Butterworth and Chebyshev functions to four possible values of passband ripple (0.5 dB, 1 dB, 2 dB and 3 dB). The use of the filter order from $k = 1$ to $k = 5$. If desired gain in dc numerator constant then have $A_0 = B_0$ (Note: the coefficient is defined according to the equation (8) (6)).

Creating a Digital Design Band-Pass Filter Chebyshev. To be able to operate on digital signal processing systems must have a sampling rate of at least 2 x frequency data in accordance with the terms Nyquist, in this design used 200 Hz (fs). Bandpass digital filter required specification, the system as follows:

- Range band pass filter of 2 Hz to 3 Hz with attenuation that is still allowed no more than 3 dB at a frequency of two sides of the ramp.
- Attenuation should be less than 18 dB at the frequency of 1 Hz and 4 Hz.
- The type of response required by the ripple Chebyshev 3 dB.
- Looking folding frequency (f_0) is $f_0 = f_s / 2 = 200/2 = 100$ Hz. There are two frequency of 2 Hz and 3 Hz to be selected as a band-edge frequency as f_1 and f_3 . For search, the values of the main and sub prime rates, the frequency will be changed to a form of normalization,

$$v_1 = f_1/f_0 = 2/100 = 0.02 \quad (9)$$

$$v_3 = f_3/f_0 = 3/100 = 0.03 \quad (10)$$

$$b = v_3 - v_1 = 0.03 - 0.02 = 0.01 \quad (11)$$

$$v_a = f_{r1}/f_0 = 1/100 = 0.01 \quad (12)$$

$$v_b = f_{r2}/f_0 = 4/100 = 0.04 \quad (13)$$

The center frequencies v_2 are:

$$\begin{aligned} \tan^2 \frac{\pi}{2} v_2 &= \tan \frac{\pi}{2} v_1 \tan \frac{\pi}{2} v_3 = \tan \frac{\pi}{2} \times 0.02 \tan \frac{\pi}{2} \times 0.03 \\ &= \tan (1.8) \tan (2.7) \\ &= 0.0314 \times 0.0472 \\ &= 0.0015 \\ \tan \frac{\pi}{2} v_2 &= 0.0385 \\ \frac{\pi}{2} v_2 &= 2.2048 \\ v_2 &= 0.0245 \quad (14) \end{aligned}$$

Transformation for the constants D and E with the assumption that $\lambda_r = 1$ rad/s.

$$D = \lambda_r \cot \frac{\pi B}{2f_0} = \lambda_r \cot \frac{\pi b}{2f_0} = \cot 0.9 = 63.6537$$

So to λ_a and λ_b representing two analog frequencies, are as follows:

$$\begin{aligned} \frac{\lambda_a}{D} &= \frac{\cos \pi v_2 - \cos \pi v_a}{\sin \pi v_a} \\ \lambda_a &= D \frac{\cos \pi v_2 - \cos \pi v_a}{\sin \pi v_a} \\ \lambda_a &= 63.6537 \frac{\cos \pi 0.0245 - \cos \pi 0.01}{\sin \pi 0.01} \\ \lambda_a &= 63.6537 \frac{\cos (4.41) - \cos (1.8)}{\sin (1.8)} \\ \lambda_a &= 63.6537 \frac{(0.997 - 0.999)}{0.0314} \end{aligned}$$

$$\lambda_a = -4.0543 \quad (17)$$

$$\lambda_b = D \frac{\cos \pi v_2 - \cos \pi v_b}{\sin \pi v_b}$$

$$\lambda_b = 63.6537 \frac{\cos \pi 0.0245 - \cos \pi 0.04}{\sin \pi 0.04}$$

$$\lambda_b = 63.6537 \frac{\cos (4.41) - \cos (7.2)}{\sin (7.2)}$$

$$\lambda_b = 63.6537 \frac{(0.997 - 0.992)}{0.1253}$$

$$\lambda_b = 2.54 \quad (18)$$

The transfer function of the analog normalized second-order Chebyshev filter with a 3dB ripple of Table 2, with the use of dummy variables p , that function becomes as below:

$$G(p) = \frac{0.708}{0.708 + 0.645 p + p^2}$$

Transformation needed

$$p = D \left[\frac{(1 - Ez^{-1} + z^{-2})}{(1 - z^{-2})} \right]$$

$$p = 63.6537 \left[\frac{(1 - 0.92 z^{-1} + z^{-2})}{(1 - z^{-2})} \right]$$

Substituting (20) into the equation(19),

$$H(z) = G(p) \quad p = D \left[\frac{(1 - Ez^{-1} + z^{-2})}{(1 - z^{-2})} \right]$$

$$H(z) = G(p) \quad p = 63.6537 \left[\frac{(1 - 0.92 z^{-1} + z^{-2})}{(1 - z^{-2})} \right]$$

$$H(z) = \frac{0.708}{0.708 + 0.645 p + p^2} \quad p = 63.6537 \left[\frac{(1 - 0.92 z^{-1} + z^{-2})}{(1 - z^{-2})} \right] \quad (21)$$

$$H(z) = \frac{0.708 (1 - z^{-2}) z^{-2}}{0.708 (1 - z^{-2}) z^{-2} + 41,054 (1 - z^{-2}) + 4051,3225 (1 - 0.92 z^{-1} + z^{-2})^2} \quad (22)$$

$$H(z) = \frac{0.00017(1 - z^{-2})z^{-2}}{1 - 1,8123 z^{-1} + 2,7894 z^{-2} - 1,7940 z^{-3} + 0,96027 z^{-4}} \quad (23)$$

the coefficient of the equation (23) are as follows:

TABLE 2
VALUE COEFFICIENT OF EQUATION (23)

No	Koefisien	Nilai	Desimal	Hexadesimal
1	A0	0.00017	17	0011
2	A2	-0.00034	-34	FFDE
3	A4	0.00017	17	0011
4	B1	-1,8123	-181230	D3C12
5	B2	2,7894	278940	4419C
6	B3	-1,7940	-179400	D4338
7	B4	0.96027	96027	1771B

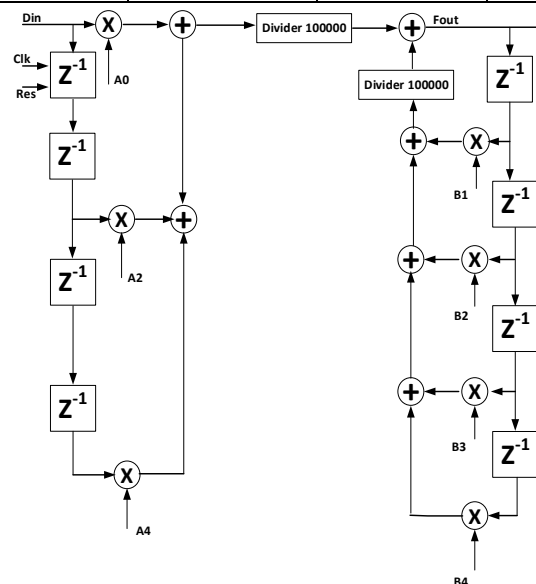


FIGURE 7. REALIZATION FOR SYSTEMS OF EQUATIONS23

2.7.2 IIR filter Deployment Process

Analog input signal must first sample and digitally using ADC(analog-to-digital con-verter). The resulting binary numbers, representing the values of consecutive samples of the input signal, will be transferred to the processors, which perform numerical calculations on them. These calculations usually involving mathematical operations by downloading galikan input values by a constant and Menam-even the same product. If necessary, result-Tungan perhi, who now represents the values of the samples of the filtered signal, is output through a DAC (digital to analog converter) to convert the signal back to its original analog form.

Modules are used for implements its-invest the IIR filter is as follows: Module Multiplication, Addition module, splitter modules, and modules Reduction, that all of it was developed using the programming language VHDL.

2.7.3 Digital Filter Design with MatLab Simulink

Band-Pass Filter: The specification of the filter as below:

- Response Type: Band-Pass
- Design Method: IIR Chebyshev
- Filter Order: 4th
- Frequency specification: F_s (200 Hz), F_{stop1} (2 Hz), F_{stop2} (3 Hz)
- Magnitude Specification: 60 dB (A_{stop})

The test results with MatLab program with the input of order 2 are required. Order test follows the tables on MatLab program with input from 2, 4, 6, ..., 20.

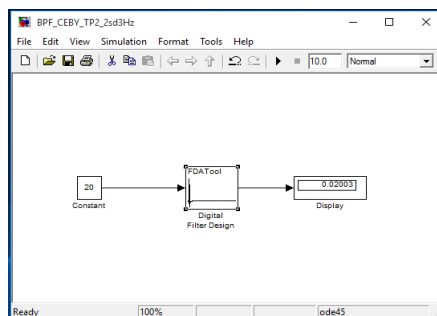


FIGURE 8.HOW TO DETERMINE THE VALUE $Y(n)$.

TABLE3.
SIMULATION RESULTS BAND-PASS FILTER WITH SIMULINK

No	$X(n)$	$Y(n)$ Simulink
1	2	0,00200
2	4	0,00400
3	6	0,00600
4	8	0,00801
5	10	0,01001
6	12	0,01202
7	14	0,01402
8	16	0,01602
9	18	0,01803
10	20	0,02003

The simulation results using MatLab with multiple inputs of different frequencies, among others:

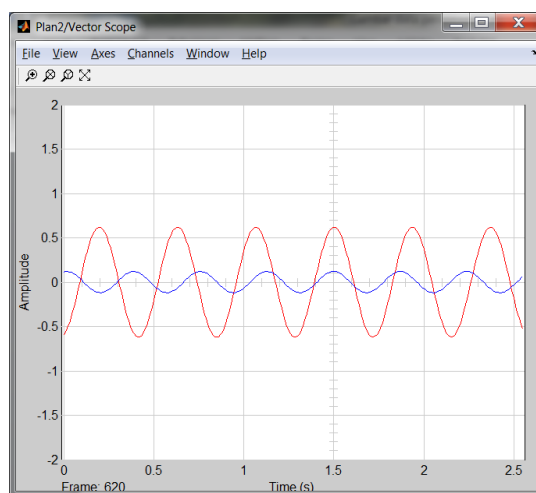


Figure 9.Input Frequency 2.3-2.7Hz

2.7.4 Implementation of Simulation-Based FPGA Digital Filter

Band-Pass Filter: coefficient that has been designed (calculated) has been given in the equation above. Now this VHDL code used to rate circuit with the use of tool SINTESI Xlink and simulated using the simulator Xlink with the input sequence (2, 4, 6, ..., 20). At this stage the researchers used a type of program Xlink with Spartan-3E, according to the FPGA board that had been prepared.

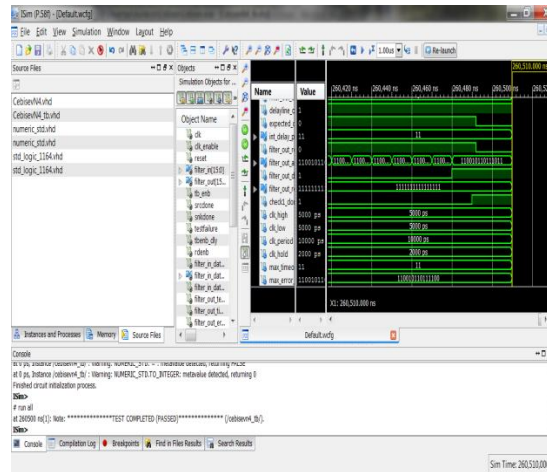


FIGURE 11.SIMULATION RESULTS IN FPGA

2.7.5 Analysis and Discussion

At the time of testing or simulation is no occurrence of errors or differences in results between MatLab programs with Xilinx. The incidence of errors is limited to the types of data types Xilinx, about fractions, as in Table4.

TABLE 4.
SIMULATION RESULTS BAND-PASS FILTER BASED ON SIMULINK WITH XILINK

No	X(n)	Y(n) Simulink	Y ₂ (n)Xilinx
1	2	0,00200	0
2	4	0,00400	0
3	6	0,00600	0
4	8	0,00801	0
5	10	0,01001	-1
6	12	0,01202	-1
7	14	0,01402	-1
8	16	0,01602	-1
9	18	0,01803	-1
10	20	0,02003	-1

Band-Pass circuit design has been completed using VHDL and implementation da-lam Xilinx Spartan-3E (package: FG320, speed: -5), in the process of making this design using theXilinxISE14.6 design tools. The tables use the following resources for Spartan-3E on the design and implementation of digital filters lawyer-data rule,

TABLE 6.
ADVANCED HDL SYNTHESIS REPORT

Circuit	Total multipliers	Total Addresses	Total Registers (Flip – Flop)
Low Pass	4	5	64
Band Pass	6	7	256
High Pass	4	5	64

III. DISCUSSION

The minimum period for a bandpass filter is 1.384 ns (maximum frequency: 722 648 MHz). Number 4 input LUTs for bandpass filter is 209. The total number of gates use in the design of implementation is 2408 to bandpass filter.

IV. CONCLUSION

Designs of Digital IIR Bandpass Filters have been achieved. A number of findings have identified during the design and simulation phases, as follows:

1. IIR filter can be implemented based FPGAs.
2. The coefficient of the IIR filter design has been done by calculation. VHDL has been used to get into the hardware description. VHDL code has been written, synthesized, mapped and then successfully configured with a prototype. IIR filter is designed completely in accordance with the design requirements.
3. In testing the filter circuit is still a difference between the results of Matlab simulation program to program Xilinx. This is influenced by the use of dividers components that perform the division process with the result of rounding.
4. One of the problems in the process simulation program Xilinx is the use of integer data, where data from the fraction should rounding first.
5. The results of the FPGA board to produce the same output analog signals with analog signal input. So that this algorithm is correct and appropriate, the file *.ucf can be used to make chips infrasound.

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