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Bandpass Filter Application with IIR (Infinite Impulse Response)

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ABSTRACT: A device that can pass the desired signal frequency and suppress unwanted frequencies is called a filter. A filter is a circuit device or program that functions to process the frequency of a signal. In digital signal processing for designing digital filters, analog filters are needed as the basis. The IIR digital filter design uses the transfer function of the $H(s)$ analog filter. The analog filter transfer function is then reduced to the transfer function of the $H(z)$ digital filter. The IIR filter application uses the bilinear transformation method by utilizing a Butterworth and Chebyshev analog filter prototype.

The method used in this research is to determine the appropriate digital filter coefficient of the bilinear transformation. The results of the digital filter coefficients obtained are described as the frequency response of each filter by setting the order (n) of the Butterworth and Chebyshev bandpass filters. The results of the order (n) variations observed that the numerator and denominator coefficients are very compatible so that it can proceed as planned. The test is carried out by adjusting the sampling frequency of the Butterworth and Chebyshev bandpass filters to 20 KHz and cutoff frequencies of 4KHz and 7KHz. Filter parameters such as cut off frequency, ripple and stop band attenuation are given randomly. This is with the aim of observing the order(n) and the different filter coefficients and their corresponding values. The conclusion from the implementation of the IIR filter design shows that the Butterworth and Chebyshev bandpass filters get an increasing filter order when the attenuation value is greater.

KEYWORDS: Filter IIR, bilinear transformation, coefficient filter, Butterworth filter and Chebyshev.

I. INTRODUCTION

Sound by definition is the result of the vibration of objects with different pressures in the air and surroundings. The pattern of oscillations that occur in the form of waves with repeated intervals is called a period. In addition, with the sound of a voice, a person's character can be recognized. Along with the development of electronics and communication technology, a speech recognition with basic characteristics is needed, namely amplitude and frequency. Amplitude is the deviation of a vibration that produces sound, while frequency is the rate at which the amplitude changes with time. Frequency is the basic component of sound formation, because it vibrates in the air. The frequency unit for the speed of sound is measured in units of Hertz (Hz), while the loudness of sound (amplitude) is in decibels (dB). Humans have the ability to hear sounds with a frequency range of 20Hz to 20KHz. The range of waves between 20Hz to 20KHz is called audiosonic, a sound that is heard by humans followed by sound waves produced in every decibel [1]. The sound that is transmitted in the communication media can experience sound quality disturbances if the sound source environment contains a lot of noise. Noise, together in the communication media becomes the background of the original sound information. Noise is a signal disturbance that is acoustic, electrical, electronic which is in one system forming unwanted interference [2]. Noise also comes from the character of electronic devices due to fluctuations in charge carriers caused by external energy. The weakness of the signal sent in the communication medium in the presence of noise becomes difficult to observe or measure.[3]. A sound signal processing method is used to reduce noise in sound. An alternative in suppressing the presence of noise in the sound is by utilizing a filter design. Filter is a circuit that can select the frequency in order to stream the desired frequency and hold the unwanted frequency. The filter circuit can be active or passive, while the type of filter is determined based on the desired frequency spectrum classification to be able to pass in the designed filter. Based on the processed signal, the filter divided into two creates the analog and the digital filter. Designed to process analog signals are analog filters, while digital filters process analog signals using digital techniques. Digital filter have digital input and output signals. Digital filters have components of addition (adder), multiplier (multiplier) and delay elements [4]. There are two digital filters, the FIR (Finite Impulse Response) digital filters and IIR (Infinite Impulse Response) digital filters. The division of digital filter types is based on the impulse filter response. IIR filters have an impulse response of infinite length, whereas FIR filters

are not. Another advantage of FIR is that it does not have a pole, so stability can be guaranteed, while for IIR with a pole it has limited stability [5]. The IIR filter requires a smaller coefficient when the frequency response is steep so that it can reduce the amount of computation time. Utilization of IIR filter at sharp cut off with high throughput. Another advantage of the IIR filter is the use of fewer parameters so that it can save memory when computing [6]. IIR filter design using a bilinear transformation have high accuracy. The bilinear transformation is equivalent to trapezoidal integration or the definite integral approximation. The processors used for IIR filter design with digital signal processing are DSK TMS320C and STM32F. DSK TMS320C is a DSP board that works both fixed-point and floating-point with a clock frequency of 225MHz, internal memory 192 kb, while the standard sampling frequency is 48KHz. TMS320C is a peripheral component used in digital signal processing experiments. While the STM32F is a 32-bit ARM processor with RISC type, equipped with ST-Link/V2 with a maximum speed of 168 MHz, 192 kb of internal RAM, voltage of 1.8 to 3.6 volts. This study aims to implement the bilinear transformation of the IIR filter as a method of minimizing noise in the voice signal. Utilization of MATLAB software as a simulation to generate filter coefficients.

II. RELATED WORK

Filter simulation using MATLAB software by entering parameters such as passband frequency, stopband frequency, ripple passband, and stopband attenuation in the filter implementation with DSK TMS67013 and CCS software. The test results on the filter implementation are the response magnitude, cut-off frequency, bandwidth, and filter quality factor as the basis for the implementation of the IIR filter by Lidyawati et al., 2015. The IIR filter design uses a GUI with MATLAB code with the Pole-zero placement method, impulse invariant, matched and bilinear z-transform. As examples of designing low-pass, high-pass and band-pass filters by Murdika and Mulyani, 2017. Utilization of the beaglebone black device with MATLAB Simulink in designing IIR filters based on time domain and frequency domain is carried out research by Prawiratama, 2017.

III. METHODOLOGY

The research method used for IIR filter design with a bilinear transformation uses a prototype of the analog transfer function that utilizes transfer function types including Butterworth, Chebyshev Type 1, Chebyshev Type 2, and Elliptic (or Cauer)[9]. Filters are used to improve signal quality by changing its characteristics, the characteristics of which are the waveform, phase, frequency and amplitude of the signal. Filters are also used to suppress noise in the signal and separate the results of the combination of two or more signals in order to obtain the original information from the signal. The input system uses voice signals with 20 Hz to 20 kHz frequency range. The process of filtering the sound signal by suppressing the noise from the disturbed sound signal. The IIR filter is a feedback loop filter so that the impulse response has an infinite or recursive time limit [10]. Utilization of the MATLAB script to achieve the desired coefficient value, then the program development and filter design calculations as a form of processing and design specifications. The use of MATLAB is also for testing variables that indicate filter parameters. This study observes the results of the sound signal filter based on the Band Pass Filter with the implementation of the IIR filter with the Butterworth and Chebyshev design. The design is in accordance with Figure 1.

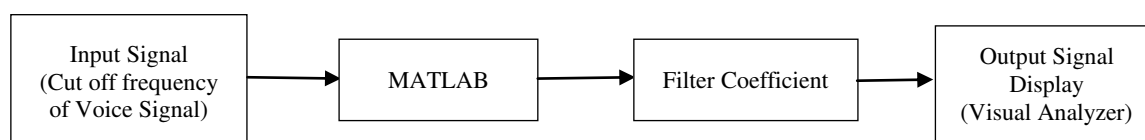


Fig. 1 System Design Diagram

Figure 1. is based on the block diagram of the system design input that entered passband frequency, stopband frequency, ripple passband, ripple stopband, and stopband attenuation in MATLAB software. Parameters determined include cut off frequency, sampling frequency and order (n) variation. Furthermore, the specification of Band Pass Filter is determined by entering the bilinear transformation and simulated in MATLAB. Parameters observed based on Simulink MATLAB include ripple passband, ripple stopband, passband attenuation, stopband attenuation, stopband frequency and passband frequency.

IV. EXPERIMENTAL RESULTS

The results of this study are based on the IIR filter design that utilizes the characteristics of the Butterworth analog filter and the Chebyshev sampling frequency is set at 20KHz, the response image of the Butterworth Bandpass filter design. Observation of stopband attenuation changes to frequency response and filter order(n), with $f_c = 4\text{KHz}$ and 7KHz . The IIR filter is expressed by the following equation,

$$y[n] = \sum_i^N = 1a_i y[n-1] + \sum_k^M = 0b_k x[n-k]$$

Wherein,

M,N= order filter

i,k = index filter

b_k =coefficient feed forward IIR filter

a_i = coefficient feedback IIR filter

From the equation that the M+N+1 value represents the total coefficient, while the order variation or filter with the symbol N, the sound signal filter based on the Butterworth bandpass filter with the design of the IIR filter from the MATLAB Simulink execution is shown in Figure 2. From the figure shown, the execution results show the frequency response of the Butterworth bandpass filter attenuation of 20dB with a filter order of $n=2$.

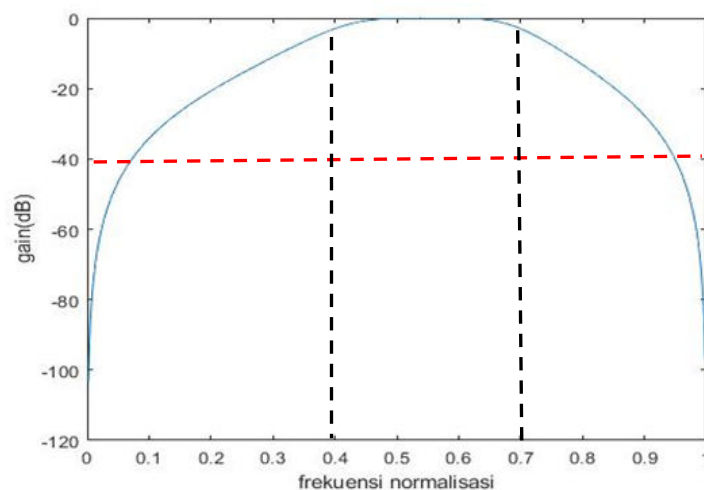


Fig. 2 Butterworth Bandpass Filter Frequency Response $A_s=20\text{dB}$ and order $n=2$

The Butterworth filter has no ripple in the frequency response, but the frequency response appears smooth and flat in the passband. Based on the results of the execution of the MATLAB program, the value of the filter coefficient in Figure 2 provides a limit with the ratio of the coefficients of the numerator and denominator according to the order of variation (n). Figure 3 graph shows the value of the change in filter response based on order (n).

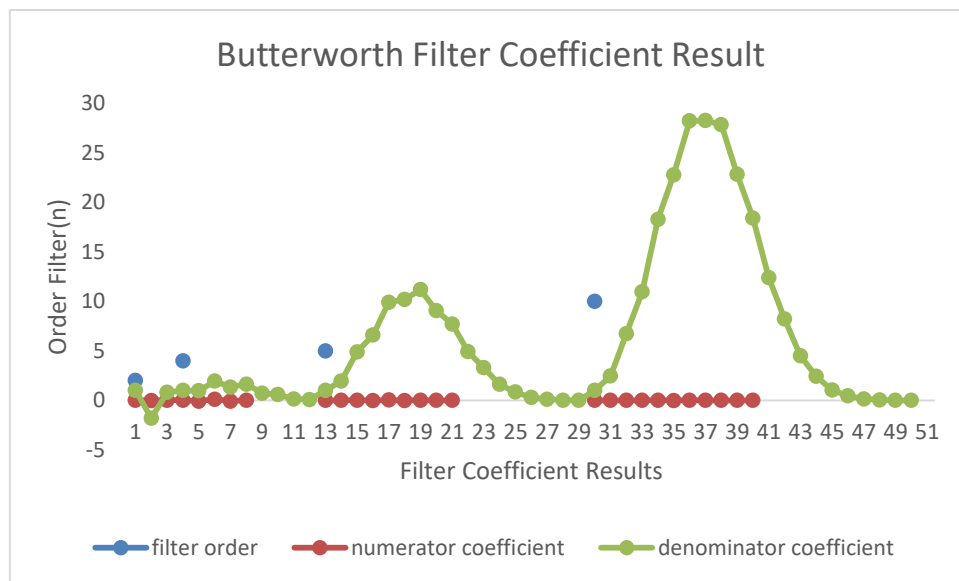


Fig. 3 Butterworth Bandpass Filter Coefficient With Variation of Order (n)

Based on the comparison graph of the order variation (n) the coefficient of the Butterworth bandpass filter with the stopband attenuation parameter setting increases with a higher order level. So that when the cut off frequencies ($f_{c1} = 4\text{KHz}$) and $f_{c2} = 7\text{KHz}$ with IIR digital filters the frequency response results are at the same cut off as shown in the black dotted line in Figure 2. Utilization of Chebyshev bandpass filters for changes in frequency response and order(n) filter with the same cut off frequency as Butterworth, namely $f_c = 4\text{KHz}$ and 7KHz . Based on the execution of the MATLAB program, the filter coefficients display data as shown in Figure 4.

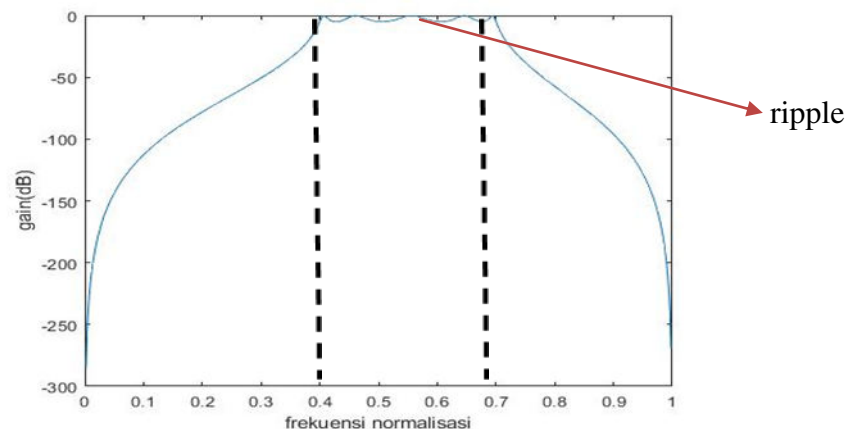


Fig. 4 Chebyshev Filter Frequency Response 1 ripple 5dB, AS 40dB with order=5

Based on the Chebyshev bandpass filter frequency response, the ripple setting as shown in Figure 4 increases with the ripple setting with the same order of attenuation getting smaller. Figure 5 shows the change in the coefficient of the denominator and numerator of the order(n) variation setting for the 4kHz and 7KHz cut off frequencies.

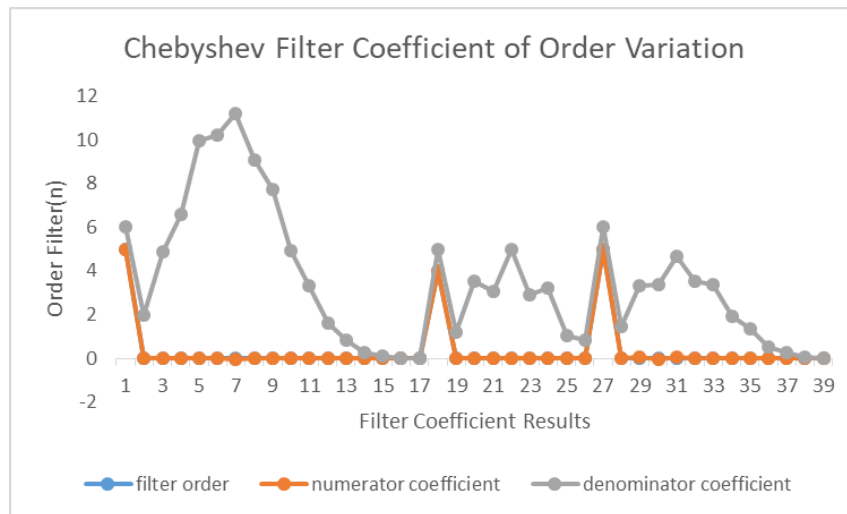


Fig. 5 Graph of Filter Coefficient Results with Variation of order (n)

Based on the graph in Figure 5 the value of the variation of the same order $n = 4$ with increasing ripple from 10 dB to 15 dB, the Chebyshev response becomes a steep roll-off by allowing ripples in the frequency response. The Chebyshev filter has ripple in its passband so that the frequency signal is preferred with constant gain. Under the same cut off conditions, the IIR filter design provides the same frequency response.

V. CONCLUSION

Based on the test results from this study, the design of an IIR filter with a prototype analog filter Butterworth and Chebyshev can utilize the bilinear transformation method, namely the analog filter transfer function is transformed into digital. The result of the bilinear transformation is a filter of constant fraction in the form of a decimal fraction. The utilization of MATLAB software in designing the IIR filter gives the results of the order (n) variation frequency response with fewer coefficients, thereby reducing the computation time. The results of the design of the IIR filter obtained coefficients that match the transfer function of the Butterworth and Chebyshev filters so that they match the input signal parameters. For filter design with the same parameters, the Butterworth filter produces a larger order than the Chebyshev filter. Larger orders result in a sharper roll off factor.

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