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Simulation Design of Chebyshev Analog Bandpass Filter for Human Sound Frequency using LTspice Software

Hartono Siswono
Electrical Engineering Department
University of Gunadarma
Depok, Indonesia

Ana Farahdiba
Electrical Engineering Department
University of Gunadarma
Depok, Indonesia

Widyastuti
Electrical Engineering Department
University of Gunadarma
Depok, Indonesia

Abstract—Filter is a very important circuit in the signal processing system. The latest technology in the field of telecommunications must have a series of filters on the system. Filters are also often found in telecommunications systems that use human voice as a source of information such as mobile phones. When recording human voices using a cell phone, the recorded sound is often mixed with noise because it is used in noisy environments, and the information recorded becomes unclear. This problem can be overcome by installing a filter that can pass the frequency range of human voice that is between 300 Hz to 3400 Hz, and in this experiment we are using an analog chebyshev bandpass filter. In this study, it was found that to pass a frequency bandwidth of 300 Hz to 3400 Hz required an order of 22 bandpass orders by arranging in series of 11 Delyannis and Friends bandpass filter circuits. Filter simulations designed in this study by using LTspice show that the filter is able to pass signals in the frequency range 300 Hz to 3400 Hz, and damped signals that have frequencies under the 300 Hz limit and above 3400 Hz according to the desired filter specifications.

Keywords—Filter; Bandpass filter; Chebyshev; Delyannis and Friends; Operational Amplifier

I. INTRODUCTION

Background: Filters are often found in telecommunications systems that use human voice as a source of information such as mobile phones. When recording human voices using a cell phone, the recorded sound is often mixed with noise because it is used in noisy environments, and the information recorded becomes unclear. This can cause the information exchange process to be repeated so that it takes longer so that information can be conveyed optimally. It is important to reduce the attenuation produces by random noise and improve the performance of the signal [1]. There are some research in this area, such as “Noise Estimation and Noise Removal Techniques for Speech Recognition in Adverse Environment” by Urmila Shrawankar[2]. A.G. Maher wrote his paper “A Comparison of Noise Reduction Techniques for Speech Recognition in Telecommunications Environments” [3]. “Noise Cancellation in Speech Signal Processing-A Review” was written by S. Lakshmikanth [4]. Minajul Haque and Kaustubh Bhattacharyya wrote the paper “A Study on Different Linear and Non-Linear Filtering Techniques of Speech and Speech Recognition”[5]. Ohnmar Win wrote “IIR Filter Design for De-Nosing Speech Signal using Matlab” [6]. R. R. Porle wrote “A Survey of Filter Design for Audio Noise Reduction” [7]. Prajoy Podder wrote “Design and Implementation of Butterworth, Chebyshev-I and Elliptic Filter for Speech Signal Analysis” [8]. Subhash Chand Samota wrote “Speech Signal Processing: A Technical Review” [9]. Ovidiu Buza wrote “Voice Signal Processing For Speech Synthesis” [10].

Proposed Solution: To solve the problem described above, we need a device that can help pass the information signal that is human voice free from noise. And this problem can be overcome by installing a filter that can pass the frequency range of human voice that is between 300 Hz to 3400 Hz [11], and the filter that passes the frequency range is a bandpass filter. Bandpass filter will free human voice from noise by passing the human voice frequency bandwidth and dampening frequency other than the human voice frequency bandwidth. For this reason in this paper, we design an analog bandpass filter that can pass the frequency bandwidth of human voice. The filter design method used by the author is the Chebyshev method[12,13]. This filter will be simulated using LTSpice software [14,15] so that it can simplify the design process and see the performance of the Chebyshev band pass filter design.

II. RESEARCH METHOD

A. Design of Chebyshev Analog Bandpass Filters

Chebyshev filter has the following magnitude response equation [16, 17].

$$|T_n(j\omega)| = \frac{A}{\sqrt{1 + \varepsilon^2 C_n^2(\omega)}} \tag{1}$$

Where the value of A is the desired filter gain, $T_n(j\omega)$ is the Chebyshev magnitude response function, ε is the ripple, $C_n(\omega)$ is the chebyshev polynomial function. And n is the filter order. Where $C_n(\omega)$ is obtained from

International Journal of Engineering Research & Technology (IJERT)
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Vol. 10 Issue 01, January-2021

$$C_n(\omega) = \cos n \cos^{-1}(\omega) \text{ for } |\omega| \leq 1 \tag{2}$$
$$C_n(\omega) = \cosh n \cosh^{-1}(\omega) \text{ for } |\omega| > 1 \tag{3}$$

From the chebyshev polynomial function above, we get the following characteristics ω.
For ω = 0

$$C_n(0) = 0, \text{ n odd} \tag{4}$$
$$C_n(0) = 1, \text{ n even} \tag{5}$$

Then

$$|T_n(j\,0)| = 1, \text{ n odd} \tag{6}$$
$$|T_n(j\,1)| = \frac{1}{\sqrt{1+\varepsilon^2}}, \text{ n odd} \tag{7}$$

For $\omega = 1$

$$C_n(1) = 1, \text{ for all n} \tag{8}$$

Then

$$|T_n(j\,1)| = \frac{1}{\sqrt{1+\varepsilon^2}}, \text{ for all n} \tag{9}$$

To be able to design an analog chebyshev bandpass filter, we must know the specifications needed in accordance with the designation of the filter. The required specifications are frequency and attenuation, these two things are usually depicted in graphical form like Figure 1.

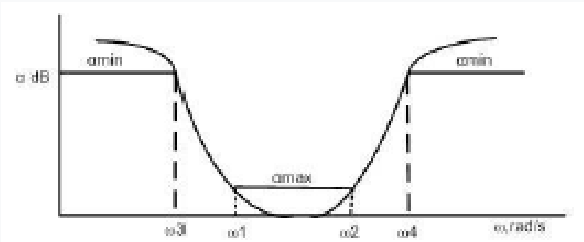

Fig. 1. Graphic Specifications of Bandpass Filter

To transform bandpass filter becomes a lowpass filter then the center frequency (ω_0) is needed, also passband frequency bandwidth (B), while the current passband frequency the filter is transformed into a lowpass (Ω_p) is made as 1, and (Ω_s) is stopband filter current frequency transformed into lowpass. Values it is obtained from the equations following.

$$\omega_0 = \sqrt{\omega_1 \cdot \omega_2} \tag{10}$$
$$B = \omega_2 - \omega_1 \tag{11}$$
$$\Omega_s = \frac{\omega_4 - \omega_3}{\omega_2 - \omega_1} \tag{12}$$

After values of ω_0, B, and Ω_s, the transformation results are obtained the bandpass filter becomes a lowpass filter so the graph shown in Figure 2 [18].

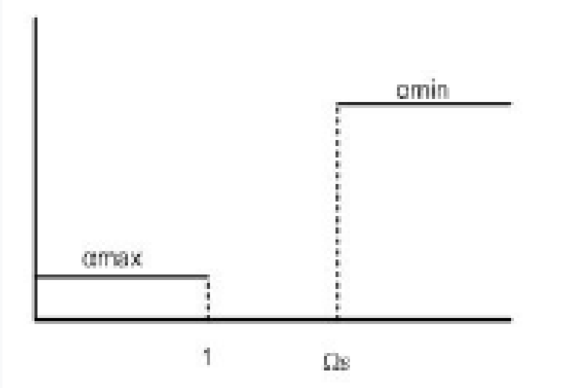

Fig. 2. Graph of Transforming Bandpass Filters into Lowpass Filter

After doing the transformation, to design the chebyshev filter the first step is to determine the order of the filter or n with the formula shown by Equation 13 [19,20].

$$n = \frac{\cosh^{-1}\left[\sqrt{\frac{10^{\alpha_{min}/10} - 1}{10^{\alpha_{max}/10} - 1}}\right]}{\cosh^{-1}\left(\frac{\Omega_s}{\Omega_p}\right)} \tag{13}$$

The value of n obtained determines how many filter poles. keep in mind that the number of chebyshev bandpass filter poles is twice the number of chebyshev lowpass filter poles. If the value of n is odd then there is a pole angle (Ψ) of value 0^0. If the value of n is even then the number of poles is obtained through Equation

$$\Psi = \pm \frac{90^o}{n} \tag{14}$$

And the distance between the poles, for n is odd or even, is obtained through Equation

$$\Psi = \pm \frac{180^o}{n} \tag{15}$$

Therefore we get the chebyshev poles expressed in Equation

$$s = -\sigma \pm j\,\omega \tag{16}$$

The values of σ and ω are obtained through Equation

$$\sigma = \cos \Psi \sinh a \tag{17}$$
$$\omega = \sin \Psi \cosh a \tag{18}$$

And the value of a is obtained through the Equation

$$a = \frac{1}{n} \sinh^{-1} \frac{1}{\varepsilon} \tag{19}$$

The value of ε is obtained from the Equation 20 [21].

$$\varepsilon = \sqrt{10^{\alpha_{max}/10} - 1} \tag{20}$$

To design a chebyshev bandpass filter, Geffe Algorithm is needed, because the poles previously obtained are still in the form of lowpass filters. Geffe Algorithm will assist in finding

International Journal of Engineering Research & Technology (IJERT)
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Vol. 10 Issue 01, January-2021

lowpass filter poles which are equivalent to bandpass filters. Geffe Algorithm is divided into two, based on the order of Geffe Algorithm of order 1 and Geffe Algorithm of order 2. Geffe Algorithm of order 1 is an algorithm used at poles which only has real value. And 2nd Order Geffe Algorithm is used on poles which have real and imaginary values. Geffe Algorithm for order 1 is shown by the following Equations.

$$s = -\sigma \quad (21)$$
$$\Sigma_1 = \sigma \quad (22)$$
$$q_c = \frac{\omega_0}{B} \quad (23)$$
$$Q = \frac{q_c}{\Sigma_1} \quad (24)$$

For the 2nd order Geffe Algorithm is shown by the following equations.

$$s = -\sigma \pm j\omega \quad (25)$$
$$\Sigma_2 = \sigma \quad (26)$$
$$\Omega_2 = \omega \quad (27)$$
$$C = \Sigma_2^2 + \Omega_2^2 \quad (28)$$
$$D = \frac{2\,\Sigma_2}{q_c} \quad (29)$$
$$E = 4 + \frac{C}{q_c^2} \quad (30)$$
$$G = \sqrt{E^2 - 4D^2} \quad (31)$$
$$Q = \frac{1}{D}\sqrt{\frac{1}{2}(E + G)} \quad (32)$$
$$K = \frac{\Sigma_2 \cdot Q}{q_c} \quad (33)$$
$$W = K + \sqrt{K^2 - 1} \quad (34)$$
$$\omega_{01} = \frac{1}{W} \cdot \omega_0 \quad (35)$$
$$\omega_{02} = W \cdot \omega_0 \quad (36)$$

After values ω_{01}, ω_{02}, and Q from the 2nd order Geffe Algorithm obtained then a filter can be designed. On Geffe Algorithm of order 1 and Geffe 2nd order algorithm, to design filter needed frequency scaling value (k_f), this value follows the value of ω_0 of each order.

In bandpass filter design, we are using the Delyannis and Friends circuit as shown in Figure 3 [22,23,24,25].

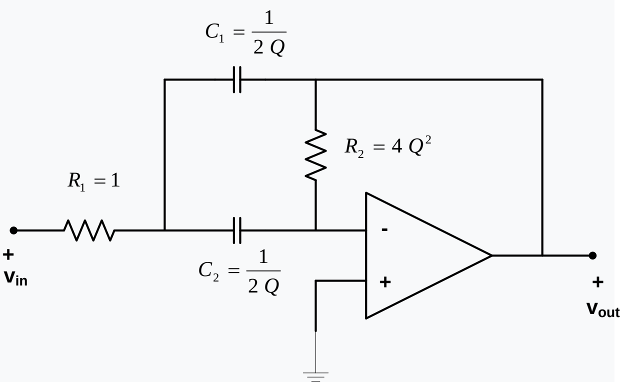

Fig. 3. The Delyannis and Friends Circuit

III. DESIGN AND RESULT

A. Block Diagram

Block diagram is a concise image statement from combination of cause and effect between input and output of a system. Block diagram used for design of an analog bandpass filter chebyshev for sound frequency humans use software LTSpice, is as shown in Figure 4.

Fig. 4. The Diagram Block

B. Input Block

In Figure 4, the input block describe input which will be processed. In the input section, the circuit given DC voltage source (V_{DD}) and sinusoidal waves (V_{in}) as signal that will be processed by filter.

DC voltage source that is inserted in the filter circuit serves as an activator of the filter circuit. The DC voltage source is required because the circuit uses the Op-Amp as the main component of the circuit and the Op-Amp will only operate when given a DC voltage. The value of the DC voltage used in the circuit is 3.3 volts. DC voltage values can be adjusted to the Op-Amp specifications. The op-amp used in filter design in this study is LTC6261 IC. The LTC6261 IC can be activated if it gets a DC voltage of 1.8 volts to 5.25 volts according to the specifications described in the IC LTC6261 datasheet.

Other inputs included in the filter circuit are sinusoidal waves (Vin) which act as human voice signals to be filtered.

C. Process Block

The first step to design this bandpass filter is to determine the specifications of the filter. Based on the theoretical basis it is known that the frequency bandwidth (B), which is passed is at a frequency of 300 Hz to 3400 Hz. The stopband frequency (ω_{stop}) in this study was set at 100 Hz for the lower limit stop frequency. For the upper limit stop frequency is set at 4000 Hz.

For the attenuation value used at the stop frequency (α_{min}) is 60 dB. Attenuation at the stop frequency is set at 60 dB because the intensity of human speech when normal speaking is below 60 dB. For attenuation at pass frequency (α_{max}) is set at 5 dB because the intensity of the sound that is emitted by the smallest human when whispering is 10 dB. If this bandpass filter specification is depicted on the frequency response graph it will look like Figure 5.

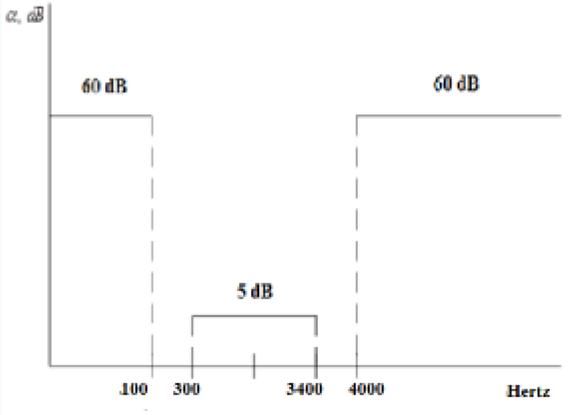

Fig. 5. Bandpass Filter Specifications in Hz

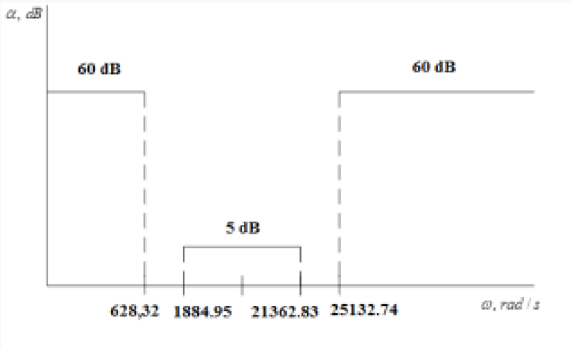

Fig. 6. Bandpass Filter Specifications in rad/s

Furthermore, the bandpass filter specification is transformed to the lowpass filter specification by determining the middle frequency filter value (ω_0), using Equation 10. To calculate bandwidth (B) use Equation 11. While the value of the frequency pass in the transformation (transformasi) must be equal to 1.

Then to determine the value of the stop frequency on the results of the transformation (Ω_s) can be determined through Equation 12. We get ω_0 = 6345.7 rad/s, B = 19,477.88 rad/s, Ω_p = 1, Ω_s = 1.26. So that the specification form of the filter after being transformed can be seen in Figure 7.

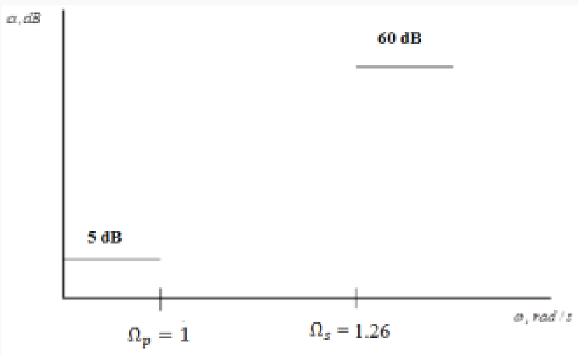

Fig. 7. Filter Specifications after Transformed

The next step is to determine the order of the circuit (n) chebyshev bandpass filter using Equation 13 and determine the pole of the filter. We get n = 10.21 and is rounded up to 11. Finding the poles angles by using Equation 15, we get ψ = 0º, ± 16.36º, ± 32.72º, ± 49.08º, ± 65.44º, ± 81.8º. After the angle value for each pole is determined, then the ripple value (ε) and the chebyshev pole variable value (a) are needed. The value (ε) can be determined through Equation 20, while to determine the value (a) through Equation 19. We get ε = 1.47 and a = 0.058. Next determine the position of the poles using Equation 17 to obtain the real value of the pole (σ), and Equation 18 to obtain the imaginary value of the pole (ω), and Equation 16 to obtain the value of the pole (s). So the value (s) of each angle is obtained as in Table 1.

The next step taken when the values of all poles or (s) have been found is to carry out a calculation process to determine the quality factor (Q_n) and the middle frequency (ω_{0n}) of the circuit generated from each pole using the Geffe Algorithm. Geffe Algorithm has two types of operations namely 1st order Geffe Algorithm and 2nd order Geffe Algorithm. First order Geffe Algorithm is used on poles which only have real numbers, from the first order Geffe Algorithm is produced by a series of filters, where the series has a middle frequency circuit (order 1) used on poles that only have real numbers. ω_{0n} obtained from the main middle. To get the quality factor (Qn) of the series in order 1, the overall quality factor (q_c) and polar real value (Σ) are required. In this study, the first pole is obtained (s_1 = - 0.058), which is a pole that only has real numbers. Because the first pole only has real numbers, the Geffe Algorithm is a first-order type, so that the values Q_1 and ω_{01} are obtained by using Equations 21, 22,23,and 24. We get Q_1 = 5.69 and ω_{01} = ω_0 = 6345.7 rad/s. For poles $s_{2\&3}$, $s_{4\&5}$, $s_{6\&7}$, $s_{8\&9}$ and $s_{10\&11}$, we use Equations 25 to 36 and the results are shown in Table 2.

From the Geffe Algorithm calculation, a chebyshev bandpass filter analog circuit diagram block is obtained, as shown in Figure 8.

From the Block Diagram, we design the circuit which is shown in Figure 9.

D. Result

The overall simulation results of designing Chebyshev analog bandpass filter to pass the frequency of human voice in this study can be seen in Figure 10.

From the simulation results, we can see that we get the filter we want, but there are ripples on the passband section that match the characteristics of the Chebyshev filter. From the simulation, the passband is between 300 Hz – 3400 Hz. This means that all signals between 300 Hz – 3400 Hz will be passed, and signals with frequencies below 300 Hz and above 3400 Hz will be damped.

IV. CONCLUSION

From the filter design research, it can be concluded that the designed filter circuit is capable of passing a frequency bandwidth of 300 Hz to 3400 Hz as desired in the initial specifications for filter manufacturing.

From the results of this study it is also known that to design a chebyshev analog bandpass filter according to the desired specifications in this study, a filter circuit that has 22 bandpass filter orders is required. And the set has 11 Delyannis and Friends filter sets arranged in cascade.

The design and simulation of the filter circuit in this study were carried out with the LTSpice software. And the design results produce a simulation that shows that the circuit meets

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Vol. 10 Issue 01, January-2021

the basic characteristics of the Chebyshev filter. This is indicated by the steep roll-off of the circuit, and the ripple of the passed frequency bandwidth.

TABLE 1. POLES OF EACH FILTER ANGLE

Poles	**Angles**					
	0^0	$\pm 16.36^0$	$\pm 32.72^0$	$\pm 49.08^0$	$\pm 65.44^0$	$\pm 81.8^0$
σ	0.058	0.056	0.049	0.038	0.024	0.0083
ω	0	0.28	0.54	0.76	0.91	0.99
s	-0.058	$-0.056 \pm j0.28$	$-0.049 \pm j0.54$	$-0.038 \pm j0.76$	$-0.024 \pm j0.91$	$-0.0083 \pm j0.99$

TABLE 2. RESULTS OF CALCULATION OF GEFFE ALGORITHM ORDER 2

Q and ω_0	**Poles**				
	$S_{2\&3} = -0.056 \pm j0.28$	$S_{4\&5} = -0.049 \pm j0.54$	$S_{6\&7} = -0.038 \pm j0.76$	$S_{8\&9} = -0.024 \pm j0.91$	$S_{10\&11} = -0.024 \pm j0.91$
Q	6.39	8.68	13.27	23.51	72.11
ω_0	ω_{02} = 4224.83 rad/s ω_{03} = 9531.24 rad/s	ω_{04} = 3014.58 rad/s ω_{05} = 13,357.7 rad/s	ω_{06} = 2358.996 rad/s ω_{07} = 17,069.93 rad/s	ω_{08} = 2048.98 rad/s ω_{09} = 19,652.63 rad/s	Ω_{10} = 1911.35 rad/s Ω_{11} = 21,067.72 rad/s

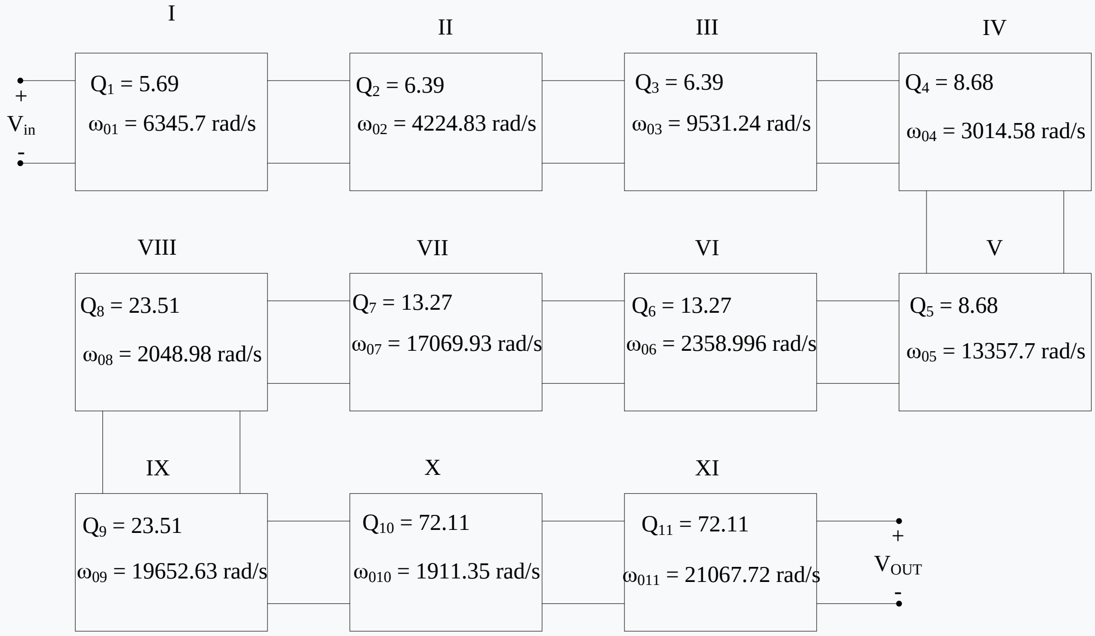

Fig. 8 Circuit Block Diagram

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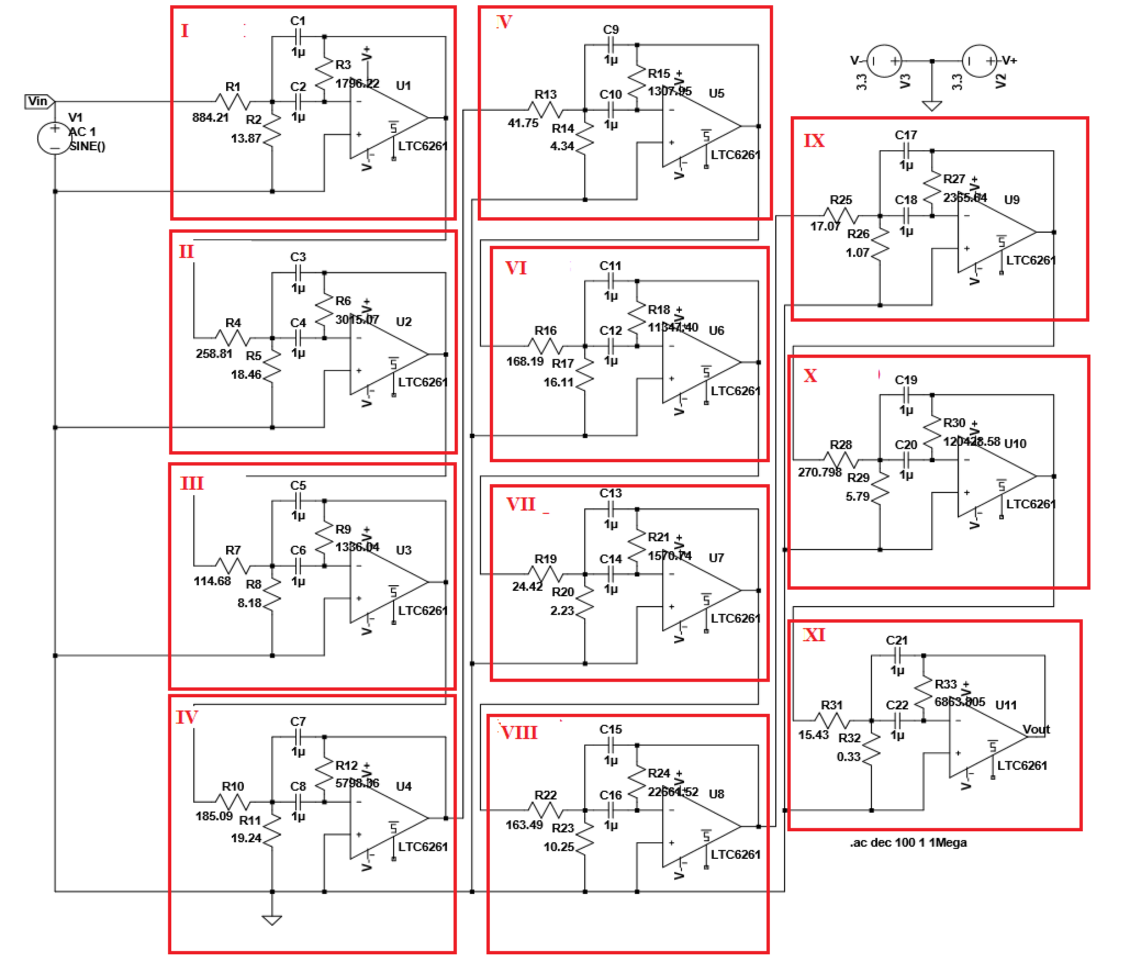

Fig. 9 The Circuit

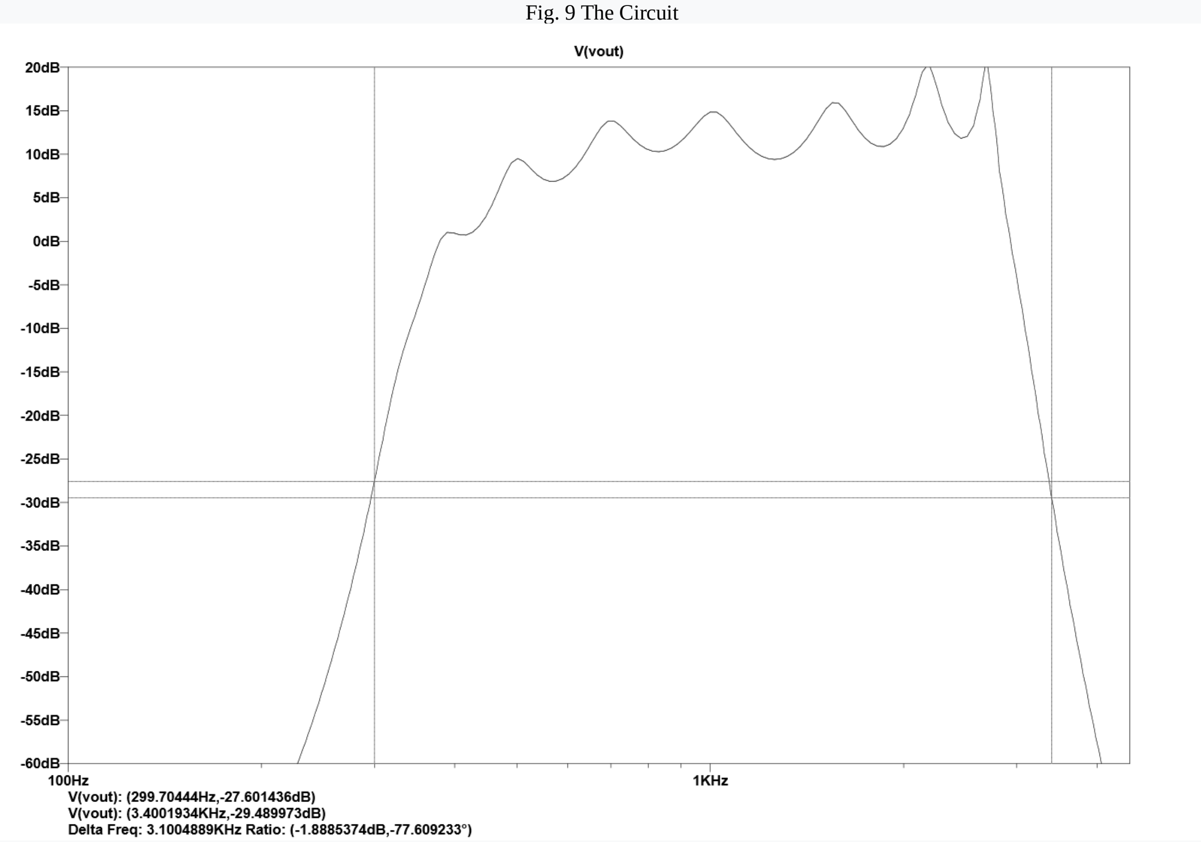

Fig. 10. Simulation Results of Chebyshev Filter Circuit Frequency Response

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