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DESIGN ANALOG RC-ACTIVE 2ND-ORDER AUDIO LOW PASS FILTER (LPF)

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ABSTRACT:- The electronic filters are considered as a very impartment element of electronic circuits . These filters have a basic rule in electronic and communication devices and systems.

Using these filters , the correct decisions can be taken depending on the applications which it have been used for it. These decision which have been done may have relations with frequency, amplitude, type of signal and the function of applications. In this research RC-active 2nd-order Low Pass Filter (LPF) is designed for low frequency (audio frequency) passing. This filter is designed and simulated using EWB technique. The simulation results obtained from implementation this designed filter is nearly identical with the results obtained from theoretical analysis approximately. These results mean that the designed filter is passed the test successfully.

Keywords:- RC, Active, LPF, Audio frequency.

1-INTRODUCTION

A filter is a circuit used to pass a specified band of frequencies while attenuating all frequencies outside that band. Filters can be classified in different forms as following:

According to configuration filters, for frequency to be processed, they can be classified as audio, radio and microwave filters [1,2].

According to frequency selectivity, low-pass, high-pass, band-pass and band reject or notch filters. Filters can be also classified as analog filters which are used to process analog signals which are function of continuous time, while, digital filters on the other hand process digital signals. Filters must be classified as passive or active depending on the type of the element used in its construction. Passive filter networks contain only resistors, inductors and capacitors, while active filters employ active devices, such as op-amp, transistor, with resistors and capacitors.

Inductors are not used in active filters because they are bulky, costly, and may have large internal resistance component.[2,3]

Active filters provide the following additional advantage: [4,5].

- Can be optimized to have a good dynamic range.
- Possibility of minimized chip area when it is realized in an integrated form.
- Increased circuit reliability because all processing steps can be automated.
-Improvement in performance because high-quality components can be readily manufactured.
-In large quantities, the costs of integrated circuit active filters are much lower than equivalent passive filters.
-A reduction in parasitic because of smaller size.

2.TYPES OF FILTERS ACCORDING TO FREQUENCY SELECTIVITY

a. A low pass filter(LPF): [6,7]

It is a circuit that has constant output voltage from d.c. up to cut off frequency fc . If frequency increases above f_c, the out output voltage is attenuated (decreases). Figure(1):Shows the plot for ideal and practical low pass filter(LPF).

The cut off frequency f_c, is also called the (0.707) frequency, or the (-3db) frequency.

b. High pass filter(HPF)

It is a circuit that attenuates the output voltage for all frequencies below the cut off frequency f_c. Above f_c, the magnitude of the output voltage is constant. Figure(2) shows the plot for ideal and practical high pass filter(HPF).

c. Band pass filter(BPF)

It is a circuit that passes only a band of frequencies while attenuating all frequencies outside the band. Figure(3):Shows the plot of ideal and practical band pass filter(BPF).

d. Band reject filter(BRF)

It is a circuit that attenuates only a band of frequencies while passing all frequencies outside the band. Figure (4): shows the plot of ideal and practical band reject filter (BRF).

3. FREQUENCY RESPONSE

The ideal characteristics cannot be realized practically .But the approximate response is realized.

Butterworth, Chebyshev and Bessel-Thomson can be used to approximate the ideal.

Butterworth function is analyzed as follow [1] :

The general form of filter transfer function:

$$H(s) = \frac{N(s)}{D(s)} = \frac{\sum_{k=0}^{k=m} a_k s^k}{\sum_{k=0}^{k=n} b_k s^k} \tag{1}$$

Where n ≥ m

N(s), D(s) are polynomial in the variable (s)

H(s), is a transfer function of second-order section as indicated in equation(2).

$$H(s) = \frac{\sum_{k=0}^{k=2} a_k s^k}{\sum_{k=0}^{k=2} b_k s^k} \tag{2}$$

Which is characterized by two parameters: [8]

$$Q = \frac{\sqrt{b_0}}{b_1} \tag{3}$$

Q: Pole quality factor

$$\omega_0 = \sqrt{b_0} \tag{4}$$

and

ω_0 : pole frequency

The transfer function of low pass filter (LPF) is indicated in the following equation:

$$H(s) = \frac{k}{s^2 + \left(\frac{\omega_0}{Q}\right)s + \omega_0{}^2} \tag{5}$$

Where

$$b_0 = \omega_0{}^2 \tag{6}$$

$$b_1 = \frac{\omega_0}{Q} \tag{7}$$

And ,

$$H(s) = \frac{\frac{k}{{\omega_0}^2}}{\frac{s^2}{{\omega_0}^2} + \left(\frac{1}{Q}\right)\left(\frac{s}{\omega_0}\right) + 1} \qquad (8)$$

yielding:

$$H(s) = \frac{H_0}{\frac{s^2}{{\omega_0}^2} + \left(\frac{1}{Q}\right)\left(\frac{s}{\omega_0}\right) + 1} \qquad (9)$$

Then

$$H_0 = \frac{k}{{\omega_0}^2}$$

4.THE FUNCTION , ANALYSIS AND DESIGN 2ND-ORDER CIRCUIT

4.1 The function & analysis 2nd-order LPF

Referring to equation (5) the transfer function for the low pass filter (LPF) can be derived as follows:

$$H(s) = \frac{k}{s^2 + \left(\frac{\omega_0}{Q}\right)s + {\omega_0}^2} = \frac{N(s)}{D(s)} \qquad (10)$$

In active circuits it can be recognize the passivity of gain and also that the associated circuit may be inverting or non inverting. Such a transfer function is

$$H(s) = \frac{\mp k}{s^2 + \left(\frac{\omega_0}{Q}\right)s + {\omega_0}^2} \qquad (11)$$

Choosing the negative sign in equation (11) meaning that anticipated an inverting realization of the transfer function. Then equation (11) will becomes as follows:

$$H(s) = \frac{-k}{s^2 + \left(\frac{\omega_0}{Q}\right)s + {\omega_0}^2} \qquad (12)$$

Beginning by rewriting equation (12) as

$$\left[s^2 + \left(\frac{\omega_o}{Q}\right)s + {\omega_0}^2\right]\mathrm{V_o} = -\mathrm{kV_i} \qquad (13)$$

To determine the output voltage V0(t) from the input Vi(t), we need to perform two integrations. But since dividing by s it is easier operation than integration (1/s is the integration operator). Let us return to the frequency domain and recast equation(13) in a form that lets us identify integration 1/s , we obtain

$$s\left[s+\left(\frac{\omega_0}{Q}\right)\right]\mathrm{V_o} = -(\mathrm{kV_i} + \omega_0{}^2\mathrm{V_o})$$

$$\left[s+\left(\frac{\omega_0}{Q}\right)\right]\mathrm{V_o} = \frac{(-\mathrm{kV_i} - \omega_0{}^2\mathrm{V_o})}{s} = \mathrm{V_B} \qquad (14)$$

It is clear that V0 can be obtained by integrating the voltage

$$\mathrm{V_o} = \frac{1}{s+\left(\frac{\omega_0}{Q}\right)}\mathrm{V_B} = \left[\frac{(-\mathrm{kV_i} - \omega_0{}^2\mathrm{V_o})}{s}\right] \qquad (15)$$

Rewriting equation (15) as:

$$\mathrm{V_B} = -\frac{1}{s}[\mathrm{kV_i} + \omega_0{}^2\mathrm{V_o}] \qquad (16)$$

And,

$$\mathrm{V_B} = -\frac{1}{s}V_\mathrm{H} \qquad (17)$$

Equation (17) says that VB is obtained by integrating, with a sign inversion. The voltage VH which in turn is obtained by summing two scaled voltages [8,9] .

$$V_\mathrm{H} = \mathrm{kV_i} + \omega_0{}^2\mathrm{V_o} \qquad (18)$$

Thus, we have identified the double integration to obtain V0

$$\mathrm{V_o} = \frac{1}{s+\left(\frac{\omega_0}{Q}\right)}\left[-\frac{1}{s}V_\mathrm{H}\right] \qquad (19)$$

We also obtain inverting low pass filter biquadrates transfer function in equation (12).The non inverting integrated is complex in mathematical computing for transferring non inverting to inverting this can be obtained by putting the negative resister to the inverting operation amplifier in order to became non inverting. Figure(5): shows the second order low pass filter (LPF) circuit [10] .

By assumption of ideal operation amplifiers , so that $V_1^- = V_2^- = 0$, then the analysis done as :

$$V_iG_6 + V_{o1}G_8 + V_{o2}C_1s = 0 \qquad (20)$$

$$V_{o2}G_{10} + V_{o2}C_2s - V_{o1}G_9 = 0 \qquad (21)$$

From equation(20) obtaining that:

$$V_{o1} = -\frac{(V_i G_6 + V_{o2} C_1 s)}{G_8} \qquad (22)$$

From equation(22) and equation(21) , one can be obtained the following:

$$\frac{V_{o2}}{V_i} = \frac{-G_6 G_9}{C_1 C_2 s^2 + C_1 G_{10} s + G_8 G_9} \qquad (23)$$

Equation(23) can be rewritten as :

$$\frac{V_{o2}}{V_i} = \frac{-G_6 G_9 / C_1 C_2}{s^2 + (\frac{G_{10}}{C_2}) s + (G_8 G_9 / C_1 C_2)} \qquad (24)$$

By comparing equation (24) with equation (12) , we will obtain:

$$\frac{\omega_0}{Q} = \frac{G_{10}}{C_2} \qquad (25)$$

$$k = \frac{G_6 G_9}{C_1 C_2} \qquad (26)$$

$$\omega_0{}^2 = \frac{G_8 G_9}{C_1 C_2} \qquad (27)$$

4.2 Design Analog RC-active 2nd –order Low Pass Filter (LPF): (11,12).

In this research analog second order LPF IS designed with cutoff frequency f_c =20KHz , the pole quality factor Q=0.707 and DC gain of 0 dB :

$$0dB = 20\log H_o \qquad , H_o = 1 \qquad \omega_0 = 2\pi f_0$$

$$\omega_0 = 40\pi(1000) \qquad (rad/sec)$$

And $k = H_o \omega_0{}^2$

$$k = 15791367040$$

From equations (25) , equation (26) and equation (27) yield:

$$177742.1586 = \frac{G_{10}}{C_2}$$

$$15791367040 = \frac{G_6 G_9}{C_1 C_2}$$

$$15791367040 = \frac{G_8 G_9}{C_1 C_2}$$

With assuming the value of the capacitors:

$$C_1 = 100nF \qquad , C_2 = 100nF$$

And the resister:

$R_9 = 80\,\Omega$

From that , yield:

$R_6 = R_8 = 79.15717473\,\Omega$

$R_{10} = 56.26127238\,\Omega$

Figure (6) shows the designed analog RC-active second-order Low Pass Filter (LPF)

5.SIMULATION RESULTS

The designed of analog RC-active 2nd-order Low Pass Filter (LPF) is shown in Figure(7).

The analog input signal with amplitude (Vr.m.s = 10 volt) and frequency (10KHz) under band limit of cut off frequency of active filter Is shown in Figure(8).

After pass this input signal through above filter the analog output signal has the same amplitude and frequency of the input signal , but with out phase at (180◦) as shown in Figure(9).

The phase shift between input and output signal can be canceled using inverting operation amplifier with unity gain, it is connected to the output of the active filter as shown in Figure(10).

The result output signal has same phase shift of the input signal as shown in Figure(11). The frequency response of this active filter Is shown in Figure (12).

The cut off frequency can be calculated practically as fellows by :
dc. gain of 0 dB that mean the gain at cut off frequency is (0dB – 3dB), (-3dB) = 20 log (0.707) . For linear (H0 (0.707) at this point of gain , cut off frequency is(20.261KHz) as shown in Figure(13).

Notice that, when input signal has frequency out off band limit (greater cut off frequency), the out put signal attenuated as shown in Figure (14).

The simulation results obtained from implementation this designed filter is nearly identical with the results obtained from theoretical analysis approximately. These results mean that the designed filter is passed the test successfully.

This design filter was designed and implemented using electronic work bench(EWB).
It is used for detecting the audio signal of communication system such as telephone networks.
It is used in different application such as to detect any signal has frequency less than 20KHZ.

CONCLUSIONS AND FUTURE PERSPECTIVES

The designed filter can be considered as ideal filter by increasing the order of filter since all-poles low pass function $|H(s)|$ rolls off for high frequencies at a rate of $-Nth * 20dB/decade$.

This electronic filter which is designed can be consider as the basic of the development techniques for the switched capacitor active filter , the current mode and current converter for more advanced high and low frequencies application .

The designed filter can be modified to accept different signal with different frequency by using variable capacitances & resistances ,this mean that the designed filter become adaptive filter.

The design filter can be implemented using orcade technique ,also can be implemented practically.

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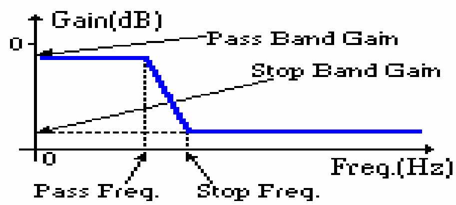

Fig.(1): The ideal and practical LPF.

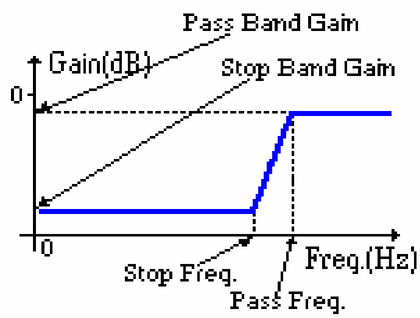

Fig.(2): The ideal and Practical HPF.

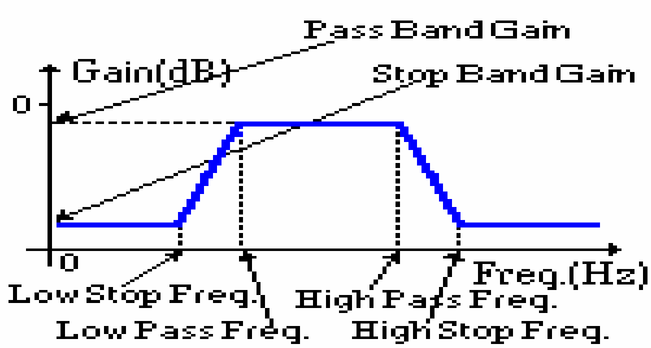

Fig.(3): The ideal and Practical BPF.

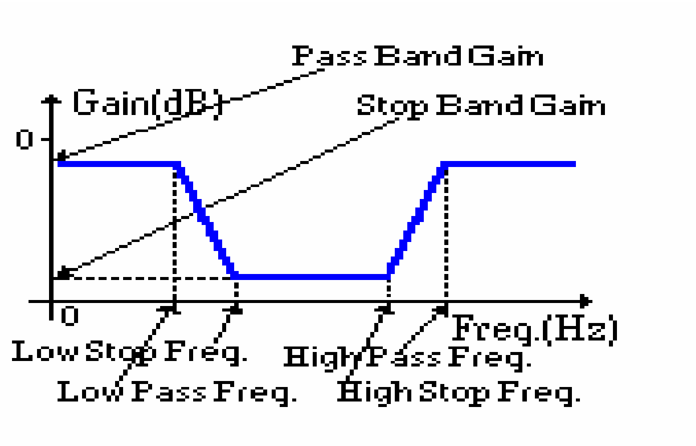

Fig.(4): The ideal and Practical BSF.

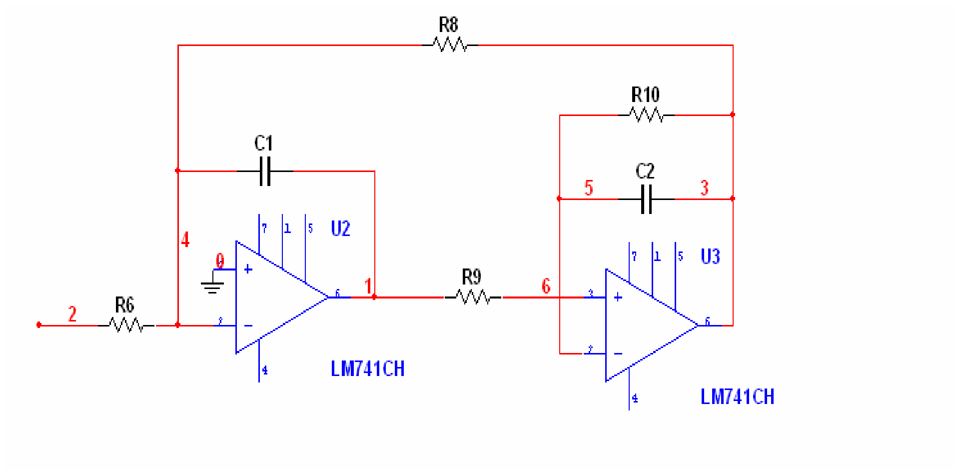

Fig.(5): Second order LPF (10) .

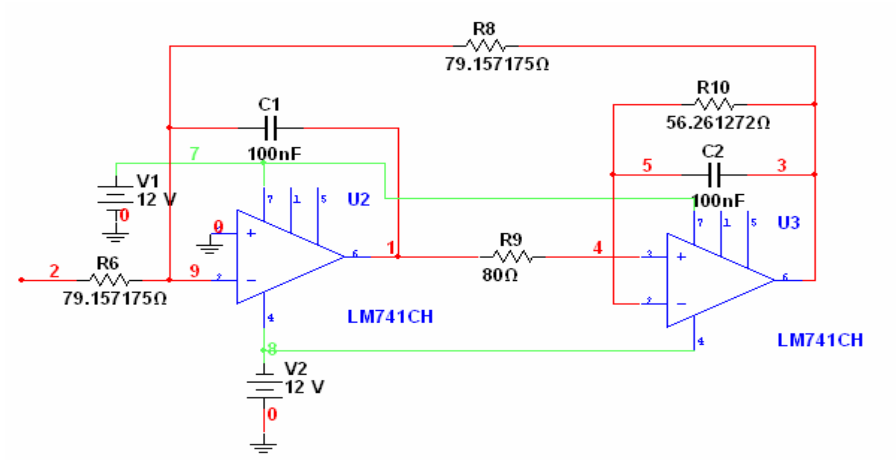

Fig. (6): The designed RC active second order LPF.

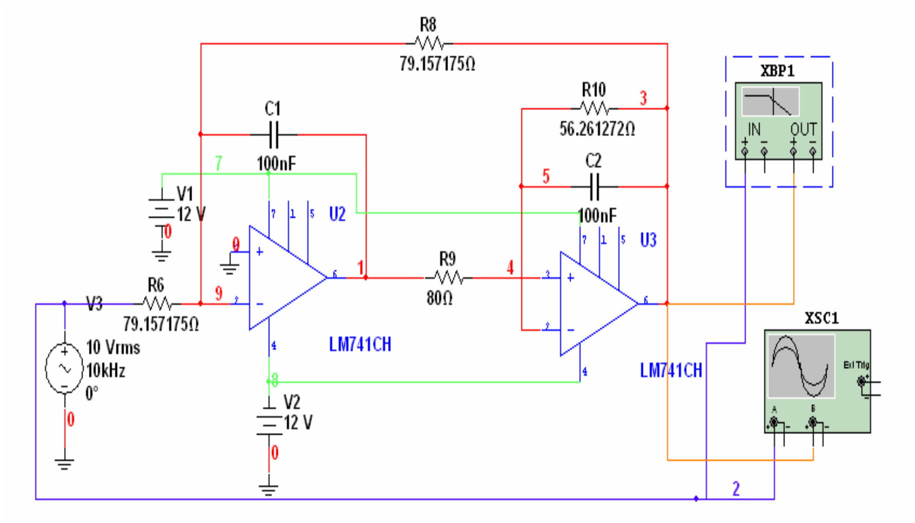

Fig:(7): Analog 2nd order LPF.

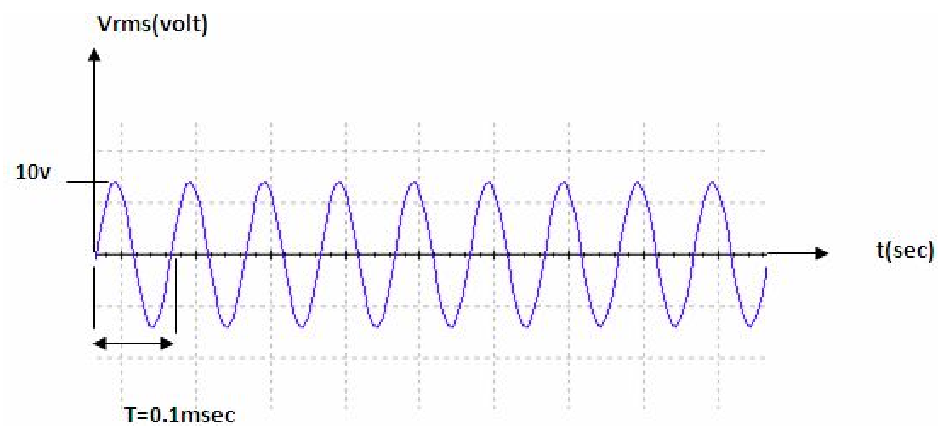

Fig.(8): Analog input signals.

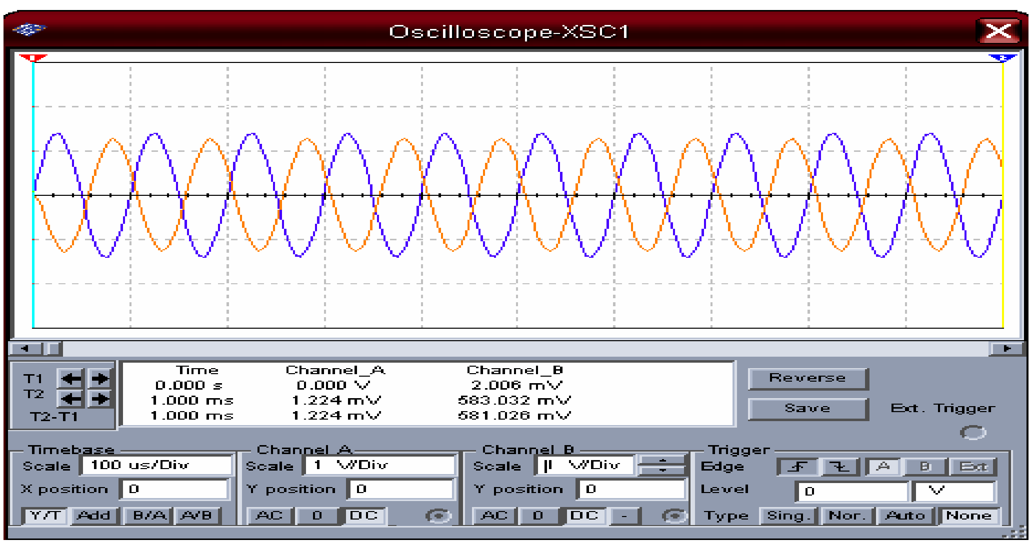

Fig.(9): Input &output signal of designed filter.

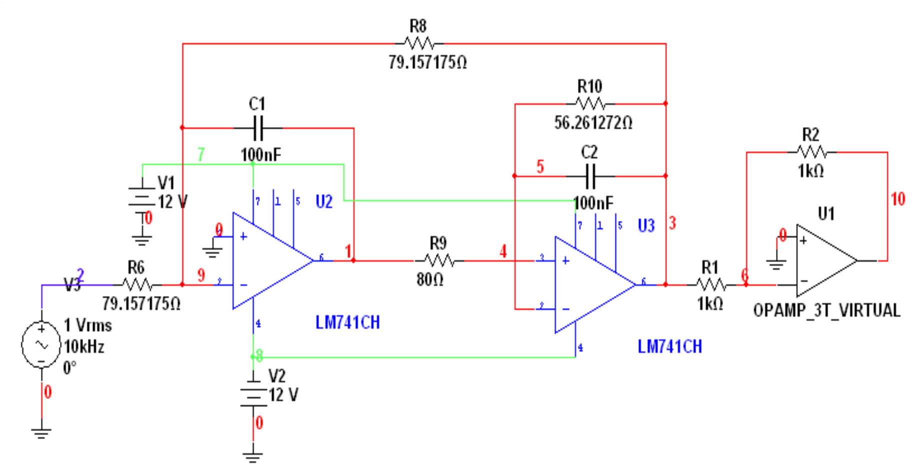

Fig.(10): The designed second order audio filter.

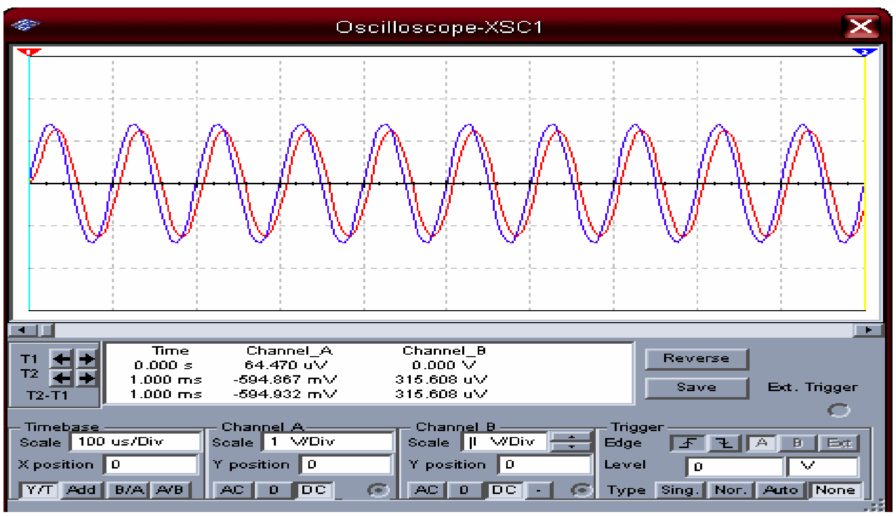

Fig.(11): Input & output signal without phase shift.

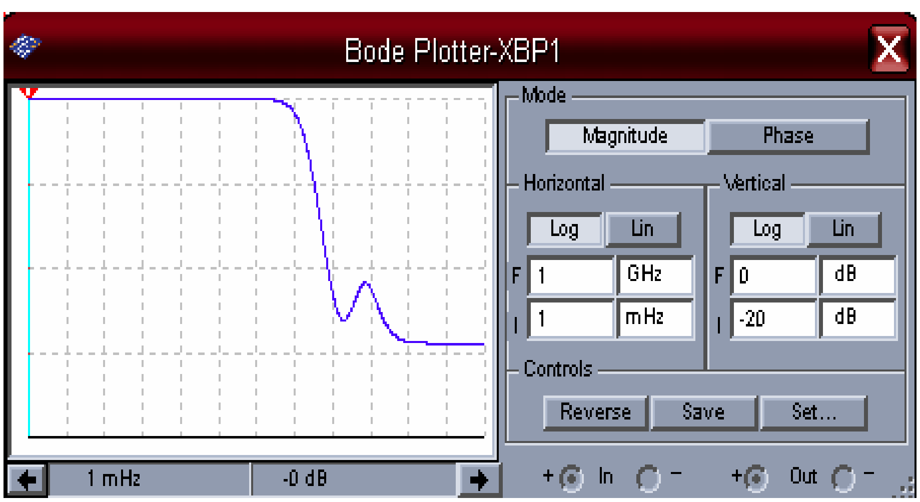

Fig. (12): The frequency response of the designed filter.

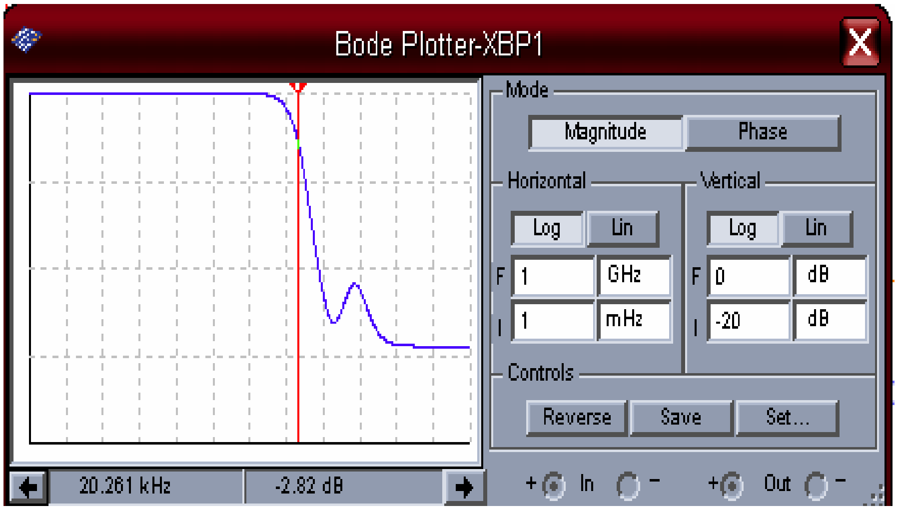

Fig.(13): Practical 3db cutoff frequency calculated .

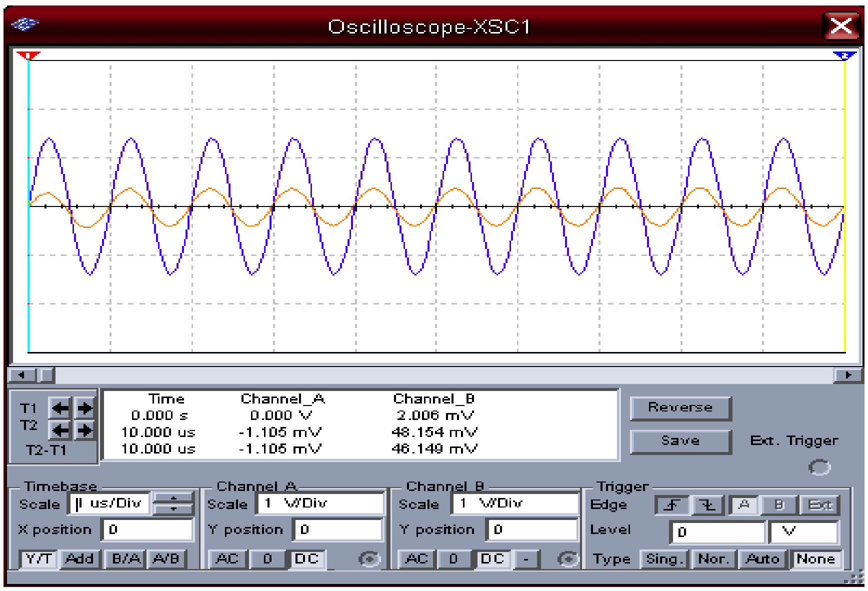

Fig.(14): Input signal & output attenuated sig.

تصميم و محاكاة مرشح تماثلي فعال من الدرجة الثانية لمرور الترددات الصوتية

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الخلاصة

تعتبر المرشحات من العناصر المهمة جدا الدوائر الالكترونية, هذه المرشحات لها دور رئيسي في أجهزة الإلكترونيك و الاتصالات وأنظمتهما . باستخدام هذه المرشحات الإجراءات الصائبة يمكن اتخاذها اعتمادا على التطبيقات التي استخدمت فيها. هذه الإجراءات التي اعتمدت ربما لها علاقة بالترددات والمقدار ونوع الإشارة ودالة عمل التطبيقات.

في هذا البحث , تم تصميم ومحاكاة مرشح تماثلي فعال لمرور ترددات الصوتية kHz(20 - 0), نفذ التصميم باستخدام تقنية منصة العمل الالكترونية. نتائج المحاكاة المستحصلة من تنفيذ المرشح المصمم كانت تقريبا مطابقة مع النتائج المستحصله من التحليل النظري للتصميم.هذه النتائج بينت أن المرشح المصمم قد اجتاز الاختبار بنجاح.