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MULTIBAND GRAPHIC EQUALIZER BASED ON AMPLIFYING STAGES WITH THE INCLUSION OF A BANDPASS FILTER IN THE FEEDBACK LOOP

In this paper, we calculated the elements of an equalizer based on an operational amplifier with a filter included in the feedback loop based on fundamental works in this area. The number of bands was chosen to be eight, with a tone control range of +15 dB. The buffer stage, the summing stage, and the parameters of their elements were also selected. The model was built in the Multisim program, and computer simulations were performed to obtain the amplitude-frequency and phase-frequency characteristics of each link and the equalizer as a whole. The 560 Hz link was considered separately at the minimum and maximum potentiometer positions (20 kΩ), and a Bode analysis was performed, which clearly shows the shortcomings of this circuit, namely the uneven adjustment range in the direction of signal attenuation. It is advisable to use high-quality capacitors with good parameter stability (in terms of capacitance and temperature). If electrolytic (oxide) capacitors are used in series, they should be connected in opposite order (plus to plus or minus to minus).

Operational amplifiers (op amps) can be used in any way suitable for high-end audio equipment. Here, the main attention should be paid to low noise levels and the high value of the input voltage rise rate. Power to the op-amp should be supplied from a stabilized, bipolar power supply. The voltage and current of the power supply depend on the specific type of chips used; we use TL084CN op amps with a bipolar supply voltage of 12 V. The equalizer has the name "graphic" because the slider controls used in it (instead of knobs) show the frequency response of the output signal by their position. You can control the entire frequency range by applying several bandpass filters, constructively assembled in one case. Each of these filters will be tuned to a specific frequency. Typically, the individual filters are placed (spaced) either an octave, half an octave, or a third of an octave apart so that when they are all in the gain state (or all in the attenuation state), the frequency response is equal. Movements usually have a (mark) that makes it easy to determine the center position.

Key words: *equalizer, amplifier, filter, spectrogram, signal summing, bandpass, feedback loop, Q-factor.*

Formulation of the problem. Equalizers are deservedly popular among sound reproduction enthusiasts. Only these devices make it possible to significantly change the quality of an acoustic sound signal and thereby correct some “imperfections” in the signal source – amplifier – speaker path, taking into account the individual perception of a particular listener [1]. By adjusting the equalizer’s transmission coefficient at selected frequency intervals of the audio signal, you can achieve improved sound reproduction even for mid-level devices, including monaural designs [2]. There is interference and resonance in every room – as a result, some frequencies of the acoustic range sound louder, and others quieter than planned. The characteristics and even the location of the loudspeakers are also important. The described eight-band simple equalizer allows you to correct these shortcomings and obtain acceptable sound processing characteristics in a certain range.

Analysis of recent research and publications. In the 1930s, John Volkman used an equalizer to change the sound of cinema sound systems and introduced equalization into sound reinforcement systems. In 1967, he developed the first set of passive 1/3-octave filters called Acousta-Voice, which marked the beginning of a new era of modern equalization. The next 20 years can be characterized by an incredible rise in the development of equalizers: a wide variety of these devices were created using microcircuits and other digital technologies. The first equalizer similar to modern ones was invented by Peter Bexendahl [3]. The research of filters to create better equalizers is ongoing, for example, in [4], the development of a graphic equalizer based on interpolated filters with a finite impulse response (IFIR) is presented, and in [5] the parallel equalizer is improved by using bandpass filters, the interaction of which with two neighboring filters at their center frequency is precisely controlled, it is called a cascade. In addition, there are ways to improve mathematical models, so in [6], the interaction matrix is created with different gains for each bandpass filter, which helps to maintain the maximum approximation error below 1 dB at and between the center frequencies when the gains of neighboring commands are the same.

Although the topic under study is not new, no illustrative calculations of a simple equalizer with summation and the resulting design were found in the form of an article, so the results of the study will be demonstrated below.

Task statement. Calculate the active filters of each link and simulate the resulting equalizer circuit based on the addition of band-pass filter signals to obtain the frequency response and frequency response of the device, perform Bode analysis, evaluate the quality of the resulting model, and draw conclusions about such a construction.

Methods. To calculate the equalizer based on an operational amplifier with the filter included in the feedback loop, we first set the tone control range in all frequency bands to $\pm$15 dB without the mutual influence of the filters on each other, so it will be advisable to include each band-pass filter in a parallel negative voltage feedback loop of the operational amplifier [1]. Figure 1 shows the inclusion of filters in the negative feedback loop of the adder using the example of one filter.

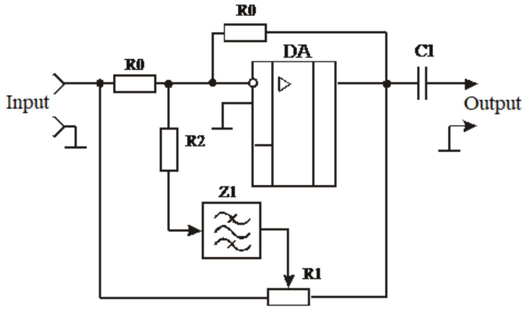


Fig. 1. Scheme of active filter inclusion in the feedback loop of the summing stage

All subsequent calculation methods are derived from [1; 2]. The gain of the active filter at the resonant frequency, provided that the optimal parameters of such a link, as well as the filter itself, are obtained, should be equal to two. If this condition is met, the gain of the circuit at the resonant frequency is equal to (1):

$$K_c = \frac{(2R_0 + R_2 - 2R_0 n)}{(2R_0 n + R_2)}, \tag{1}$$

where K_c is the adder's transmission coefficient; R_0 and R_2 are the values of the corresponding resistances; n is the coefficient characterizing the position of the potentiometer R_1 (in the leftmost position n = 0, in the rightmost position n = 1).

Given the previous expression, the maximum (2) and minimum (3) adder gains are equal:

$$K_{c\max} = \frac{(2R_0 + R_2)}{R_2}; \tag{2}$$

$$K_{c\min} = \frac{R_2}{(2R_0 + R_2)}. \tag{3}$$

When choosing the recommended values [2] of the resistors, note that the input resistance of the filter (Figure 1) will be equal to the resistance R_2 and will be 5.1 kΩ. Then, by selecting the resistor R_0, we will achieve the required control factor. Then R_0=13 kΩ. The dynamic range is determined by (4):

$$D_p = \frac{K_{c\max}}{K_{c\min}} = \frac{(2\cdot 13 + 5{,}1)}{5{,}1} / \frac{5{,}1}{(2\cdot 13 + 5{,}1)} = 37{,}18(times) \approx 31{,}4(dB). \tag{4}$$

The capacitance C_1 (Figure 1) is calculated as follows (5), so that it introduces distortion at lower frequencies of 1.5 dB at a resistance of 600 Ohms:

$$C_1 = \frac{1}{2\pi \cdot R_i \cdot F_i \cdot \sqrt{M_i^2 - 1}} = \frac{1}{6{,}28 \cdot 50 \cdot 600 \cdot \sqrt{1{,}18^2 - 1}} = 8{,}26 \text{ (uF)}. \tag{5}$$

Let's calculate the required number of bands in the case of an octave equalizer. The step factor for the octave filter is equal to (6), and it is also known that for the octave filter (7):

$$K = \left(\frac{f_{\max}}{f_{\min}}\right)^{1/N}; \tag{6}$$

$$K = 2^{1/M} = 2, \tag{7}$$

where N is the number of control bands, and M is the number of bands per octave (M=2 according to the above). Then, taking into account expression (7) and going to expression (5), we have (8):

$$N = \frac{\lg\left(\frac{f_{\max}}{f_{\min}}\right)}{\lg K} = \frac{\lg 250}{\lg 2} = 7.9 \approx 8 \text{ (bands)}. \tag{8}$$

Let's calculate the values of the central or so-called average frequencies of the filters using formula (9) and put them in Table 1:

$$F_{0i} = \left[F_{\max}^{\frac{2i-1}{2N}}\right]\left[F_{\min}^{\frac{2(N-i)+1}{2N}}\right] \text{ (Hz)}, \tag{9}$$

where F_{0i} is the average frequency of the i-th band; F_{min}, F_{max} are the upper and lower limit frequencies, selected within 50 Hz – 12.5 kHz, respectively; N is the number of frequency bands.

The Q-factor at the maximum rise or fall of the frequency response is selected from condition (10):

$$Q = \frac{\sqrt{k}}{k-1} = \frac{\sqrt{2}}{2-1} = 1,41, \qquad (10)$$

where k is the step coefficient of the filter.

It is easy to verify the correctness of the calculation of the average filter frequencies, which was carried out according to expression (9) since the obtained values of neighboring frequencies differ by a factor of two, which indeed corresponds to k=2.

The application of active RC filters, especially when used as active elements of operational amplifiers, is suitable for the above-calculated parameters. The simplest example of such a filter is a filter based on a low-Q bandpass filter, the schematic diagram of which is shown in Figure 2 [2]:

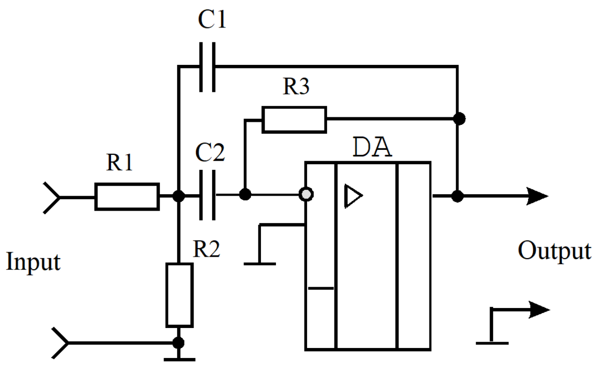


Fig. 2. Schematic of a low-pass bandpass RC filter on an operational amplifier

Although this filter belongs to the class of low-Q filters, it allows you to achieve a Q factor of two, which is higher than the required calculated value.

Let's calculate the equalizer filters, knowing their resonant frequencies (from Table 1). The initial data is the same for all filters: Q = 1.4; gain at the resonant frequency: $K_p = 2$; the value of the resistor R_3, the same for all filters, is chosen arbitrarily and is assumed to be 15 kΩ [2]; resistor R_1 is calculated by the formula (11):

$$R_1 = \frac{Q \cdot R_3}{2 \cdot K_p} = \frac{1,4 \cdot 15000}{2 \cdot 2} = 5250\ (\Omega). \qquad (11)$$

The values of the resistor R2 and capacitors are calculated by the following expressions (12) and (13):

$$R_{2i} = \frac{1}{(2 \cdot Q - 1) \cdot 2\pi \cdot F_{0i} \cdot C_{1i}}; \qquad (12)$$

$$C_{1i} = C_{2i} = \frac{2 \cdot Q}{2\pi \cdot F_{0i} \cdot R_3}. \qquad (13)$$

Table 1

The values obtained after the calculations

Filter	**1**	**2**	**3**	**4**	**5**	**6**	**7**	**8**
F_{0i}, Hz	71	141	281	560	1116	2226	4439	8852
C, uF	0,5	0,22	0,1	0,056	0,033	0,015	0,0068	0,0033
R_2, kΩ	3,48	4,02	4,42	4,42	3,32	3,74	4,12	4,22

The buffer stage should be an op-amp inverter on an op-amp. Initial data: Gain: K = 1 (0 dB); Nonlinear distortion in the lowest frequency region: M_l = 1.18 (1.5 dB); Higher frequencies: M_h = 1.41 (3 dB); Input impedance: R_{in} = 10 kΩ; Operating frequency band: 50 Hz – 12 kHz; Bipolar power supply: $\pm$12B.

Considering the initial data, the buffer cascade will be performed according to the scheme shown in Figure 3 [1].

In this scheme, its input resistance is determined by the resistor R_1, so you can take $R_1 = R_{in} = R_1$ = 10 kΩ.

The value of the capacitor C_1 will be calculated taking into account the nonlinear distortion in the lower frequency region (14) and the higher frequencies of C_2 (15):

$$C_1 = \frac{1}{2\pi \cdot F_{min} R_{in} \sqrt{M_l^2 - 1}} = \frac{1}{6,18 \cdot 50 \cdot 10^4 \cdot \sqrt{1,18^2 - 1}} = 0,5084\ \text{(uF)}. \qquad (14)$$

$$C_2 = \frac{\sqrt{M_h^2 - 1}}{2\pi \cdot F_{max} \cdot R_2} = \frac{\sqrt{1,41^2 - 1}}{6,18 \cdot 12,5 \cdot 10^7} = 1261,1\ \text{(pF)}. \qquad (15)$$

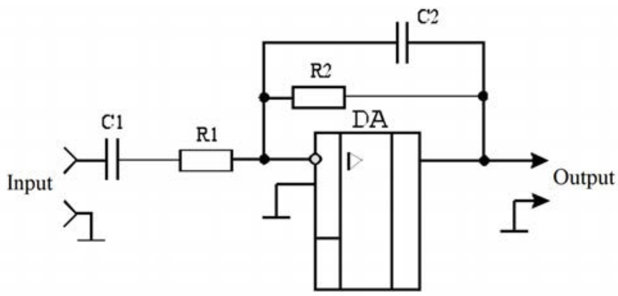


Fig. 3. Schematic diagram of the buffer cascade

Let's build a modeling circuit based on frequency-dependent feedback, which is realized using filters (Figure 4). For the task, we will use the TL084CN op-amp. To take the frequency response and frequency frequency response, the corresponding device will be connected to the output and input points of the filters.

To see if all the filters on the frequency axis have the same Q-factor and the correct location of the center frequencies, we represent them in the same coordinate system (Figure 5) for the frequency response similarly (Figure 6).

The overall frequency response and phase response of the equalizer will be as shown in Figure 7. To find out the depth of adjustment relative to the average level, we set the mini mum and maximum

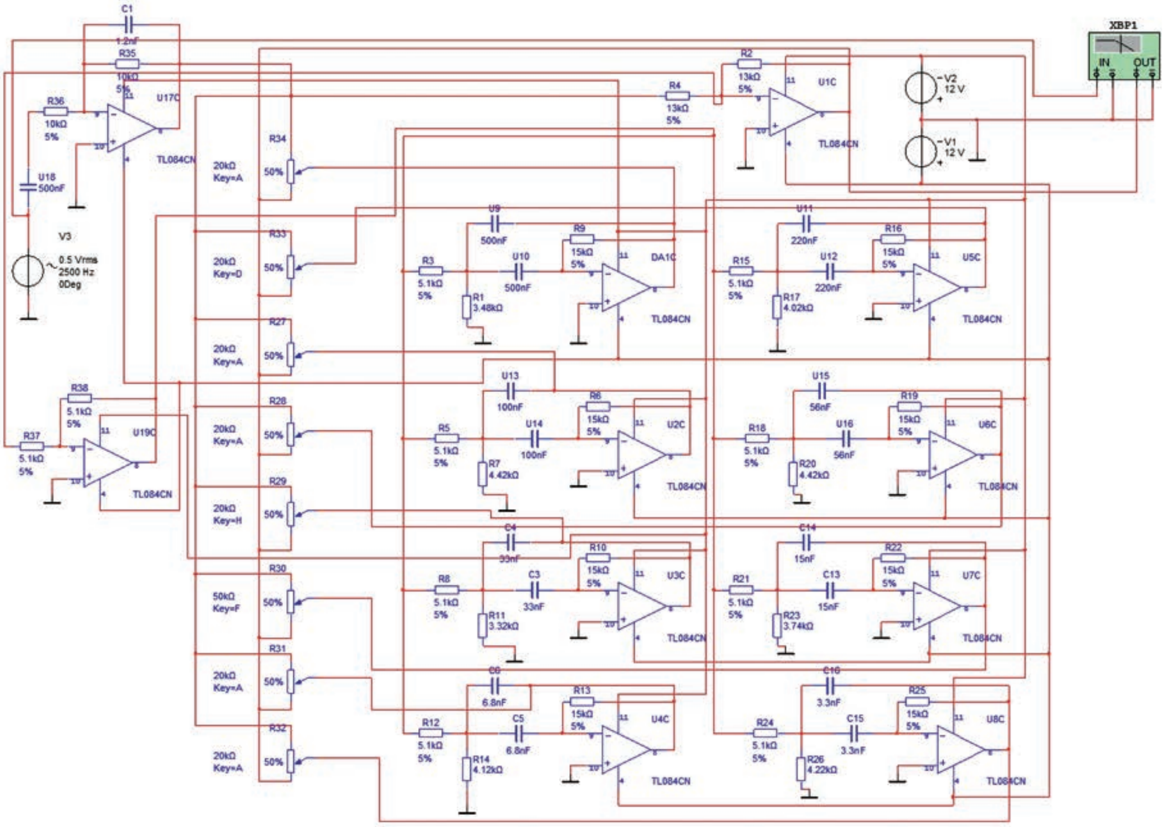


Fig. 4. Scheme prepared for modeling

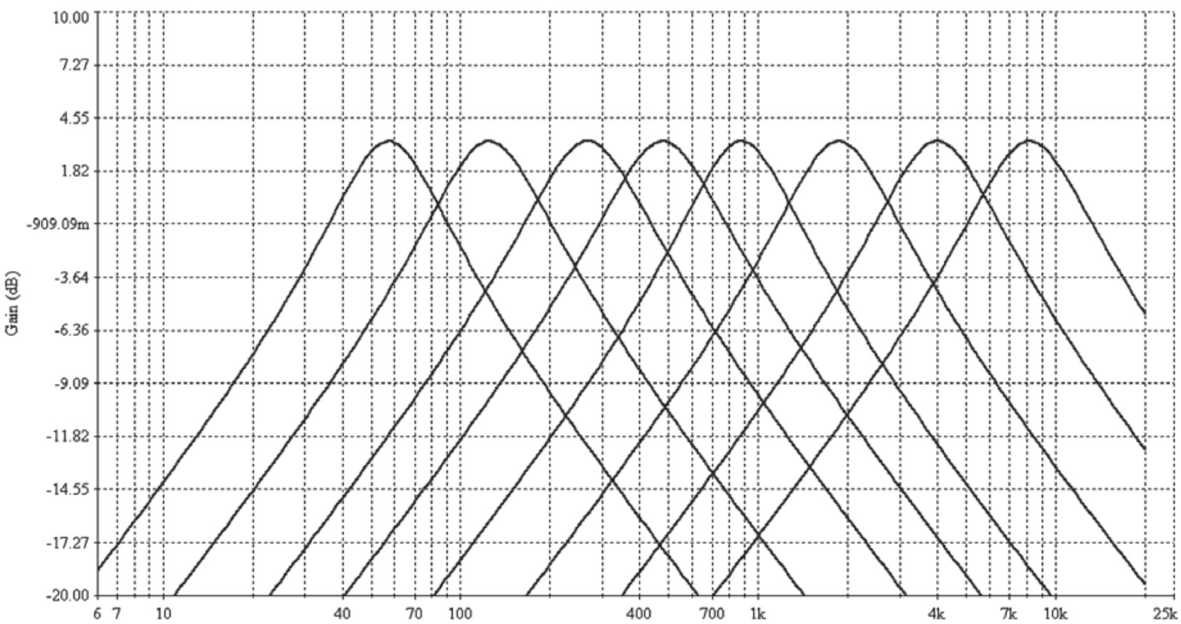


Fig. 5. Frequency response of all equalizer filters

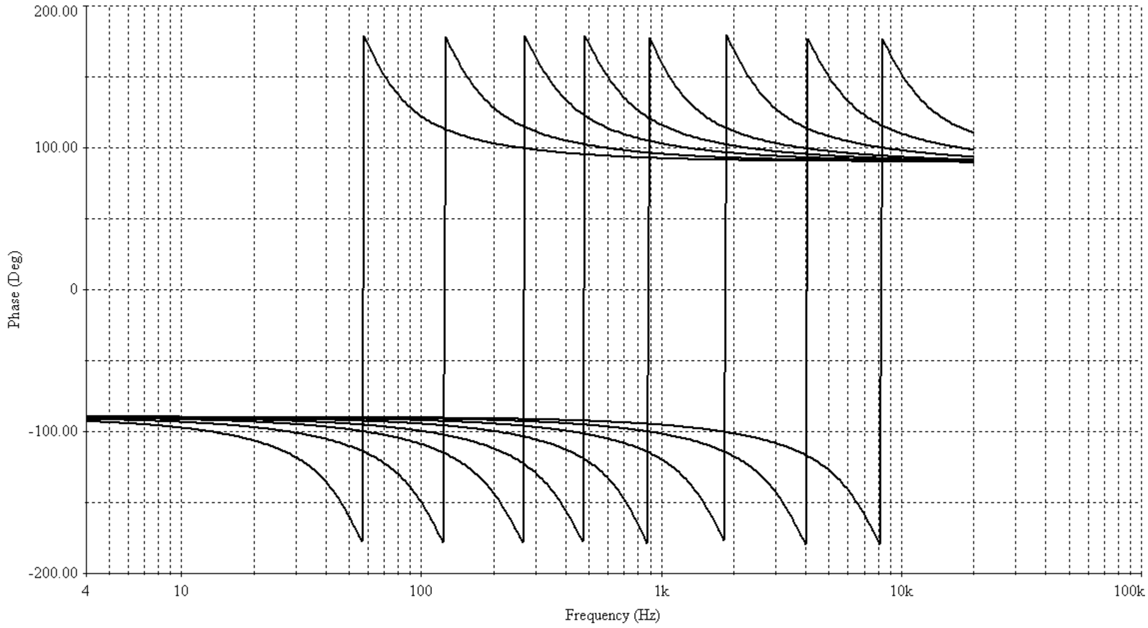


Fig. 6. Phase response of all equalizer filters

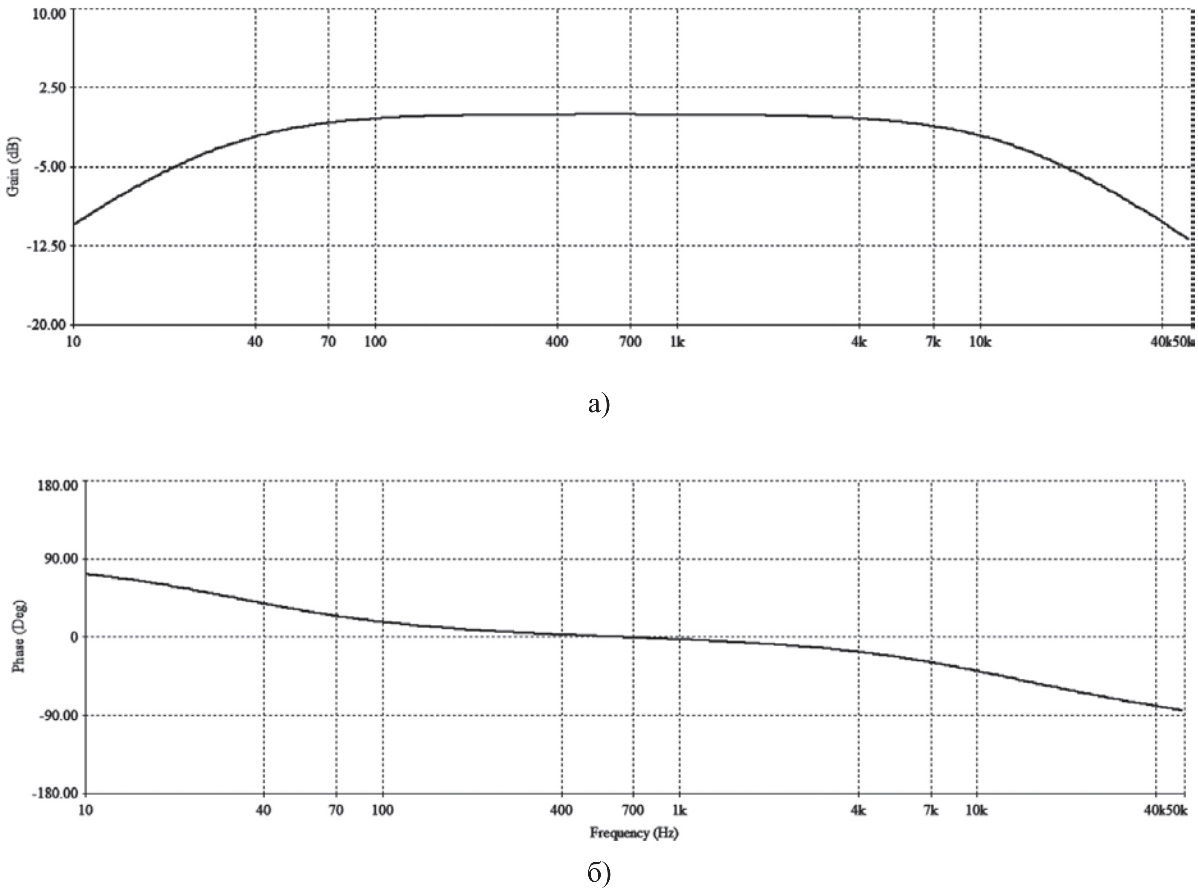
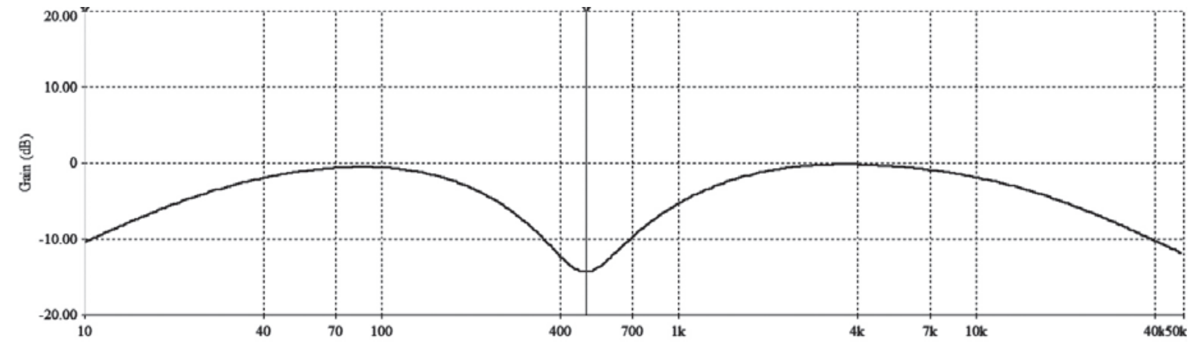
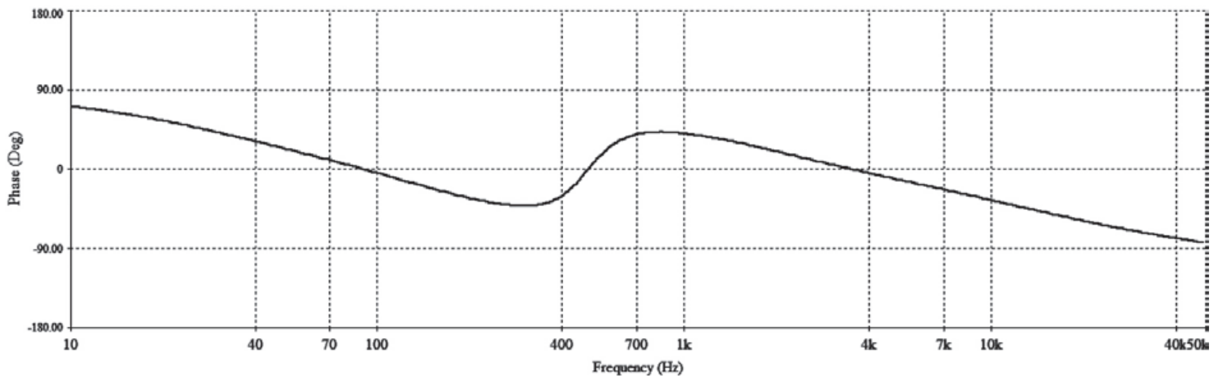


Fig. 7. Overall frequency (a) and phase response (b) of the equalizer

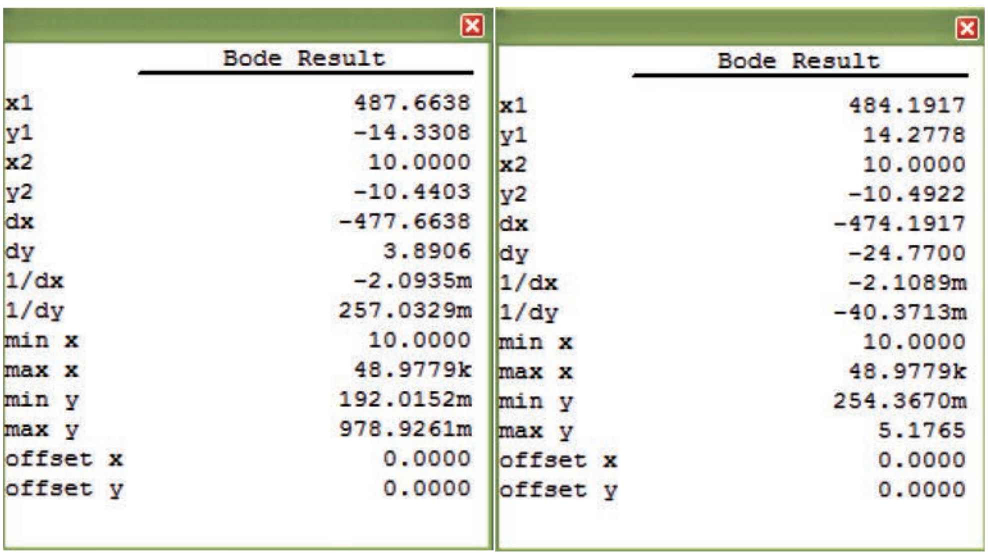


а)



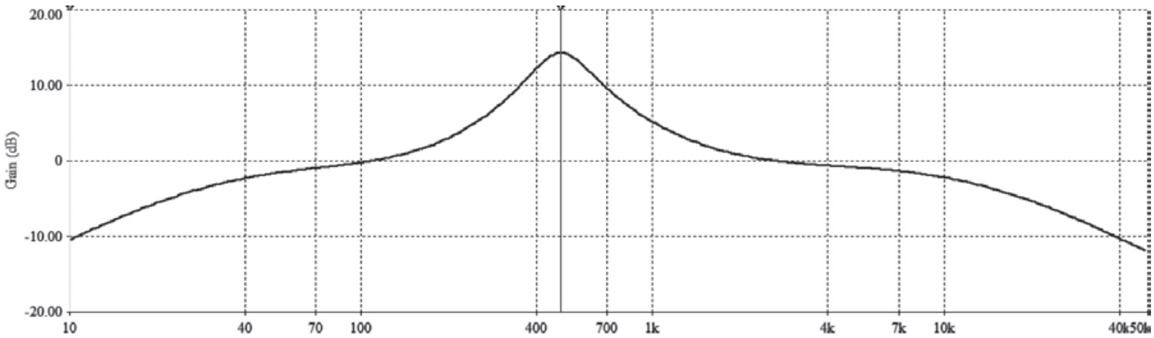
б)

Fig. 8. Frequency response phase response of the equalizer at the minimum position of the slider of one of the circuits (560 Hz)



a) b)

Fig. 9. Resulting values at the minimum (a) and maximum (b) slider positions



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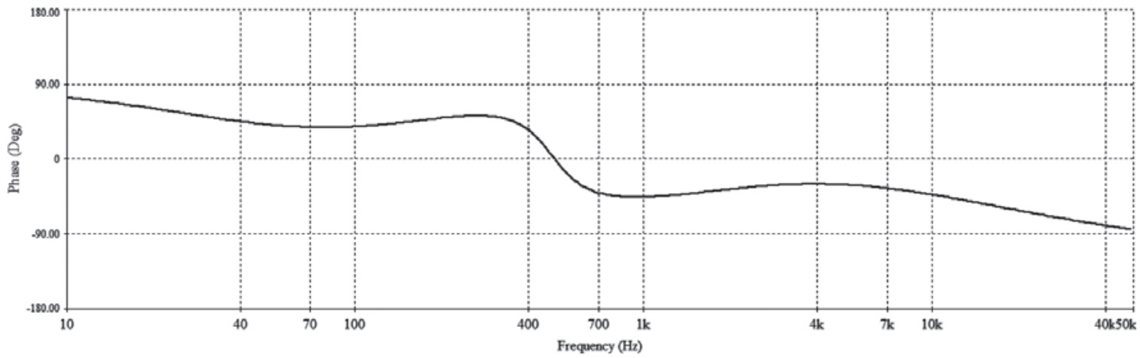

б)

Fig. 10. Frequency response and phase response of the equalizer at the maximum position of the slider of one of the circuits (560 Hz)

values of the potentiometer of one circuit, the results are shown in Figures 8 (a, b), 9 (a, b), 10 (a, b).

Conclusions. The simulation showed that the device does not work exactly according to the calculations. The errors are related to the elements selected from the standard element series, but in general, the filters, buffer stage, and summing stage were designed correctly. The Bode analysis indicates that the signal attenuation is -24.77 dB, and the gain is only 3.89 dB, which is a disadvantage, so comparing this result with the result obtained when building an equalizer based on an operational amplifier in [7], we can conclude that it is inexpedient to use the proposed circuit, using a gyrator, you can achieve greater quality factor, uniform depth of adjustment and smoother frequency response. Therefore, it is recommended to use gyrator circuits.

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Семенов А.О., Пінаєв Б.О., Хльоба А.А., Шурхал М.Ю., Ольхович В.М. БАГАТОСМУГОВИЙ ГРАФІЧНИЙ ЕКВАЛАЙЗЕР НА ОСНОВІ ПІДСИЛЮВАЛЬНИХ КАСКАДІВ ІЗ УВІМКНЕННЯМ СМУГОВОГО ФІЛЬТРА В КОЛО ЗВОРОТНОГО ЗВ'ЯЗКУ

У роботі проведено розрахунок елементів еквалайзеру на основі операційного підсилювача з увімкненням фільтру в коло зворотного зв'язку, ґрунтуючись на фундаментальних роботах в даному напрямку. Кількість смуг була обрана рівною восьми, діапазон регулювання тембру ±15 дБ. Також був обраний буферний каскад, сумуючий каскад, параметри їх елементів. Модель була побудована у програмі Multisim, проведене комп'ютерне моделювання для зняття амплітудно-частотної і фазочастотних характеристик кожної ланки і еквалайзеру в цілому. Окремо було розглянуто ланку на 560 Гц при мінімальному і максимальному положенні потенціометра (20 кОм), проведено аналіз Боде, на якому наочно видно недоліки даної схеми, а саме нерівномірний діапазон регулювання в сторону послаблення сигналу. Усі конденсатори в схемі бажано застосовувати високоякісні, з хорошою стабільністю параметрів (за ємністю і температурою). У разі послідовного вмикання електролітичних (оксидних) конденсаторів, їх слід вмикати зустрічно (плюс до плюса або мінус до мінуса).

Операційні підсилювачі (ОУ) можна застосувати будь-які, що підходять за параметрами для використання в звуковій апаратурі високого класу. Тут основну увагу слід приділити низькому рівню шумів і високому значенню швидкості наростання вхідної напруги. Живлення на ОУ слід подавати від стабілізованого, двополярного джерела живлення. Напруга і струм джерела живлення залежать від конкретного типу застосованих мікросхем, використовуються ОП – TL084CN з двополюсною напругою живлення 12 В. Еквалайзер має назву «графічний» тому, що застосовані в ньому движкові регулятори (замість ручок) своїм положенням показують частотну характеристику вихідного сигналу. Можна керувати всім частотним діапазоном шляхом застосування декількох смугових фільтрів, конструктивно зібраних в одному корпусі. Кожен із цих фільтрів буде налаштований на певну частоту. Зазвичай окремі фільтри розміщені (на відстані) або в октаву, або в половину октави, або в третину октави таким чином, щоб коли всі вони перебувають у стані посилення (або всі в стані ослаблення), частотна характеристика була рівна. Движки зазвичай мають (позначку), яка дає змогу легко визначити середнє положення.

Ключові слова: *еквалайзер, підсилювач, фільтр, спектрограма, підсумовування сигналів, смуга пропускання, зворотний зв'язок, добротність.*