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Projects of analogue active filters

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Keywords: filters, analog, active, lowpass filter, highpass filter

SUMMARY:

The starting point for the article was the fact that it is not possible to use digital filters in every application, therefore active analogue filter have been designed. The paper describes methods of designing active analogue filters, three main design methods are presented, such as Sallen-Key, Multiple Feedback and Linkwitz-Riley. The second part of the work presents two designs: low-pass filter and 1 kHz frequency limit high-pass filter. The objective was that filters should have a high attenuation in the stop band.

1. DESIGNING FILTERS

Wanting to design a filter, equipped with basic knowledge of filters and their characteristics, two basic methods of implementing the structure of active filters can be applied:
- Sallen-Key filter,
- Multiple Feedback Filters (MFB)

These methods allow designing filters of any type, i.e. low-pass, high-pass, band-pass, and high-order band-stop filters.

1.1 Sallen-Key

Sallen-Key configuration, also known as Voltage Controlled Voltage Source (VCVS), was invented in 1955 at MIT Lincoln Labs by R. P. Sallen and E. L. Key. It is one of the most widely used topologies for building active filters, because it is primarily designed for simplicity. Another aspect of the popularity of this solution is the fact that the operational amplifier is configured as an amplifier, not an integrator, which minimizes the bandwidth of the gain. This, for a given operational amplifier, means that we can design a higher frequency filter than for other structures, because the filter gain does not affect its efficiency, as if it was configured as an adder. The phase of the signal passing through the filter is essentially maintained, since it is connected to the input of the non-inverting phase operational amplifier. Frequency and quality factors are quite independent, although they are sensitive to the signal gain parameter. The structure of such a filter is shown in diagram (Fig. 1).

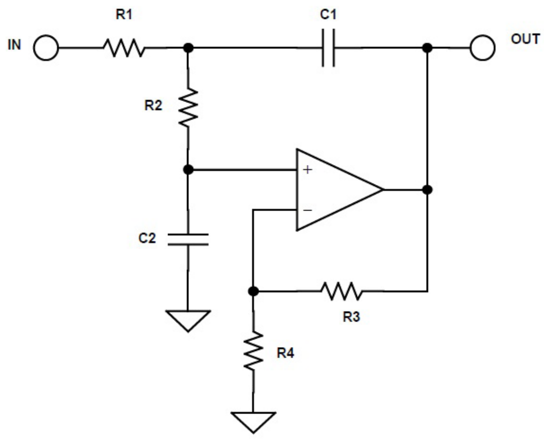

Figure 1 A Sallen-Key low-pass filter design

This is implementation of a second order of active low-pass filter. Resistors R1, R2 and capacitors C1, C2 are responsible for the border frequency of this filter, while R3 and R4 resistors are responsible for signal amplification. Another feature of this configuration is that the ratio of the highest value of the resistor to the smallest one and the ratio of the largest value of the capacitor to the smallest one is relatively low, making it profitable in production.

Even though Sallen-Key filters are widely used, they also have drawbacks. One of them is that they are not easily tuned due to the interaction of the components of the F_0 frequency and the quality factor of the system Q [1, 2].

1.2 Multiple Feedback

Multiple feedback filters use operational amplifiers as integrators, which makes the dependence of the transmittance on the amplifier parameters greater than in Sallen-Key configuration. In filters of this topology, it is difficult to generate a high quality factor and high frequency section due to the open loop constraints of the operational amplifier.

The rule is that the amplification in the open loop of the operational amplifier should be at least 20 dB (x10) above the amplitude of the response to the resonant or cut-off frequency, including the boost achieved by the quality factor of the filter. The peak value of amplitude in relation to the quality factor is expressed by the formula:

$$A0 = H*Q \quad (1)$$

where H is the amplification of the system. The multi-feedback filter reverses the phase, so it is equivalent to adding a 180° constant to the phase shift. The ratio of the highest value of the elements to the smallest one is much larger in the multiple-feedback filters than in the Sallen-Key [2] method. Filters higher than fourth-order may turn out to be useful due to the increasing peak latency in the group of frequencies oscillating around f_0 [3].

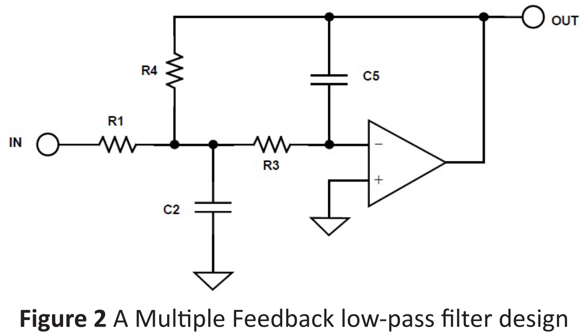

Figure 2 A Multiple Feedback low-pass filter design

1.3 Linkwitz-Riley

One of Linkwitz-Riley filter's application is in audio crossovers. It was designed by Siegfried Linkwitz i Russ Riley, who originally described audio crossover networks in JAES in February 1976. This filter is also known as Butterworth squared filter. L-R filters are usually designed as a cascade of two Butterworth filters, each with -3 dB amplification at cutoff frequency, as a result of which L-R filter has -6 dB amplification for this frequency [8].

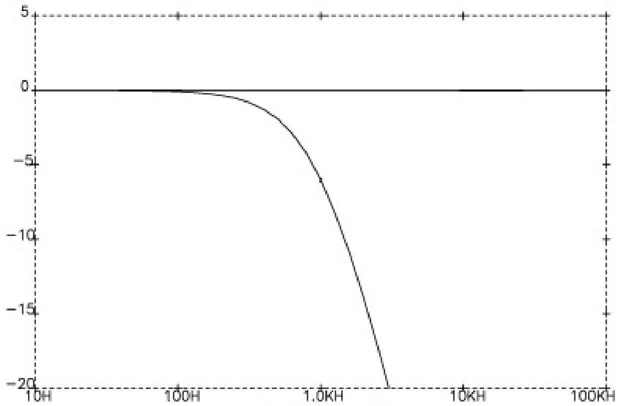

Figure 3 Sample characteristics of L-R filter with border frequency 1 kHz

L-R filters make use of active Sallen-Key structure to achieve second order filter.
Frequency response is defined by ω_0 i Q_0.

$$\omega_0 = 2\pi f_0 \quad (2)$$

$$\omega_0 = \frac{1}{R\sqrt{C_1C_2}} \quad (3)$$

$$Q_0 = 0{,}5\sqrt{C_1C_2} \quad (4)$$

$$R = \frac{1}{2Q_0\omega_0C_2} \quad (5)$$

$$C_1 = 4Q_0^2C_2 \quad (6)$$

Each order of L-R filter can be implemented through a cascade second order Sallen-Key modules. Q_0 values for each stage are shown Table 1. Values of particular components for each stage at a given cutoff frequency can be calculated with use of Q_0 and a convenient value of C_2 or R_2 in the above formulas. Filters of order higher than the fourth one can prove unsuitable in view of rising latency peak within a range of frequencies close to f_0 [3].

Table 1 Quality factor values for a Linkwitz-Riley filter for particular orders

	LR2	LR4	LR6	LR8	LR10
Q_0 section 1	0.5	0.71	0.5	0.54	0.5
Q_0 section 2		0.71	1.0	1.34	0.62
Q_0 section 3			1.0	0.54	1.62
Q_0 section 4				1.34	0.62
Q_0 section 5					1.62
dB/slope per octave	12	24	36	48	60

2. FILTER PROJECTS

The aim of the work was to design two different filter types:
- a low-pass filter with a limiting frequency of 1 kHz
- a high-pass filter also with a limiting frequency of 1 kHz.

It was decided that both filters will be constructed according to Sallen-Key methodology.

2.1 Low-pass filter

As assumptions for the design of this filter, the following conditions have been adopted:
In the passband the maximum allowed attenuation is 3dB at 1KHz, while for filtered frequencies attenuation is -40dB per octave.
In addition, the signal should have no gain in the operational band and should be optimized for low power consumption.
The power supply of this filter should be at least+ Vs = 5 V, -Vs = 5 V.
Additionally, it was assumed that the resistor tolerances should be 1%, while for capacitors 5%.
The filter which fulfils the above mentioned assumptions is the eight-order Linkwitz-Riley filter. Such a high value of order is justified in that it should be allowed to obtain amplitude-frequency characteristics with a high drop.
Below is a schematic diagram (Fig. 4a) of connections for such a filter along with the indicated equipotentiality of the system.

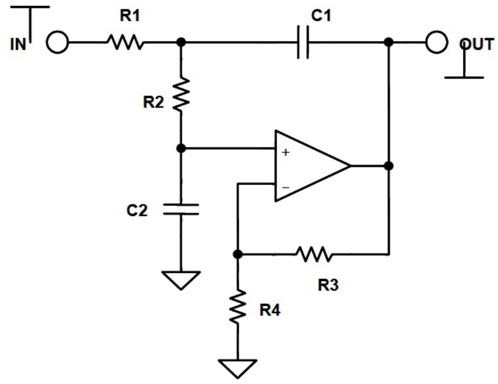

Figure 4a A Linkwitz-Riley filter design

An element group consisting of resistors, capacitors and operational amplifier is responsible for the implementation of two orders of low-pass filter, so the designed filter will consist of 4 such blocks. R3 and R4 resistors will be omitted because they are responsible for possible amplification of the signal.

Despite exclusion of the resistors, the system will work correctly, assuming that in the operational amplifier the feedback loop is closed, as shown in the final Figure 4b.

The following formulas show the system's transmittance (the ratio of the output signal to the input signal):

$$G(s) = \frac{+H\omega_0^2}{s^2 + \alpha\omega_0 s + \omega_0^2} \qquad (7)$$

$$\frac{V_0}{V_{IN}} = \frac{H\frac{1}{R_1R_2C_1C_2}}{s^2 + s\left[\left(\frac{1}{R_1} + \frac{1}{R_2}\right)\frac{1}{C_1} + \frac{(1-H)}{R_2C_2}\right] + \frac{1}{R_1R_2C_1C_2}} \qquad (8)$$

It was assumed for the calculations that the values of resistors will be equal.

Selecting resistor R = 160 kOhm and using the formulas before, one can calculate the remaining values of the elements.

The values obtained for the construction of this filter are:

a) For Q = 0.54 (stages one and three)

R = 160 kOhm
C_1 = 0.91 nF
C_2 = 1.07 nF

b) For Q = 1.34 (stages one and three)

R = 160 kOhm
C_1 = 370 pF
C_2 = 265.5 pF

After calculating values of resistors and capacitors, the elements were combined into stages, each according to the scheme (3) shown earlier. Next, the stages were connected in series one after the other. If there was a need to amplify the signal, a branch consisting of elements R3 and R4 should be used.

Selection of the second resistor can be determined by the following formula:

$$R_4 = \frac{R_3}{(H-1)} \qquad (9)$$

2.2 High pass filter

As assumptions for the design of this filter, the following conditions have been adopted:

In the passband the maximum allowed attenuation is 3dB at 1KHz, while for filtered frequencies attenuation is -40dB at 500 Hz.

In addition, the signal should be characterized by a lack of gain in the operational band and should be optimized for low power consumption.

The power supply of this filter should be +Vs = 5 V, -Vs = -5 V.

In addition, it has been assumed that the resistances for resistors can be 1%, and for capacitors 5%.

The filter which fulfils the above mentioned assumptions is the seventh-order Butterworth filter.

Such a high value of order is justified in that the author wanted to achieve a large fall in amplitude characteristics, and the Buterwerworth filters are characterized by the dependence of a 6 dB drop on the octave multiplied by filter order n.

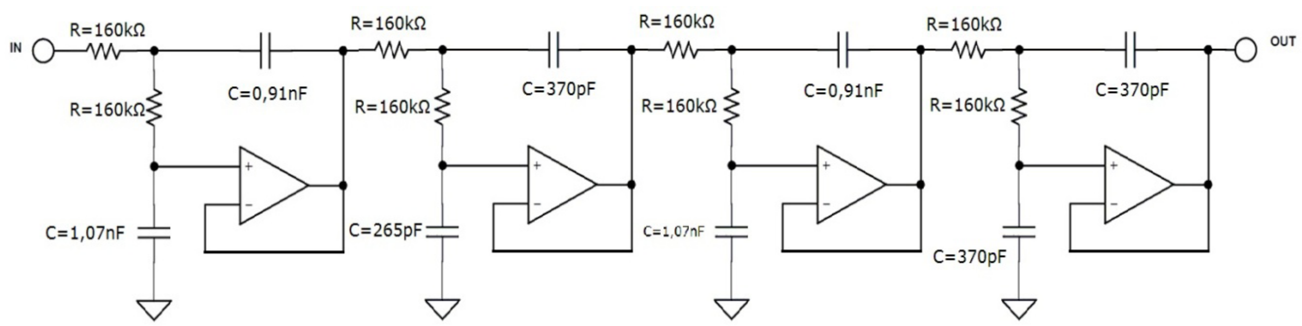

Figure 4b Final design of a low-pass eighth-order filter

Thanks to such an order value, the filter allows for steep characteristics of amplitude-frequency in the transient band.
Below is a wiring diagram for such a filter.

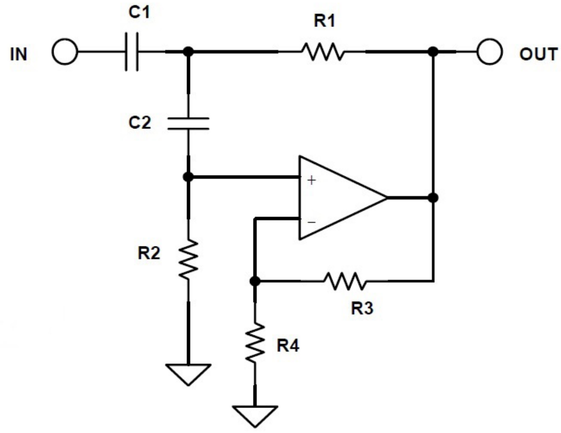

Figure 5 A Sallen-Key high-pass filter design

The transmittance of the system is expressed by the following formula:

$$\frac{V_0}{V_{IN}} = \frac{Hs^2}{s^2 + s\left[\dfrac{\dfrac{C_2}{R_2} + \dfrac{C_1}{R_2} + (1-H)\dfrac{C_2}{R_1}}{C_1C_2}\right] + \dfrac{1}{R_1R_2C_1C_2}} \quad (10)$$

By selecting the C1 capacitor, one can calculate what the values of the other elements are.

$$k = 2\pi F_0 C_1 \quad (11)$$

$$C_2 = C_1 \quad (12)$$

$$R_1 = \frac{\alpha + \sqrt{\alpha^2 + (H-1)}}{4k} \quad (13)$$

$$R_2 = \frac{4}{\alpha + \sqrt{\alpha^2 + (H-1)}} * \frac{1}{k} \quad (14)$$

Table 2 Coefficient values for Butterworth filters [2]

Filter order	Section	Real number	Imaginary number	F_o		Q	Frequency -3 dB	Peak frequency	Peak value
2	**1**	0.7071	0.7071	1	1.4142	0.7071	1.0000		
3	**1**	0.5000	0.8668	1	1.0000	1.0000		0.7071	1.2499
	2	1.0000		1			1.0000		
4	**1**	0.9239	0.3827	1	1.8478	0.5412	0.7195		
	2	0.3827	0.9239	1	0.7654	1.3065		0.8409	3.0102
5	**1**	0.8090	0.5878	1	1.6180	0.6180	0.8588		
	2	0.3090	0.9511	1	0.6180	1.6182		0.8995	4.6163
	3	1.0000		1			1.0000		
6	**1**	0.9659	0.2588	1	1.9319	0.5176	0.6758		
	2	0.7071	0.7071	1	1.4142	0.7071	1.0000		
	3	0.2588	0.9659	1	0.5176	1.9319		0.9306	6.0210
7	**1**	0.9010	0.4339	1	1.8019	0.5550	0.7449		
	2	0.6235	0.7818	1	1.2470	0.8019		0.4717	0.2204
	3	0.2225	0.9749	1	0.4450	2.2471		0.9492	7.2530
	4	1.0000		1			1.0000		
8	**1**	0.9808	0.1951	1	1.9616	0.5098	0.6615		
	2	0.8315	0.5556	1	1.6629	0.6013	0.8295		
	3	0.5556	0.8315	1	1.1112	0.9000		0.6186	0.6876
	4	0.1951	0.9808	1	0.3902	2.5628		0.9612	0.3429
9	**1**	0.9397	0.3428	1	1.8794	0.5321	0.7026		
	2	0.7660	0.6428	1	1.5320	0.6527	0.9172		
	3	0.5000	0.8680	1	1.0000	1.0000		0.7071	1.2493
	4	0.1737	0.9848	1	0.3474	2.8785		0.9694	9.3165
	5	1.0000		1			1.0000		
10	**1**	0.9877	0.1564	1	1.9754	0.5062	0.6549		
	2	0.8910	0.4540	1	1.7820	0.5612	0.7564		
	3	0.7071	0.7071	1	1.4142	0.7071	1.0000		
	4	0.4540	0.8910	1	0.9060	1.1013		0.7667	1.8407
	5	0.1564	0.9877	1	0.3128	3.1970		0.9752	10.2023

Using Table 2 and earlier formulas, the high pass filter scheme shown in Figure 6 was obtained.

3. CONCLUSIONS

Thanks to the above-described methods, two analog filters were designed: a high-pass and a low-pass one, both with a border frequency of 1 kHz. Using the described methods, a high damping value was obtained in the blocking band and no damping in the transmission band. Filters designed in this way are characterized by a very narrow transition band, which is desirable feature for end users. Implementation of design assumptions was possible only with the application of active filters. The use of passive filters would result in partial signal attenuation in the pass band. Active analogue filters are much more efficient than passive filters, although an additional power supply for the operational amplifiers in the filter is required for their operation.

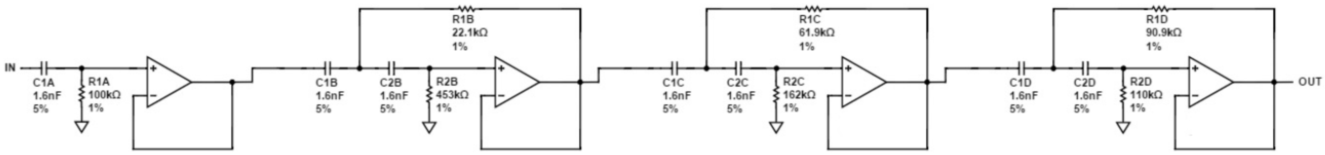

Figure 6 Final design of a high-pass seventh-order filter

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