

# **Active Filter Design**

by

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## ***Title page***

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## **1. Abstract**

This report describes how an active low pass filter is designed, simulated and tested. The filter is here constructed from some given specifications, where one of them is that the filter need to have a Butterworth response. The procedure we in this group have chosen to solve the assignment, is to design the filter with two different methods, one with a Sallen and Key architecture and the other is with the Multiple-Feedback (MFB) architecture.

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## 2. Introduction

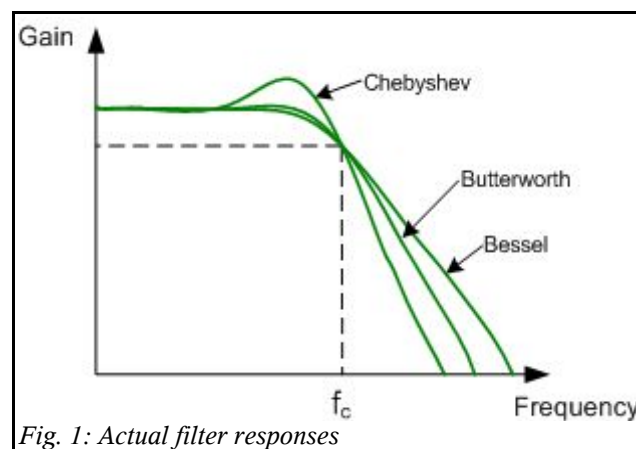
Today noise reduction is a very hot topic. In this case it is electrical noise in a digital communication system, that needs to be reduced. The purpose of reducing (band limit analogue signal) the noise is cause its not wanted in the following sampling and encoding process. Like in the given assignment, it is very common to insert some kind of filter in DSP applications, where a false input signal can be critical to the outcome on the output.

Long before Op-Amps were available electrical filters have been used. That was when only passive components such as resistors and capacitors could be used for the filters. Today the Op-Amps are low-cost and very reliable active devices, which is the reason for mainly using active filters today.

There are various types of active filters where some of them are named Chebyshev, Bessel and Butterworth. The design process of the filters can consist of different types of architectures, where the two most common are named Sallen and Key, and Multiple Feedback (MFB) architecture. In this assignment the filter is an active low-pass filter with a Butterworth response. To satisfy our own curiosity and examine the differences, we will design the Butterworth filter with both of the above described architectures.

### 2.1. Butterworth filters

The figure 1 below shows the actual responses for three different types of filters the Bessel, Chebyshev and the Butterworth filter. The Butterworth response is the one that is we are going to concentrate about in this report.



#### **The characteristics for the butterworth filter response are:**

- Maximal flat magnitude response filter.
- Optimised for gain flatness in the passband.
- -3dB at the cut-off frequency.
- -20dB per decade per order above the cut-off frequency.
- Transient response to a pulsed input shows moderate overshoot and ringing.

**Butterworth polynomials:**

Unlike other polynomials the Butterworth polynomials requires the least amount of work, because the frequency scaling scaling factor are always equal to one. The transfer function for the second order low-pass Butterworth filter are shown below.

$$H_{LP} = \frac{K}{-\left(\frac{f}{f_c}\right)^2 + 1,414 \cdot \frac{jf}{f_c} + 1}$$

**Where:**     K = the gain factor.  
                   f = the frequency variable.  
                    $f_c$  = the cut-off frequency.

The normalized Butterworth polynomials for the first and second order filter are then:

1.  $s + 1$
2.  $s^2 + 1,414 \cdot s + 1$

**Where:**      $s = j\omega$

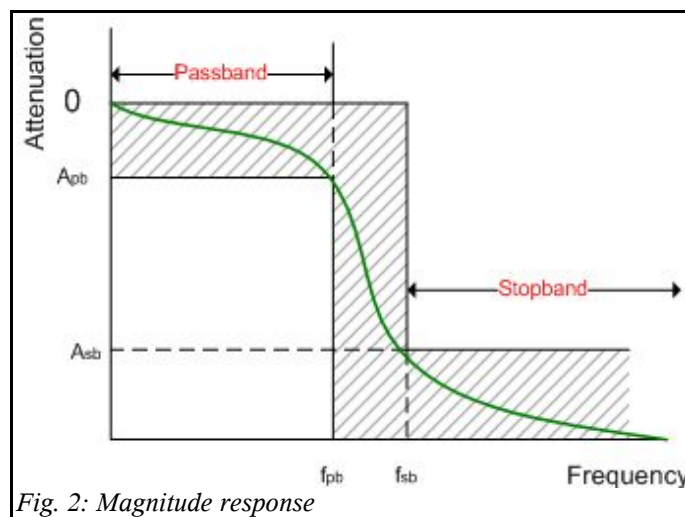
To make it easier to design active filters there are made some filter coefficient tables for each type of responses. Later in the Multiple Feedback architecture the table with values for the Butterworth response are shown and used in the calculations.

### 3. Assignment specifications

In a digital communication system an anti-aliasing low-pass active filter is required to band the limit of the analogue signal prior to sampling and encoding process. The filter must satisfy the following specifications:

- |                         |                      |
|-------------------------|----------------------|
| 1. Filter response      | = Butterworth.       |
| 2. -3dB frequency       | = 3,4kHz.            |
| 3. Stopband frequency   | = 6kHz.              |
| 4. Stopband attenuation | $\geq 10\text{dB}$ . |
| 5. Passband gain        | = 2.                 |

A visual indication of the magnitude response of the filter is shown in the figure 2 below:



The filter is to be designed and simulated, using TINA software development package, and investigate the performance of the filter. The simulation results must include a plot of the magnitude response of the filter. Construct the filter on a breadboard and investigate its performance. Plot the magnitude response of the filter on log paper.



## 4. Filter order

With the specifications for the Butterworth filter, it is possible to calculate the filter order. The result can then be used to show how many components are needed in the design of the active filter. Calculations are made with the formulas from the filter handout.

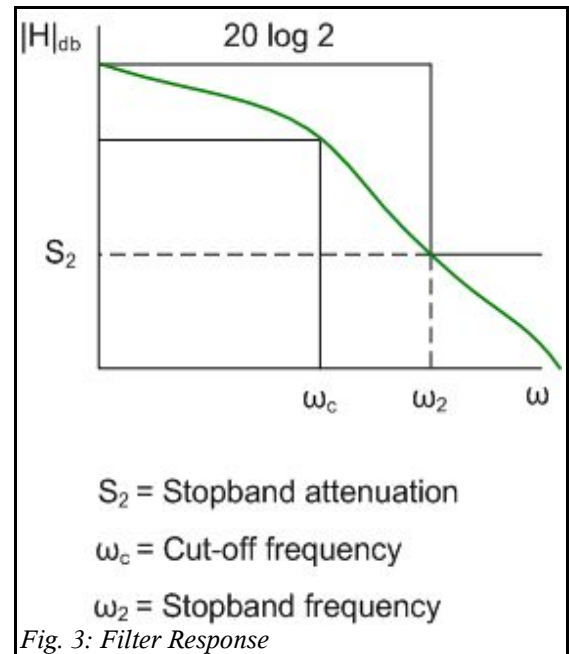
Figure 3 shows the filter response, where the  $S_2$ ,  $\omega_c$  and  $\omega_2$  from the specifications are needed to calculate the filter response. The  $20 \log 2$  explains the passband gain of 2.

**The main formula to calculate the filter order:**

$$n \geq \frac{\log\left(\frac{1}{S_2^2} - 1\right)}{2 \cdot \log\left(\frac{\omega_2}{\omega_c}\right)}$$

**The known values are:**

- $\Omega_c = 3,4\text{kHz}$ .
- $\omega_2 = 6\text{kHz}$ .
- $A_{sb} = 10\text{dB}$ .



**$S_2$  is found:**

$$20 \cdot \log(S_2) = -10 \text{ dB} \Rightarrow S_2 = \log\left(\frac{-10}{20}\right)^{-1} \Rightarrow \underline{\underline{S_2 = 316,2m}}$$

**The filter order is calculated:**

$$n \geq \frac{\log\left(\frac{1}{S_2^2} - 1\right)}{2 \cdot \log\left(\frac{\omega_2}{\omega_c}\right)} = \frac{\log\left(\frac{1}{316,2m^2} - 1\right)}{2 \cdot \log\left(\frac{6000}{3400}\right)} = 1,934 \sim \underline{\underline{n=2}}$$

The filter order of 2 means that the Butterworth filter now can be designed using one OP-amp for the Sallen-Key and the MFB architectures. When the filter order is 2, the number of resistors and capacitors that is needed for the filter function are also 2 of each.

## 5. Sallen and Key

The Sallen and Key architecture is the easiest way to design an active filter. It is the design that uses the least number of components, and the equations are relatively straight forward. It has been discussed, analysed and reviewed in great depth on the web and in text books. The Sallen and Key is very often a preferred design compared to the MFB architecture because it does not invert the signal. Another advantage of using a Sallen and Key architecture, is that it is not sensitive to component variation at unity gain. A disadvantage is the high frequency response of the filter, is limited by the frequency response of the amplifier.

### 5.1. Low-pass architecture

The simplest design of the Sallen and Key filter as a two-pole, low-pass filter is shown in figure 4 using TINA. Look aside from the component values, which will be calculated to the correct values shortly. From TINA the general ideal transfer function for this circuit can be derived (shown below). With this transfer function it is possible to calculate the values of the passive components. Simplified formulas derived from the transfer function, will for that purpose be used. -The transfer functions denominator will also be used later on to calculate the poles of the filter.

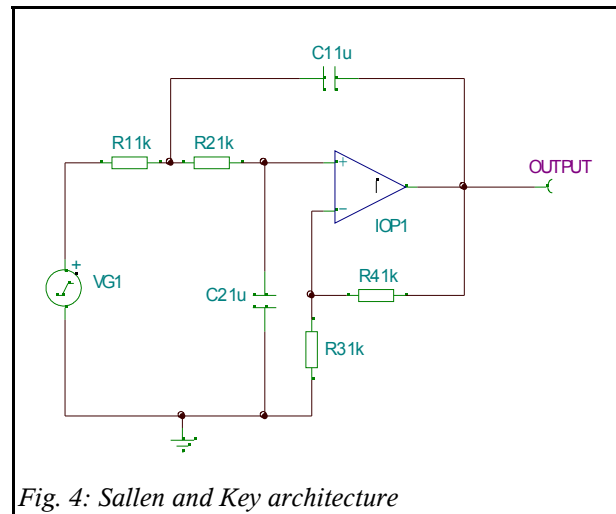


Fig. 4: Sallen and Key architecture

#### Low-pass Sallen and Key ideal transfer function:

$$\omega(s) = \frac{-R_3 - R_4}{-R_3 + (C_1 \cdot R_4 \cdot R_1 - C_2 \cdot R_3 \cdot R_2 - C_2 \cdot R_3 \cdot R_1) \cdot s - C_2 \cdot C_1 \cdot R_3 \cdot R_1 \cdot R_2 \cdot s^2}$$

### 5.2. Calculations

#### Chosen values:

- $C_1, C_2 = 2,2\text{nF}$ .
- $R_3 = 47\text{k}\Omega$ .

#### Impedance scaling factor is calculated:

$$M = \frac{1}{2 \cdot \pi \cdot \omega_c \cdot C} = \frac{1}{2 \cdot \pi \cdot 3400 \cdot 2,2\text{n}} = \underline{\underline{21,28\text{k}\Omega}}$$

#### $R_1$ and $R_2$ is calculated:

$$R_1 = \frac{1}{\sqrt{2}} \cdot M = \frac{1}{\sqrt{2}} \cdot 21,28\text{k} = 15,05\text{k} \sim \underline{\underline{15\text{k}\Omega}} \quad (\text{E12 value})$$

$$R_2 = \sqrt{2} \cdot M = \sqrt{2} \cdot 21,28\text{k} = 30,09\text{k} \sim \underline{\underline{33\text{k}\Omega}} \quad (\text{E12 value})$$

***R<sub>4</sub> is calculated with the gain formula:***

$$K = \frac{R_3 + R_4}{R_3} \Rightarrow R_4 = K \cdot R_3 - R_3 = 2 \cdot 47\text{ k} - 47\text{ k} \Rightarrow \underline{\underline{R_4 = 47\text{ k}\Omega}} \quad (\text{E12 value})$$

***Control calculation for the cut-off frequency with the E12 values:***

$$\omega_c = \frac{1}{\sqrt{R_1 \cdot R_2 \cdot C_1 \cdot C_2}} = \frac{1}{\sqrt{15\text{ k} \cdot 33\text{ k} \cdot 2,2\text{ n} \cdot 2,2\text{ n}}} = 20,43\text{ k} \frac{\text{rad}}{\text{sek}}$$

$$f_{\text{cut-off}} = \frac{\omega_c}{2 \cdot \pi} = \frac{20,43\text{ k}}{2 \cdot \pi} = \underline{\underline{3,25\text{ kHz}}}$$

***Calculations of the poles:***

From the transfer function the poles of the Sallen and Key filter can be derived. This is done with the help of Mathcad where the transfer function is inserted. The s in the denominator, that indicates the poles of the circuit can then be found by letting Mathcad calculate on the equation. The result for the Sallen and Key pole calculation are shown below.

***Poles for the ideal Sallen and Key circuit:***

$$-R_3 + (C_1 \cdot R_4 \cdot R_1 - C_2 \cdot R_3 \cdot R_2 - C_2 \cdot R_3 \cdot R_1) \cdot s - C_2 \cdot C_1 \cdot R_3 \cdot R_1 \cdot R_2 \cdot s^2 = 0$$

$$\underline{\underline{s = -15,1\text{ k} \pm j15,11\text{ k} \frac{\text{rad}}{\text{sek}}}}$$

***Poles for the non-ideal Sallen and Key circuit:***

$$-R_3 + (C_1 \cdot R_4 \cdot R_1 - C_2 \cdot R_3 \cdot R_2 - C_2 \cdot R_3 \cdot R_1) \cdot s - C_2 \cdot C_1 \cdot R_3 \cdot R_1 \cdot R_2 \cdot s^2 = 0$$

$$\underline{\underline{s = -15,15\text{ k} \pm j13,71\text{ k} \frac{\text{rad}}{\text{sek}}}}$$

### 5.3. Simulations

Simulations are made using TINA with calculated ideal and the non-ideal E12 values of the components. The circuits shown to the right in figure 5, indicates how the the Sallen and Key are made in TINA to get the simulation results that can be compared.

The figures below and on the following pages, shows how the output response is with an ideal- and the 741-OP-amp used for the practical tests. These can then be compared with the calculations illustrated above for Sallen and Key filter architecture.

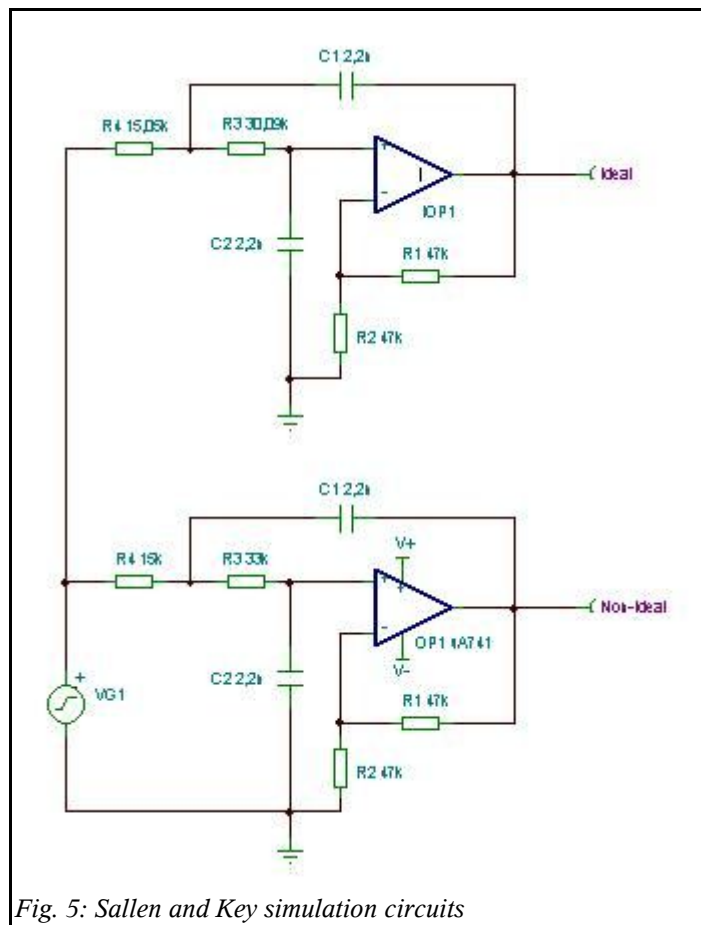


Fig. 5: Sallen and Key simulation circuits

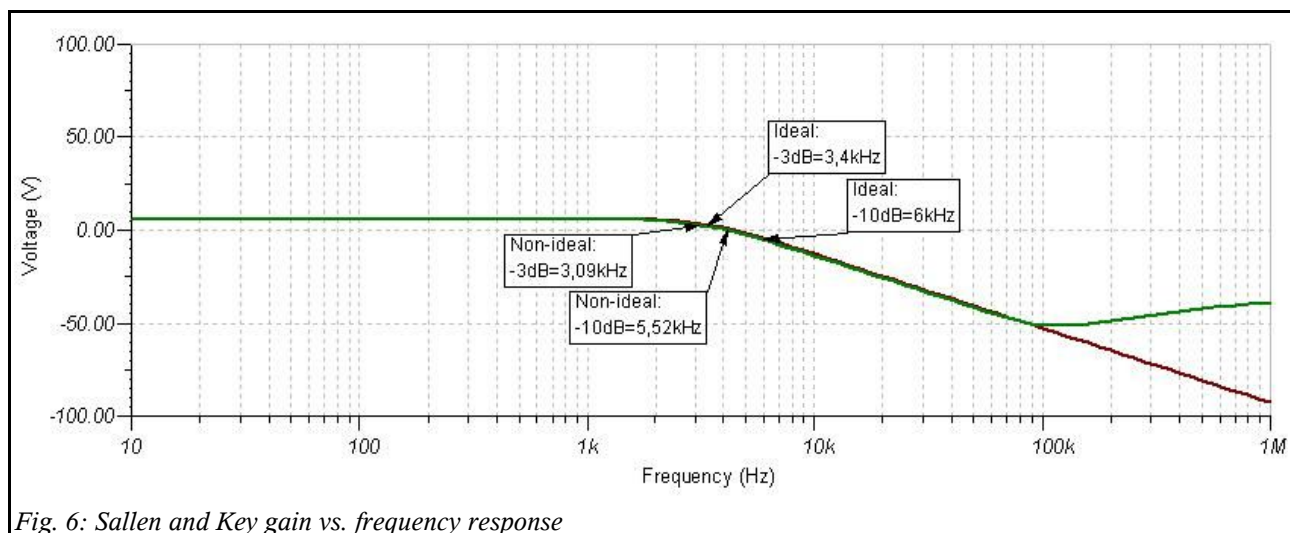
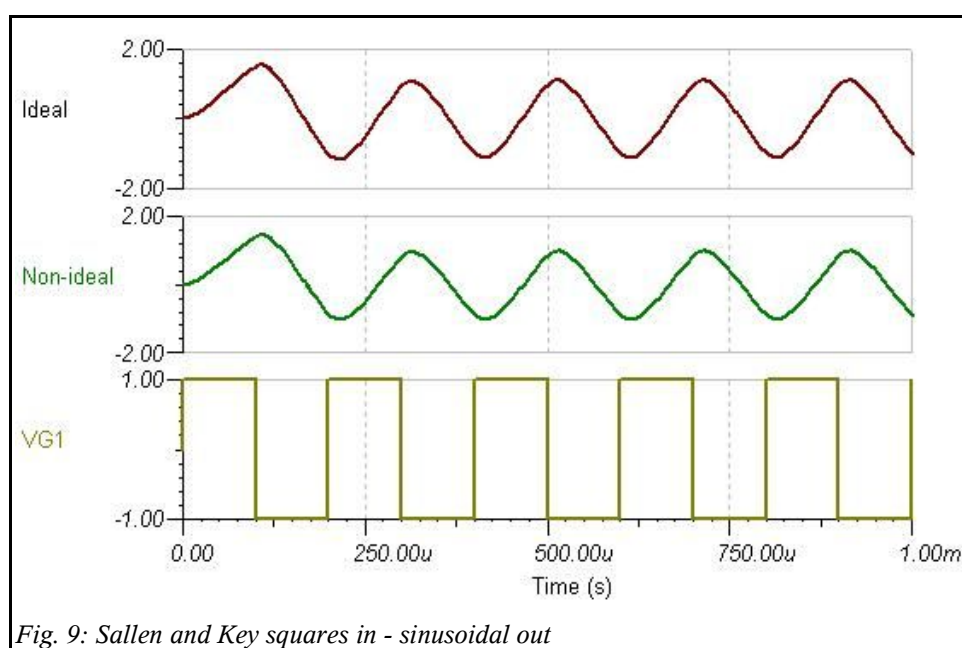
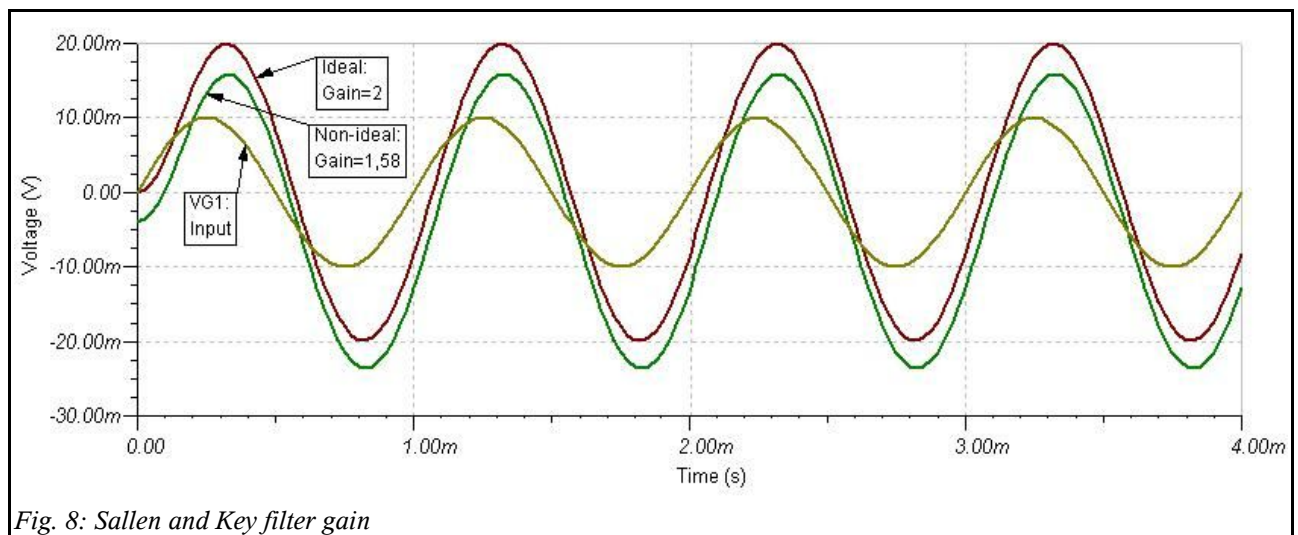
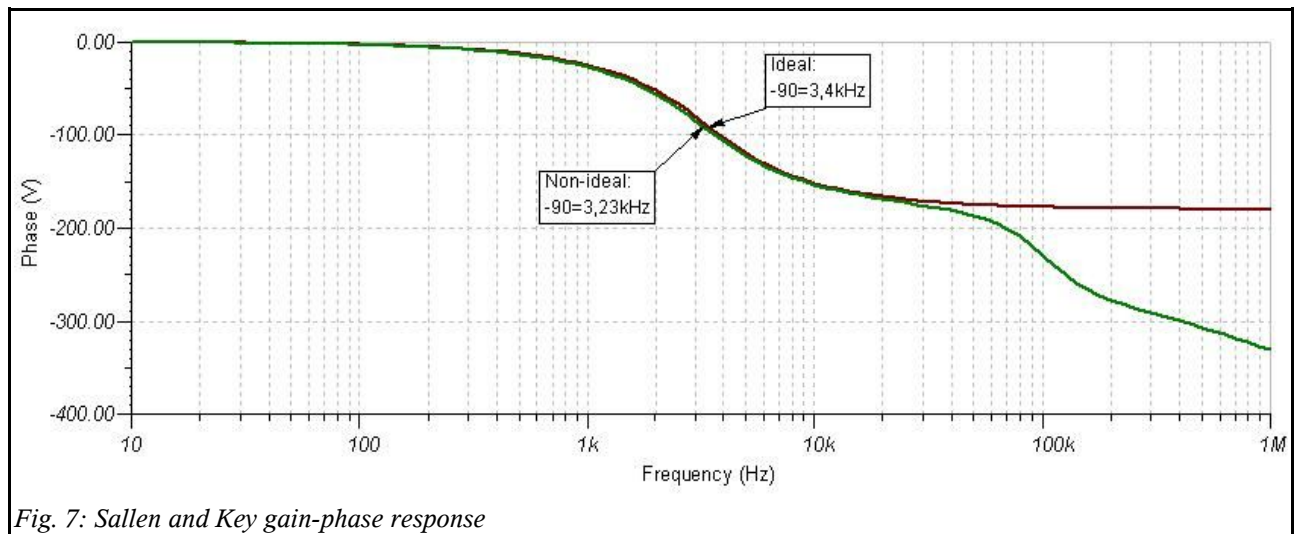


Fig. 6: Sallen and Key gain vs. frequency response



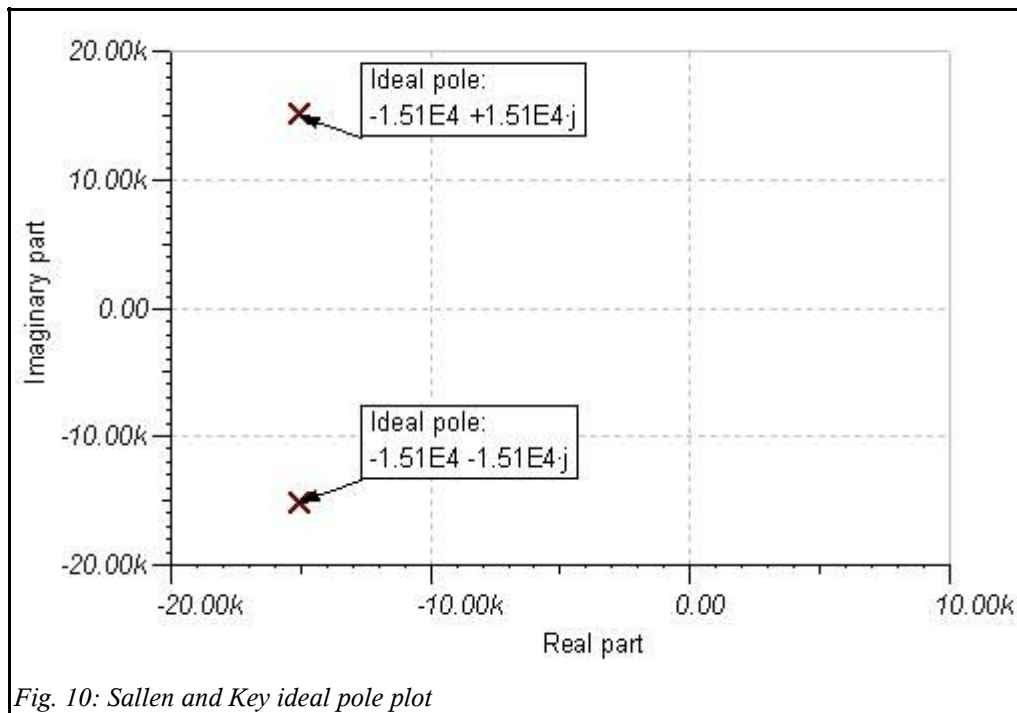


Fig. 10: Sallen and Key ideal pole plot

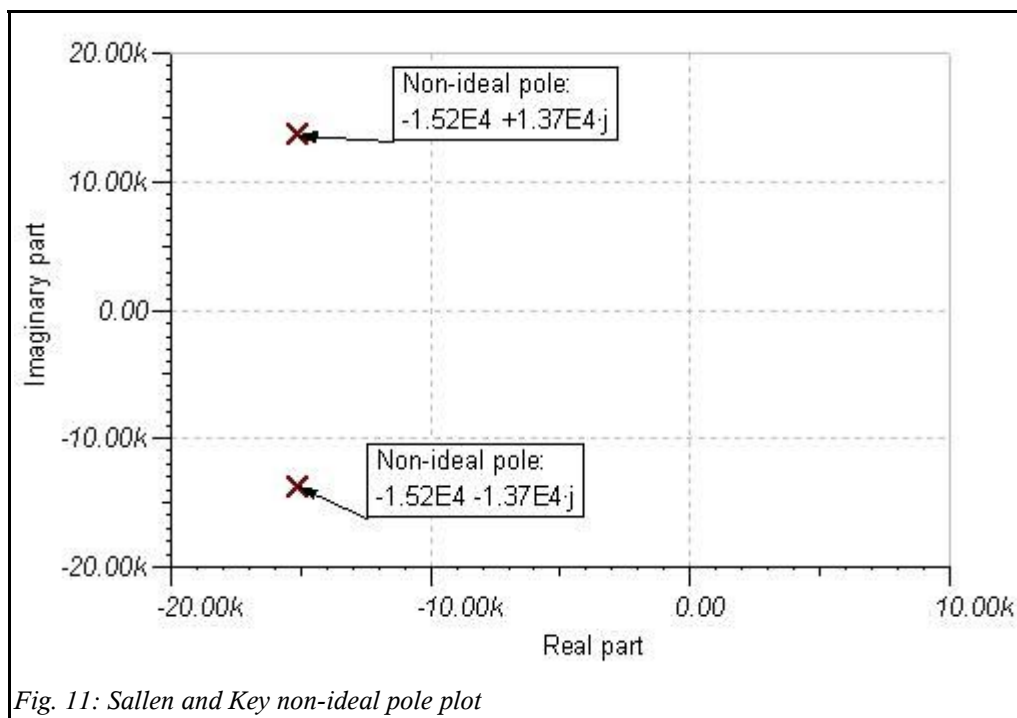


Fig. 11: Sallen and Key non-ideal pole plot

The simulations clearly states that there is a good relationship to the calculations. There is also a big difference between the ideal OP-amp and the 741's characteristics. To get better calculation results that will make the filter response better, these characteristics needs to implemented when the passive components is calculated. Figure 6 indicates how big the difference between the ideal and the real world really is, where the ideal OP-amp will keep on damping, and the 741 reach its physical limit near 100kHz.

## 6. Multiple Feedback (MFB)

The MFB filter are a two-pole topology, and in this case a low-pass filter response are used to satisfy the specifications for the assignment. The gain of this topology can easily be accommodated. The calculations to find the component values can be quite formidable. Meaning that it is not easy to modify the circuits values, because frequency, gain and type of response all interact. Although the MFB architecture has one more component than the Sallen and Key, this filters response has better a stopband rejection. One of the other advantages using the MFB topology is that it is less sensitive to component variations and has a superior high-frequency response compared to the Sallen and Key topology.

### 6.1. Low-pass architecture

The ideal low-pass architecture for a standard MFB filter is shown in figure 12. Here the values for the resistors and capacitors will also be calculated below. The formulas to calculate the values of the passive components are referred in the application notes from Texas instruments.

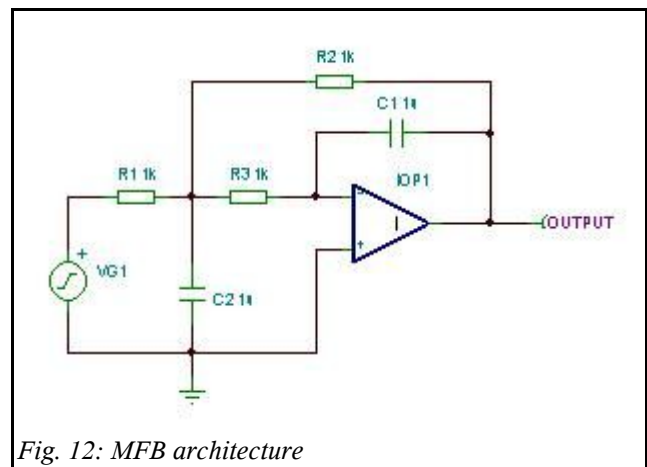


Fig. 12: MFB architecture

**Low-pass MFB ideal transfer function:**

$$\omega(s) = \frac{R_2}{-R_1 + (R_1 \cdot R_3 - R_2 \cdot R_3 - R_2 \cdot R_1) \cdot C_1 \cdot s - C_2 \cdot C_1 \cdot R_2 \cdot R_1 \cdot R_3 \cdot s^2}$$

### 6.2. Calculations

**Known:**

- $K = 2$
- $a_1 = \sqrt{2}$
- $b_1 = 1$
- $\omega = 2 \cdot \pi \cdot 3,4 \text{ k}$

**Chosen:**

- $R_1 = 15 \text{ k}\Omega$
- $C_1 = 1 \text{ nF}$

The  $a_1$  and  $b_1$  are the filter coefficients, that determines the gain behaviour in the passband.

$a_1 = \sqrt{2}$  -Cause the filter that is designed is a 2<sup>nd</sup> order filter. This can be derived from the Butterworth polynomials where  $a_0 = 0$  and  $a_1 = \sqrt{2}$ .

$b_1 = 1$  -This value can be found in the Butterworth filter polynomial table which is illustrated below in figure 13. The table is from the Texas Instruments application note SLOA049B.

Table 1. Butterworth Filter Table

FILTER ORDER	Stage 1		Stage 2		Stage 3		Stage 4		Stage 5	
	FSF	Q	FSF	Q	FSF	Q	FSF	Q	FSF	Q
2	1.000	0.7071								
3	1.000	1.0000	1.000							
4	1.000	0.5412	1.000	1.3065						
5	1.000	0.6180	1.000	1.6181	1.000					
6	1.000	0.5177	1.000	0.7071	1.000	1.9320				
7	1.000	0.5549	1.000	0.8019	1.000	2.2472	1.000			
8	1.000	0.5098	1.000	0.6013	1.000	0.8999	1.000	2.5628		
9	1.000	0.5321	1.000	0.6527	1.000	1.0000	1.000	2.8802	1.000	
10	1.000	0.5062	1.000	0.5612	1.000	0.7071	1.000	1.1013	1.000	3.1969

Fig. 13: Butterworth filter table

**$R_2$  is calculated with the gain formula:**

$$K = \frac{-R_2}{R_1} \Rightarrow R_2 = K \cdot R_1 = 2 \cdot 15k = 30k \sim R_2 = \underline{\underline{33k\Omega}} \quad (\text{E12 value})$$

**$R_3$  is calculated with the  $a_1$  filter coefficient formula:**

$$a_1 = \omega \cdot C_1 \cdot (R_2 + R_3 + \left(\frac{R_2 \cdot R_3}{R_1}\right)) \Rightarrow R_3 = -\left(a_1 + \omega \cdot C_1 \cdot R_2\right) \cdot \frac{R_1}{\omega \cdot C_1 \cdot (R_1 + R_2)} =$$

$$-(-\sqrt{2} + 2 \cdot \pi \cdot 3,4k \cdot 1n \cdot 30k) \cdot \frac{15k}{2 \cdot \pi \cdot 3,4k \cdot 1n \cdot 15k + 30k} = 12,07k \sim$$

$$\underline{\underline{R_3 = 12k\Omega}} \quad (\text{E12 value})$$

**$C_2$  is calculated with the  $b_1$  filter coefficient formula:**

$$b_1 = \omega^2 \cdot C_1 \cdot C_2 \cdot R_2 \cdot R_3 \Rightarrow C_2 = \frac{b_1}{\omega^2 \cdot C_1 \cdot R_2 \cdot R_3} = \frac{1}{(2 \cdot \pi \cdot 3,4k)^2 \cdot 1n \cdot 30k \cdot 12k} \Rightarrow \underline{\underline{C_2 = 6nF}}$$

-Two capacitors will be inserted in parallel here, one with the value 3,3nF and the other at 2,7nF.

**Control calculation for the cut-off frequency with the E12 values:**

$$\omega_c = \frac{1}{\sqrt{R_2 \cdot R_3 \cdot C_1 \cdot C_2}} = \frac{1}{\sqrt{33k \cdot 12k \cdot 1n \cdot 6n}} = 20,52k \frac{\text{rad}}{\text{sek}}$$

$$f_{\text{cut-off}} = \frac{\omega_c}{2 \cdot \pi} = \frac{20,52k}{2 \cdot \pi} = \underline{\underline{3,26kHz}}$$



**Calculations of the poles:**

The calculations for the poles of the MFB filter are made with the same procedure as in the Sallen and Key filter. The result for the MFB pole calculation are shown below.

**Poles for the ideal MFB circuit:**

$$-R_1 + (R_1 \cdot R_3 - R_2 \cdot R_3 - R_2 \cdot R_1) \cdot C_1 \cdot s - C_2 \cdot C_1 \cdot R_2 \cdot R_1 \cdot R_3 \cdot s^2 = 0$$

$$\underline{\underline{s = -15,24 \text{ k} \pm j15,1 \text{ k} \frac{\text{rad}}{\text{sek}}}}$$

**Poles for the non-ideal MFB circuit:**

$$-R_1 + (R_1 \cdot R_3 - R_2 \cdot R_3 - R_2 \cdot R_1) \cdot C_1 \cdot s - C_2 \cdot C_1 \cdot R_2 \cdot R_1 \cdot R_3 \cdot s^2 = 0$$

$$\underline{\underline{s = -15,03 \text{ k} \pm j13,97 \text{ k} \frac{\text{rad}}{\text{sek}}}}$$

**6.3. Simulations**

Simulations of the MFB architecture are done with the same procedure as the Sallen and Key architecture. The circuits on the right in figure 14, shows how the MFB are made in TINA for the simulations.

The figures on the following pages indicates the simulation output responses of the MFB Butterworth filter.

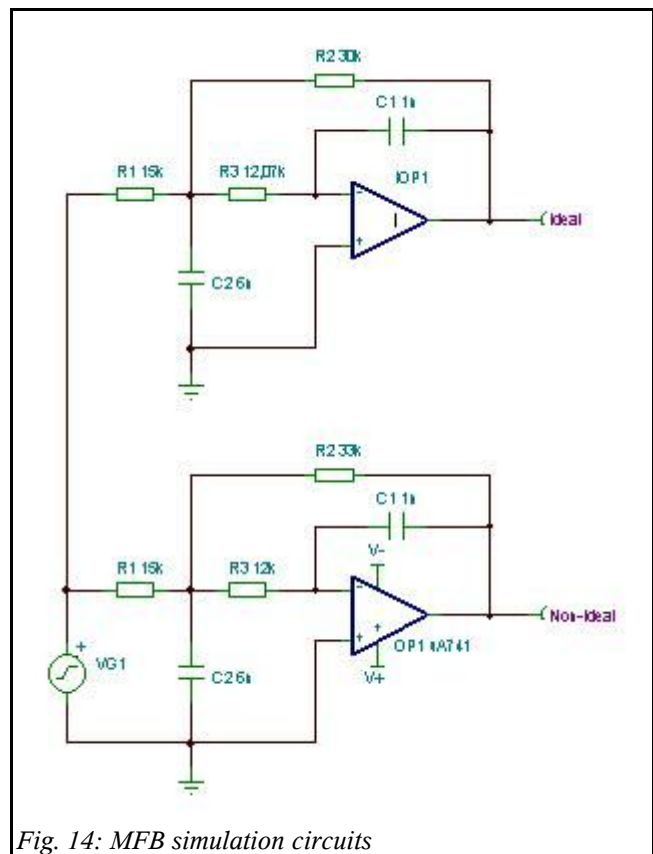


Fig. 14: MFB simulation circuits

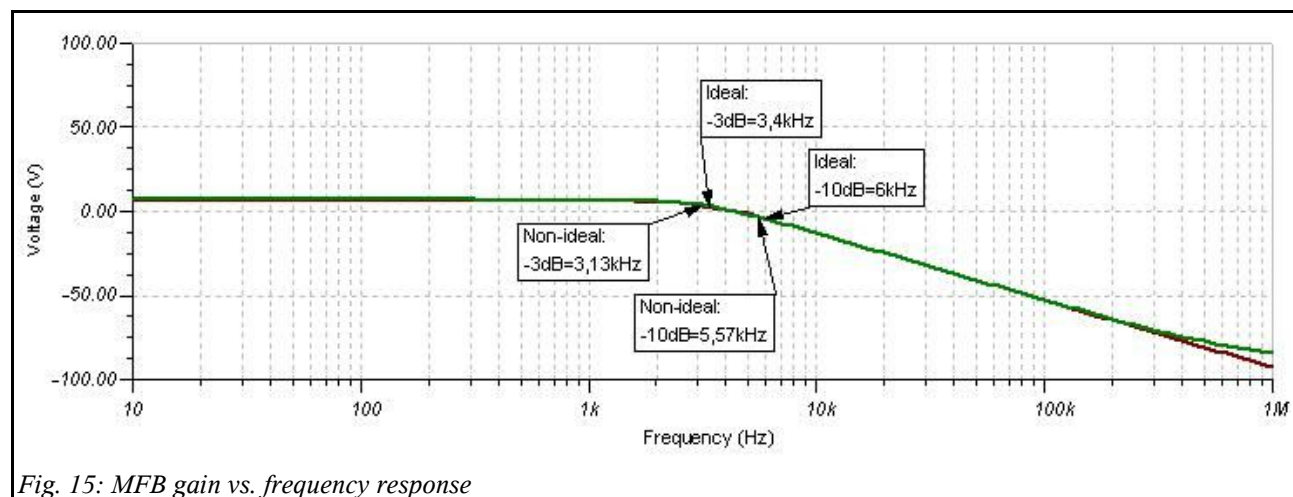


Fig. 15: MFB gain vs. frequency response

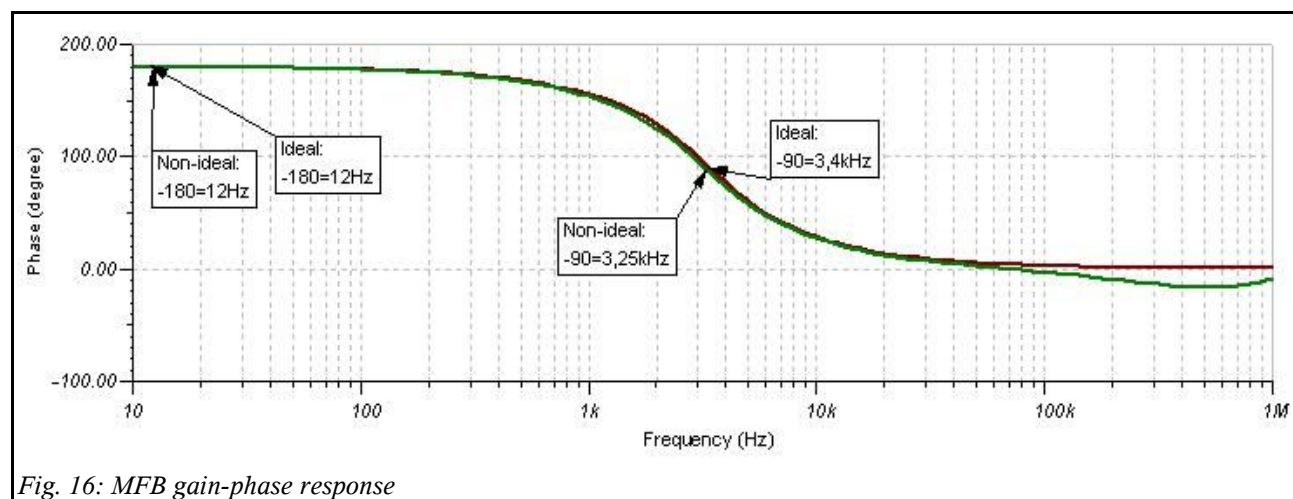


Fig. 16: MFB gain-phase response

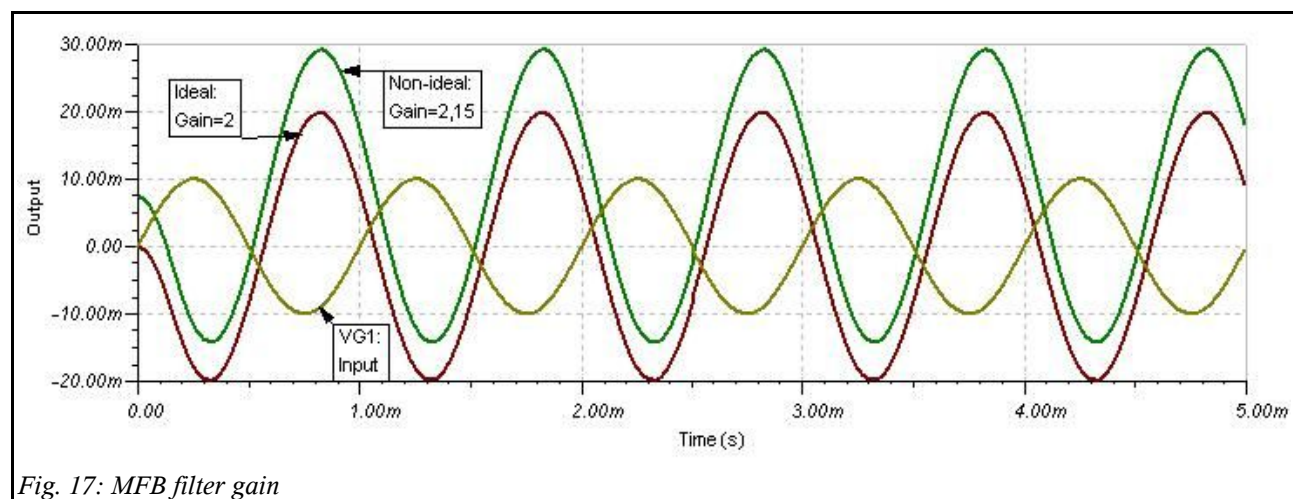


Fig. 17: MFB filter gain

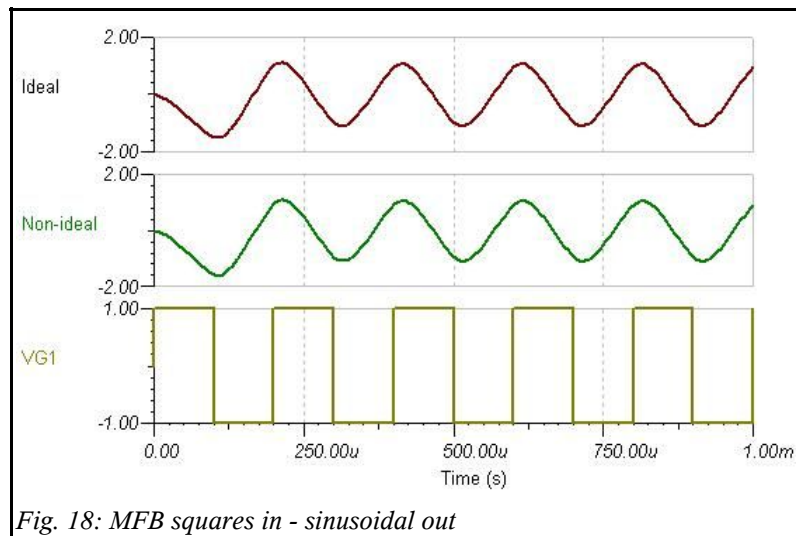


Fig. 18: MFB squares in - sinusoidal out

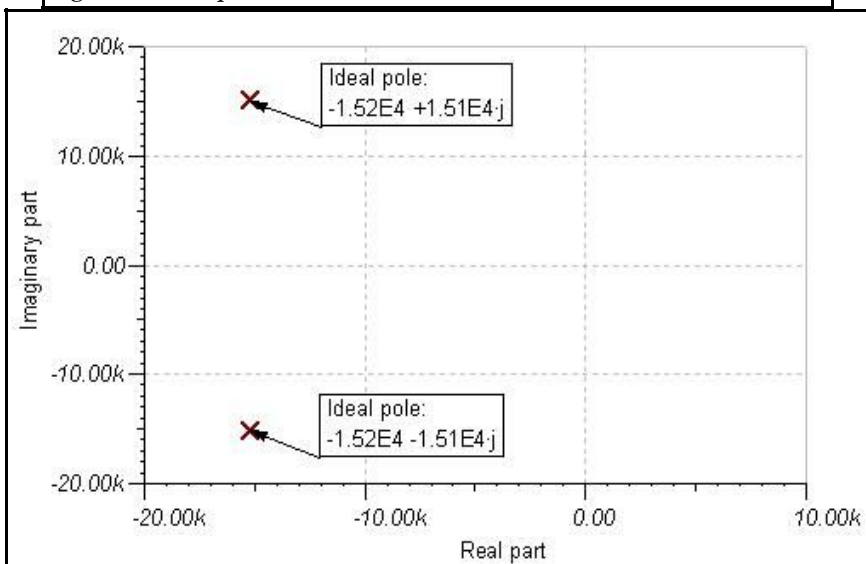


Fig. 19: MFB ideal pole plot

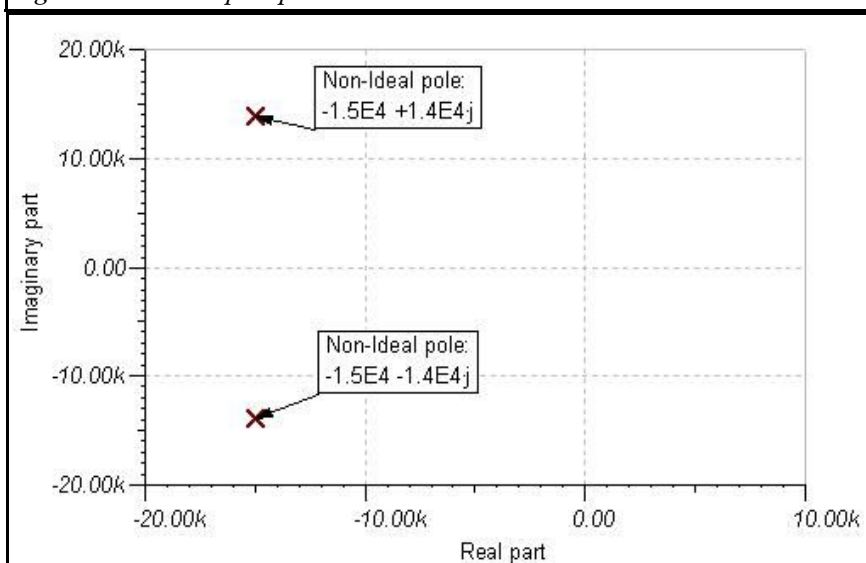


Fig. 20: MFB non-ideal pole plot

## 7. Test and measurements

### *Instruments used:*

#### **TDS220 – Tektronix digital oscilloscope:**

Used for: Measuring the phase-angle between input and output.

#### **TDS220 – Tektronix digital oscilloscope:**

Used for: Measuring the gain and frequency.

#### **Sinus generator:**

Used for: Frequency generator on the input.

#### **2 Power supply's:**

Used for:  $\pm 15\text{V}$  supply for the filter.

### 7.1. Test setup

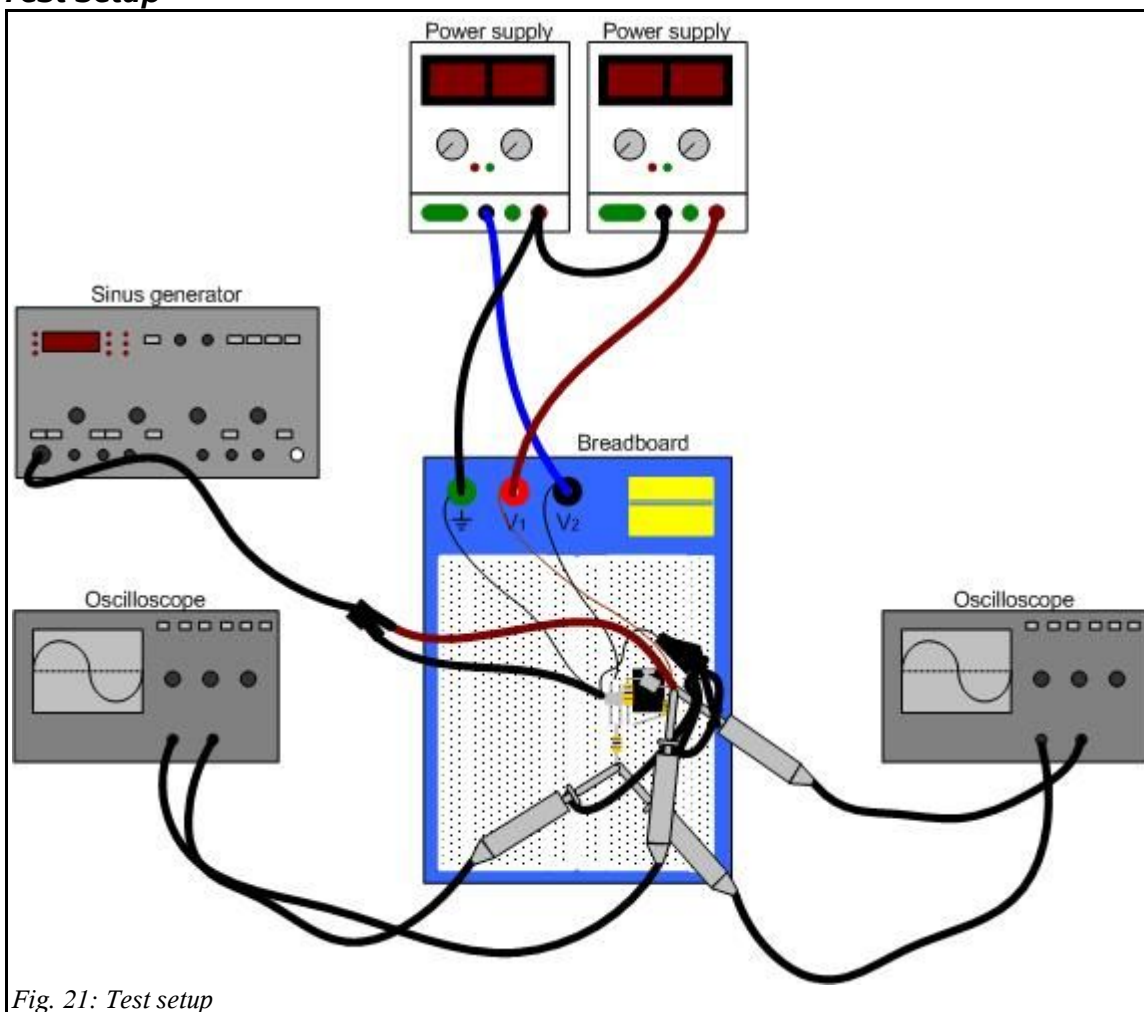


Fig. 21: Test setup

## 7.2. Gain-phase

The practical part of the assignment where the Sallen and Key, and the MFB circuits are build on a breadboard, and their performance is investigated are done as the test setup in figure 21 shows. Each of the circuits are build and tested separately, first the Sallen and Key and then the MFB circuit. The Tektronix oscilloscopes are the measuring instruments for all the tests (gain, phase and frequency), made in the assignment. The oscilloscope outputs in appendix 3, where obtained with one of the Tektronix oscilloscope plugged on to a computer. The figures indicates how the Sallen and Key and the MFB filters response where at different input frequencies.

### Method for measuring the phase:

The method used for measuring the phase relationship can be quit tricky, but if we have a look at how it is done it can be done with a little effort. With the test setup as shown, where one of the oscilloscopes are measuring the input signal from the sinus generator at channel 1, and the output signal from the filter at channel 2, the phase differences between input-output can be measured and calculated. To illustrate this better, the input-output result of the Sallen and Key, at the cut-off frequency is shown in figure 22 to the right. The input signals' phase is spread with one half of its sinus wave, and adjusted to the most left position of the x-axis. It is now possible to conclude that the phase for the output signal is lagging. Then it is possible to calculate the phase angle differences, by looking at how many lines at the x-axis there are between them (this is split up into 10 great/50 small lines on the screen). The number of lines used by the input are by looking at the figure 22 then equal to 6,4 steps. And the difference between the start of the input signal to the start of the output signal equal to 3,2 steps. The formula and calculation for the phase angle difference will be as shown below:

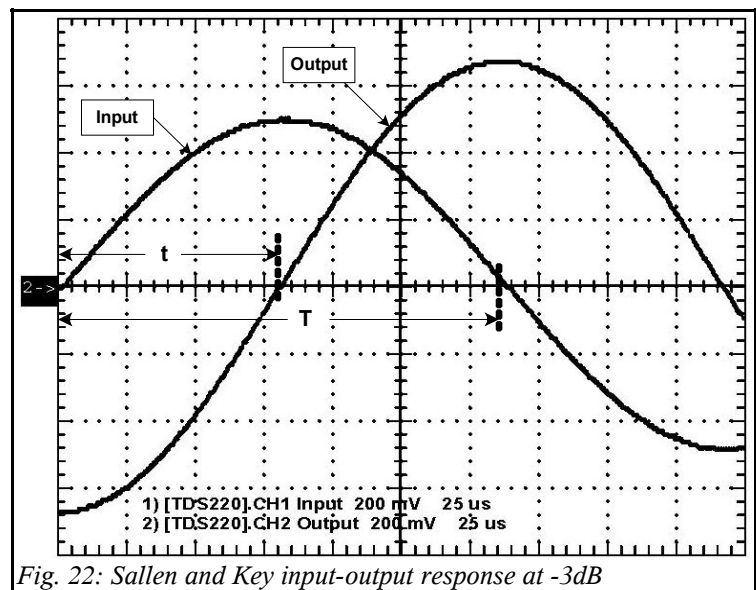
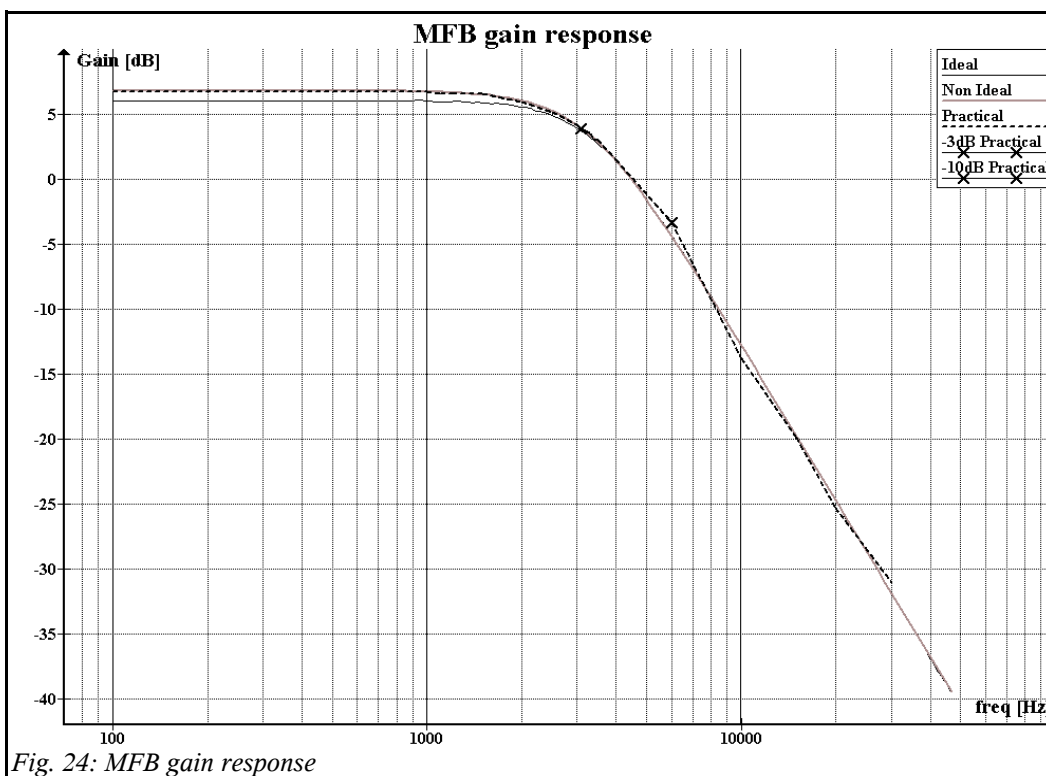
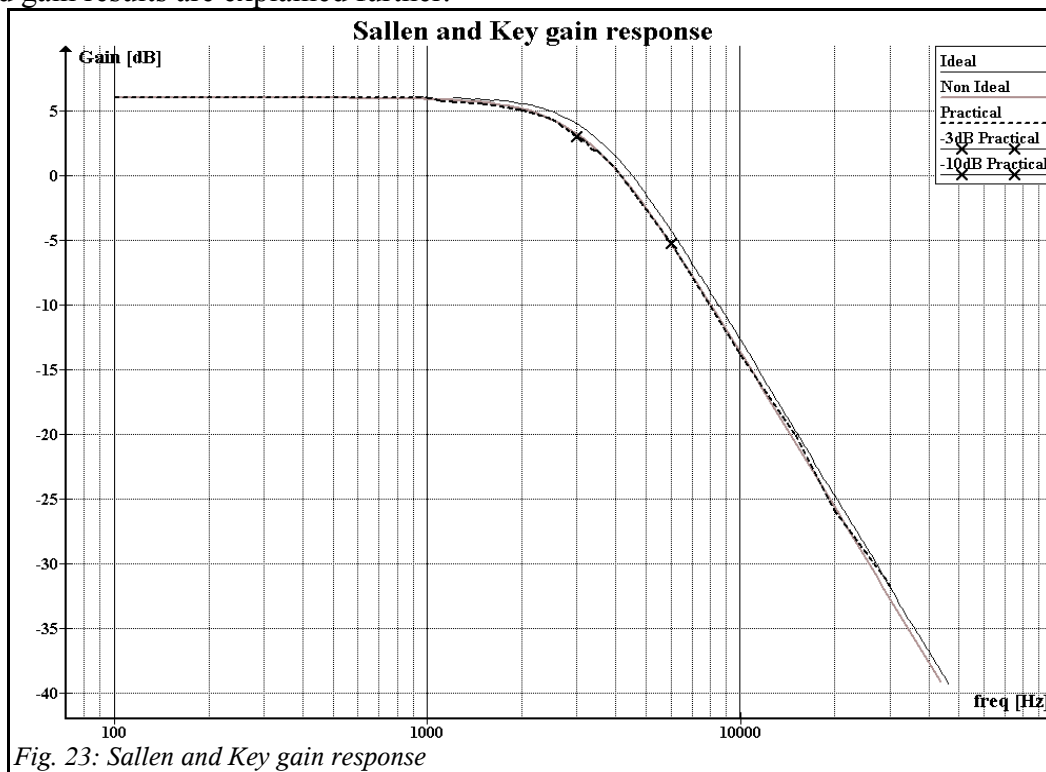


Fig. 22: Sallen and Key input-output response at -3dB

$$\theta = \frac{\Delta t}{T} \cdot 180 = \frac{3,2}{6,4} \cdot 180 = \underline{\underline{90 \text{ deg}}}$$

### 7.3. Gain results

With reference to the tables in the appendix 2, that shows the practical measurement results for the filters, their gain are plotted into graphs which the figure 23 and figure 24 indicates. In the conclusion the relationship for the Sallen and Key, MFB simulated ideal, non-ideal and the measured gain results are explained further.



### 7.4. Phase results

The phase has been measured parallel with the gain response of the filter as described previously. The result of the measurements refers to the tables in the appendix 2, and are plotted into a graph as shown below in figures 25 and 26.

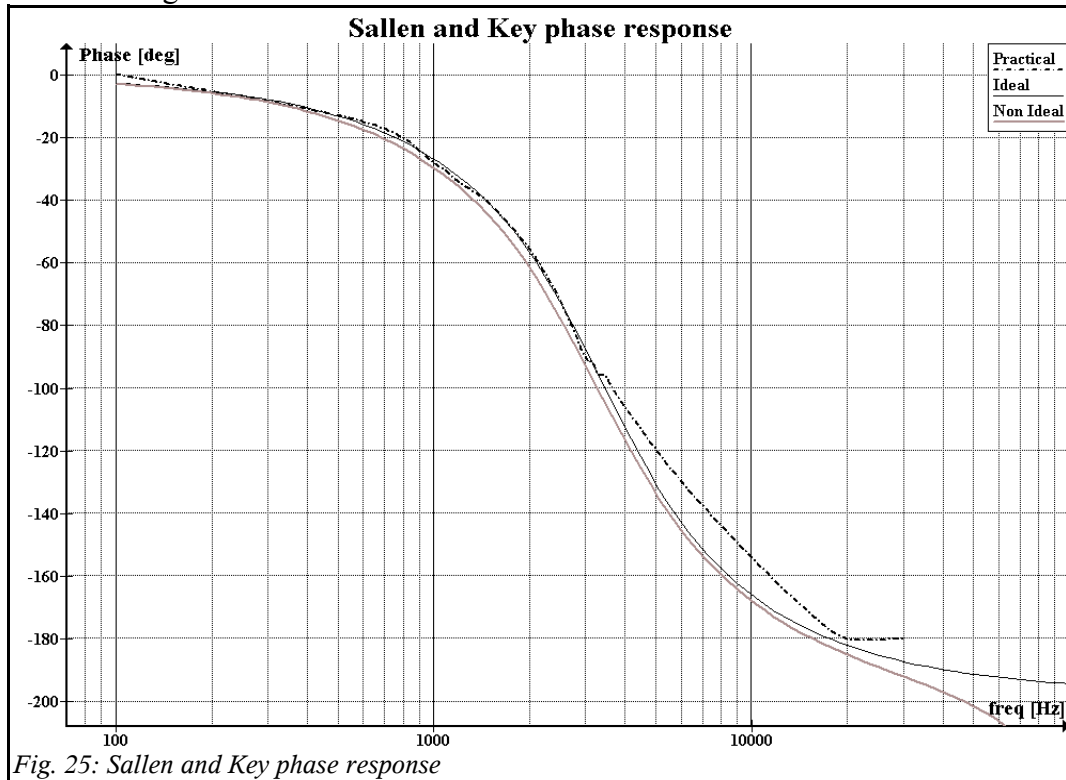


Fig. 25: Sallen and Key phase response

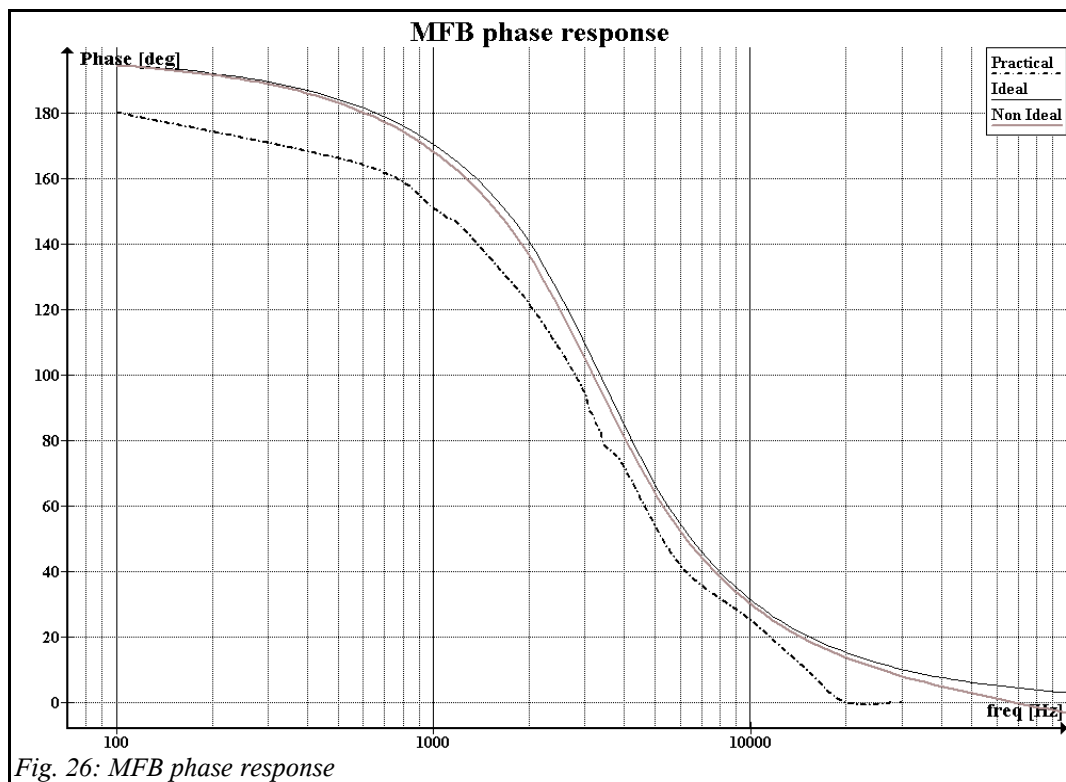


Fig. 26: MFB phase response



## 7.5. Pole point results

### Sallen and Key pole points:

Sallen and Key			
Constant gain	$f_c$	Phase	Gain
6,02dB	3kHz	-90°	2,98dB

$$\omega_K = f_c \cdot 2 \cdot \pi = 3 \text{ k} \cdot 2 \cdot \pi = 18,849 \text{ k} \frac{\text{rad}}{\text{sec}}$$

$$Q = 10^{\frac{\text{Gain}}{20}} = 10^{\frac{6,02-2,98}{20}} = 704,7 \text{ m}$$

$$s = -\frac{\omega_K}{2 \cdot Q} \pm j(\omega_K \cdot \sqrt{1 - \frac{1}{4 \cdot Q^2}}) = -\frac{18,849 \text{ k}}{2 \cdot 704,7 \text{ m}} \pm j(18,849 \text{ k} \cdot \sqrt{1 - \frac{1}{4 \cdot 704,7 \text{ m}}}) \Rightarrow$$

$$\underline{\underline{s = -13,37 \pm j15,14 \text{ k} \frac{\text{rad}}{\text{sec}}}}$$

### MFB pole points:

MFB			
Constant gain	$f_c$	Phase	Gain
6,77dB	3,1kHz	-90°	3,86dB

$$\omega_K = f_c \cdot 2 \cdot \pi = 3,1 \text{ k} \cdot 2 \cdot \pi = 19,478 \text{ k} \frac{\text{rad}}{\text{sec}}$$

$$Q = 10^{\frac{\text{Gain}}{20}} = 10^{\frac{6,77-3,86}{20}} = 715,3 \text{ m}$$

$$s = -\frac{\omega_K}{2 \cdot Q} \pm j(\omega_K \cdot \sqrt{1 - \frac{1}{4 \cdot Q^2}}) = -\frac{19,478 \text{ k}}{2 \cdot 715,3 \text{ m}} \pm j(19,478 \text{ k} \cdot \sqrt{1 - \frac{1}{4 \cdot 715,3 \text{ m}}}) \Rightarrow$$

$$\underline{\underline{s = -13,62 \pm j15,71 \text{ k} \frac{\text{rad}}{\text{sec}}}}$$



### Comparison

The calculated pole points from respectively the simulated ideal and non-ideal circuits, and the practical tests are compared in figure 27. The results indicates that practical measurements for the Sallen and Key and the MFB circuits are close to each other, as are the ideal and non-ideal calculated and simulated results. The conclusion explains further what the cause of this can be and how better results can be achieved. For the practical test and setup of the circuit, the same operational amplifier (741), and some of the resistors has been used in both applications.

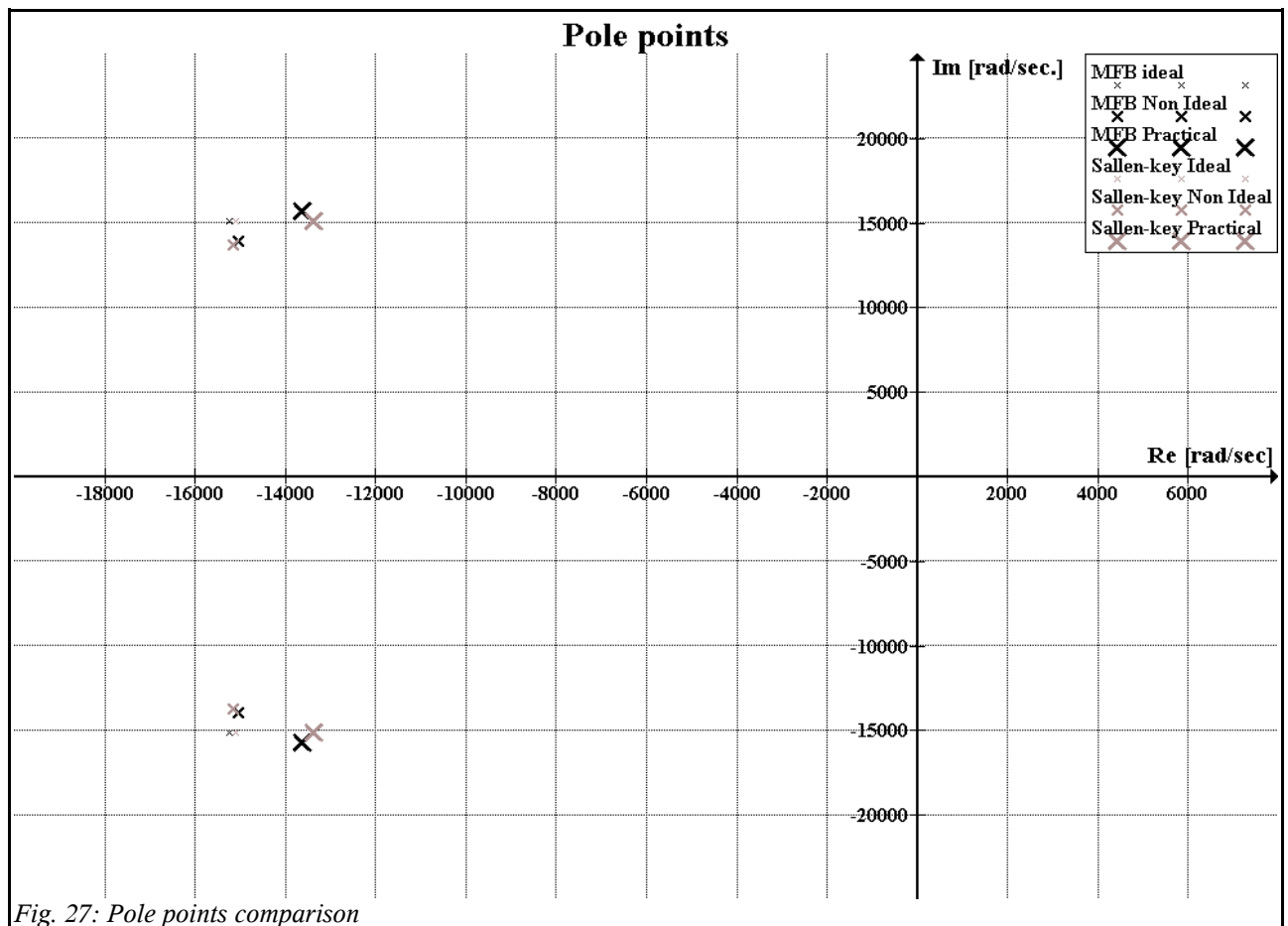


Fig. 27: Pole points comparison

## 8. Conclusion

The active filter design project has been completed successfully with two different approaches to the specified assignment. The intention by doing this, was to find out how the two different architectures would react individually, and when we compared the results with each other, would get to a final result where the two architectures would give the same output responses.

To make an active filter is not a new subject, but it has been with a different approach in this assignment. The previously project with the design of an active filter has been with Laplace calculations, where the method for completing this assignment where with more simplified formulas. That has made the interest for this assignment grow, and further more build active filters in the future, cause I have now experienced that there are more than one approach, and tools for designing these type of filters. The research we as a group have been doing for this project has also been different. By reading a lot of application notes and books related to making Butterworth filters, there was a lot of stuff that came in quite handy for this assignment and also gave a lot second thoughts in designing active filters.

The capacitors for the circuits have been selected after the principle of what was “in stock”. Of course the capacitors have to be in a special material to get the best results. For these applications metallized polycarbonate capacitors is a good choice (cause usual capacitors as ceramic capacitors can cause errors to the filter circuit) where the tolerance is at 1%. The resistors have been selected after the same principle as the capacitors (what's in stock) , but with a slight difference in the tolerance which is at 5%. If better results should be achieved, which means that the calculations results should be closer to the practical results, a change in the resistor tolerance could be a method for achieving this.

**The Sallen and Key architecture** has proven once again that the use of it, can give a very successfully result. With the calculations of the non-ideal Sallen and Key architecture compared with the practical results it can be concluded that they are leaning up quite next to each other. The comparison for this and the ideal result, where the difference is greater, is the output of using E12 values for the components.

**For the MFB** circuit that has shown great comparison for the Sallen and Key architecture it can be concluded that when using the same component values as much as possible, the MFB still has less sensitivity to component variations (or tolerance) as the Sallen and Key circuit. As shown in the simulations for these two architectures the MFB is also superior at higher frequency performance. But there are still the 180 degrees problem from the input to the output of this architecture. One way to correct this (if needed), would be to insert another inverting OP-amp at the output of the MFB circuit, that would shift the phase 180 degrees.

Another neat function that the active filter has, is that it can convert square signals to sinusoidal signals between the input and the output. If a really good sinus is needed, this is possible to derive by making the filter order number higher. This method of converting a square signal to a sinusoidal are seen in a lot of new micro processors today like for instance the PSoC micro processors where the internal contents consists of “building blocks” where active filters is a possibility. The square to sinusoidal conversion are shown in the simulation results for the Sallen and Key and the MFB, and has not been tested practically in this assignment.

Overall this assignment has been interesting and has given some new tools on how to design active filters and a good understanding of the different types of architectures.

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Carsten Kristiansen

## 9. References

### 9.1. Internet

- TI application note SBFA001A, <http://focus.ti.com/lit/an/sbfa001a/sbfa001a.pdf>
- TI application note SLOA049B, <http://focus.ti.com/lit/an/sloa049b/sloa049b.pdf>
- TI application note SLOA088, <http://focus.ti.com/lit/ml/sloa088/sloa088.pdf>
- Maxim application note AN733, <http://pdfserv.maxim-ic.com/en/an/AN733.pdf>

### 9.2. Literature

- Passive and active filters – Theory and implementations, Wai-Kai Chen.
- Introduction to electrical engineering, Mulukutla S. Sarma.
- Handout notes for Active Filter Design.

### 9.3. Software tools

- TINA Pro for Windows, [www.designsoft.com](http://www.designsoft.com)
- Filter Wiz Pro, [www.schematica.com](http://www.schematica.com)
- Graph 3.2.2, [www.padowan.dk](http://www.padowan.dk)

## 10. Appendix

### 10.1. Appendix 1 – Calculations from Mathcad

**Sallen and Key:**

$$s = \frac{1}{2C_2C_1R_3R_1R_2} \left[ C_1R_4R_1 - C_2R_3R_2 - C_2R_3R_1 + \left[ (C_1)^2(R_4)^2(R_1)^2 - 2C_1R_4R_1C_2R_3R_2 - 2C_1R_4(R_1)^2C_2R_3 + (C_2)^2(R_3)^2(R_2)^2 + 2(C_2)^2(R_3)^2R_2R_1 + (C_2)^2(R_3)^2(R_1)^2 - 4C_2C_1(R_3)^2R_1R_2 \right]^{\frac{1}{2}} \right]$$

$$s = \frac{1}{2C_2C_1R_3R_1R_2} \left[ C_1R_4R_1 - C_2R_3R_2 - C_2R_3R_1 - \left[ (C_1)^2(R_4)^2(R_1)^2 - 2C_1R_4R_1C_2R_3R_2 - 2C_1R_4(R_1)^2C_2R_3 + (C_2)^2(R_3)^2(R_2)^2 + 2(C_2)^2(R_3)^2R_2R_1 + (C_2)^2(R_3)^2(R_1)^2 - 4C_2C_1(R_3)^2R_1R_2 \right]^{\frac{1}{2}} \right]$$

**MFB:**

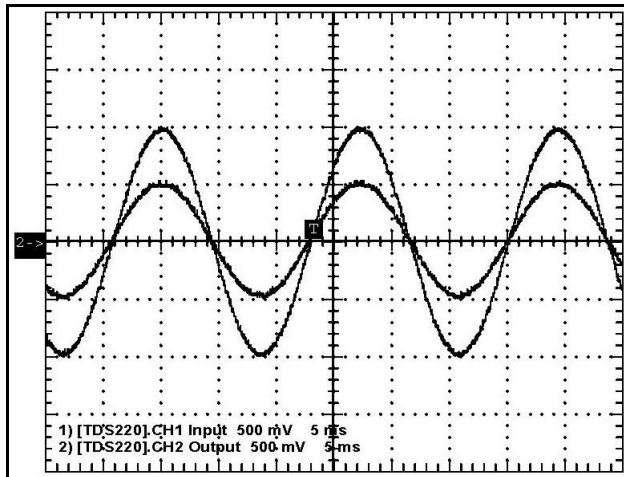
$$s = \frac{1}{2C_2C_1R_2R_3R_1} \left[ -C_1R_3R_1 - C_1R_2R_1 - C_1R_2R_3 + \left[ (C_1)^2(R_3)^2(R_1)^2 + 2(C_1)^2R_3(R_1)^2R_2 + 2(C_1)^2(R_3)^2R_1R_2 + (C_1)^2(R_2)^2(R_1)^2 + 2(C_1)^2(R_2)^2R_1R_3 + (C_1)^2(R_2)^2(R_3)^2 - 4C_2C_1R_2R_3(R_1)^2 \right]^{\frac{1}{2}} \right]$$

$$s = \frac{1}{2C_2C_1R_2R_3R_1} \left[ -C_1R_3R_1 - C_1R_2R_1 - C_1R_2R_3 - \left[ (C_1)^2(R_3)^2(R_1)^2 + 2(C_1)^2R_3(R_1)^2R_2 + 2(C_1)^2(R_3)^2R_1R_2 + (C_1)^2(R_2)^2(R_1)^2 + 2(C_1)^2(R_2)^2R_1R_3 + (C_1)^2(R_2)^2(R_3)^2 - 4C_2C_1R_2R_3(R_1)^2 \right]^{\frac{1}{2}} \right]$$

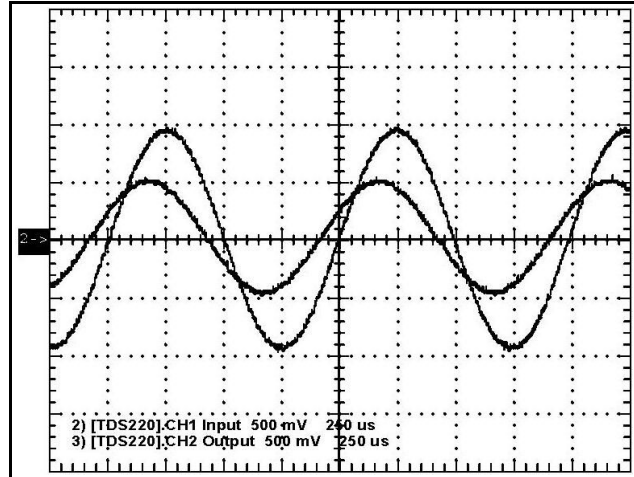
### 10.2. Appendix 2 – Measurement Tables

Sallen and Key					MFB			
Freq: [Hz]	Gain: [V/V]	Gain: [dB]	Phase: [deg.]		Freq: [Hz]	Gain: [V/V]	Gain: [dB]	Phase: [deg.]
50	2	6,02	0		50	2,18	6,77	180
600	2	6,02	-11		600	2,18	6,77	154
1000	2	6,02	-31		1000	2,18	6,77	151
1100	1,95	5,80	-31		1100	2,14	6,61	148
1200	1,93	5,71	-34,3		1200	2,14	6,61	146
1500	1,89	5,53	-41,1		1500	2,14	6,61	136
2000	1,8	5,11	-55,4		2000	1,98	5,93	122
2500	1,66	4,40	-72		2500	1,81	5,15	108
<b>3000</b>	<b>1,41</b>	<b>2,98</b>	<b>-90</b>		3000	1,63	4,24	95
3100	1,39	2,86	-91,3		<b>3100</b>	<b>1,56</b>	<b>3,86</b>	<b>90</b>
3200	1,35	2,61	-92,9		3200	1,53	3,69	87
3300	1,28	2,14	-96		3300	1,5	3,52	84
3400	1,26	2,01	-96		3400	1,43	3,11	82
3500	1,24	1,87	-96,43		3500	1,42	3,05	78
4000	1,068	0,57	-105,9		4000	1,19	1,51	72
6000	0,547	-5,24	-129,7		6000	0,679	-3,36	42
10000	0,202	-13,89	-154		10000	0,203	-13,85	25
15000	0,099	-20,09	-171		15000	0,101	-19,91	10
20000	0,051	-25,85	-180		20000	0,054	-25,35	0
30000	0,026	-31,70	-180		30000	0,028	-31,06	0

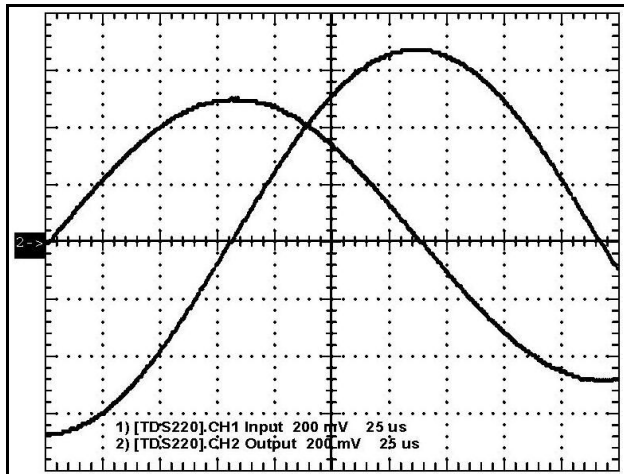
### 10.3. Appendix 3 – Oscilloscope outputs



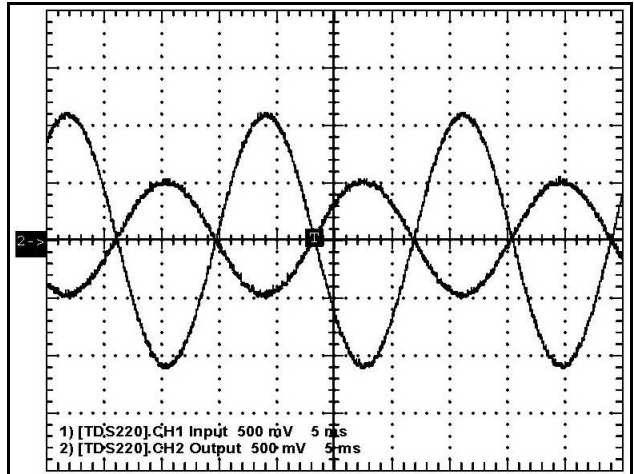
**Sallen and Key:** Freq. = 60Hz, gain = 6,02dB, phase angle = 0 degrees.



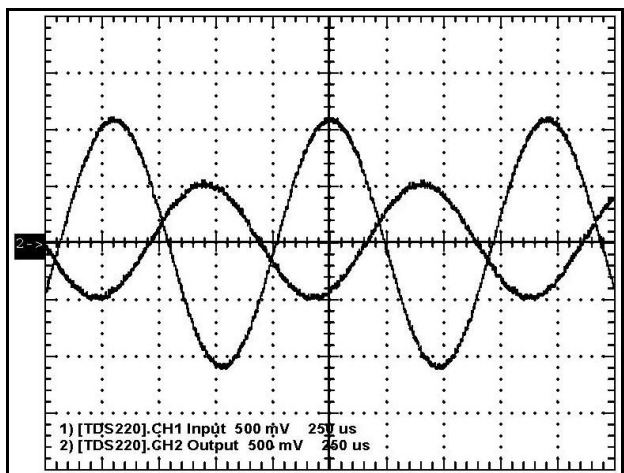
**Sallen and Key:** Freq. = 1kHz, gain = 6,02dB, phase angle = -31 degrees.



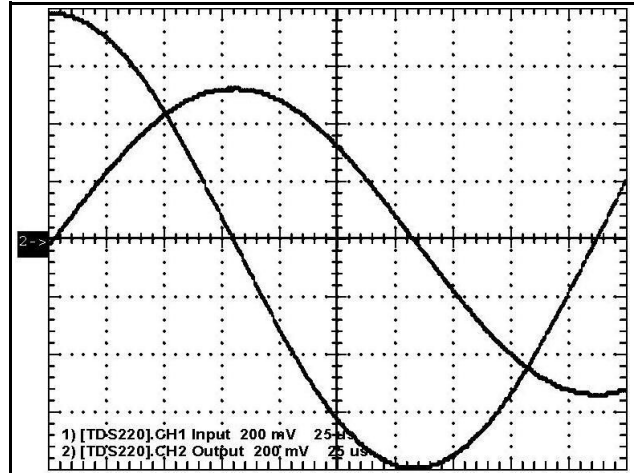
**Sallen and Key:** Freq. = 3,0kHz, gain = 2,98dB, phase angle = -90 degrees.



**MFB:** Freq. = 60Hz, gain = 6,77dB, phase angle = 180 degrees.



**MFB:** Freq. = 1kHz, gain = 6,77dB, phase angle = 151 degrees.



**MFB:** Freq. = 3,1kHz, gain = dB, phase angle = 90 degrees.