Active filter one version 1.0

DELTA AUDIO

Jens Rasmussen

1 Introduction

This document is a guide on how to use the Active filter one printed circuit board. The PCB has room for a complete 4 way active cross over network for one channel, and features the option of no less then 14 equalizers (EQ's) in all. The filter designed to this PCB can be either second or fourth order, depending on what you need.

For a stereo set two PCB's are needed, along with a power supply and casing for the crossover unit.

In this document you will find a 2 way design using low frequency EQ's. It's intended as a template for DIY builders and professional speaker designers.

Before designing a filter and EQ's with this PCB, data of the wanted filter must be at hand. This means that info about cut off frequencies; Q and gain are needed for the different filter sections on the PCB.

Happy building.

Contents

2 Overview

A PCB module can be configured to make the following filter.

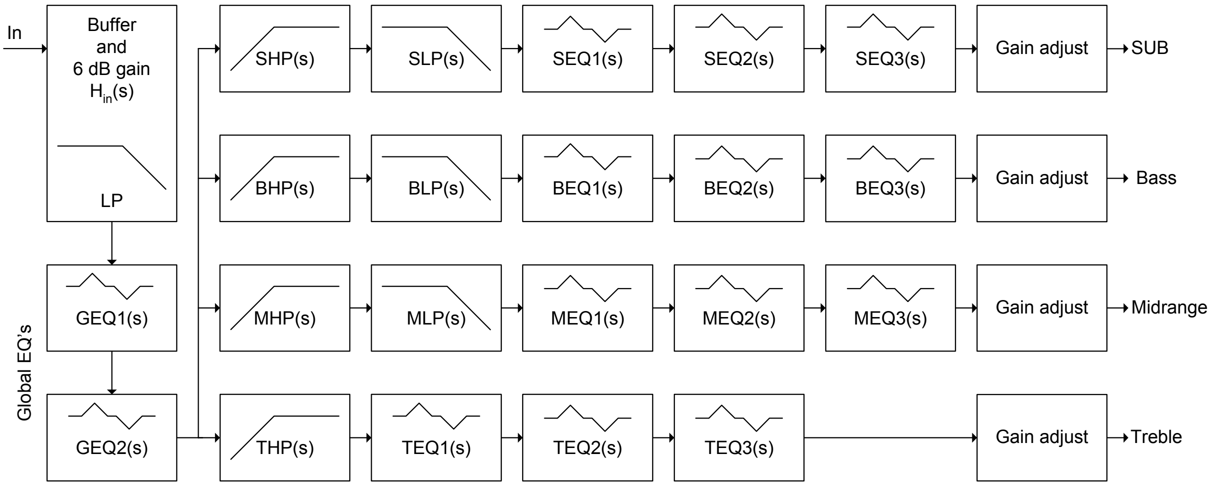


Figure 1: Block schmatic of the PCB

The names in the boxes refer to the calculations (transferfunction) of each filter block.
The PCB has the blocks laid out like this:

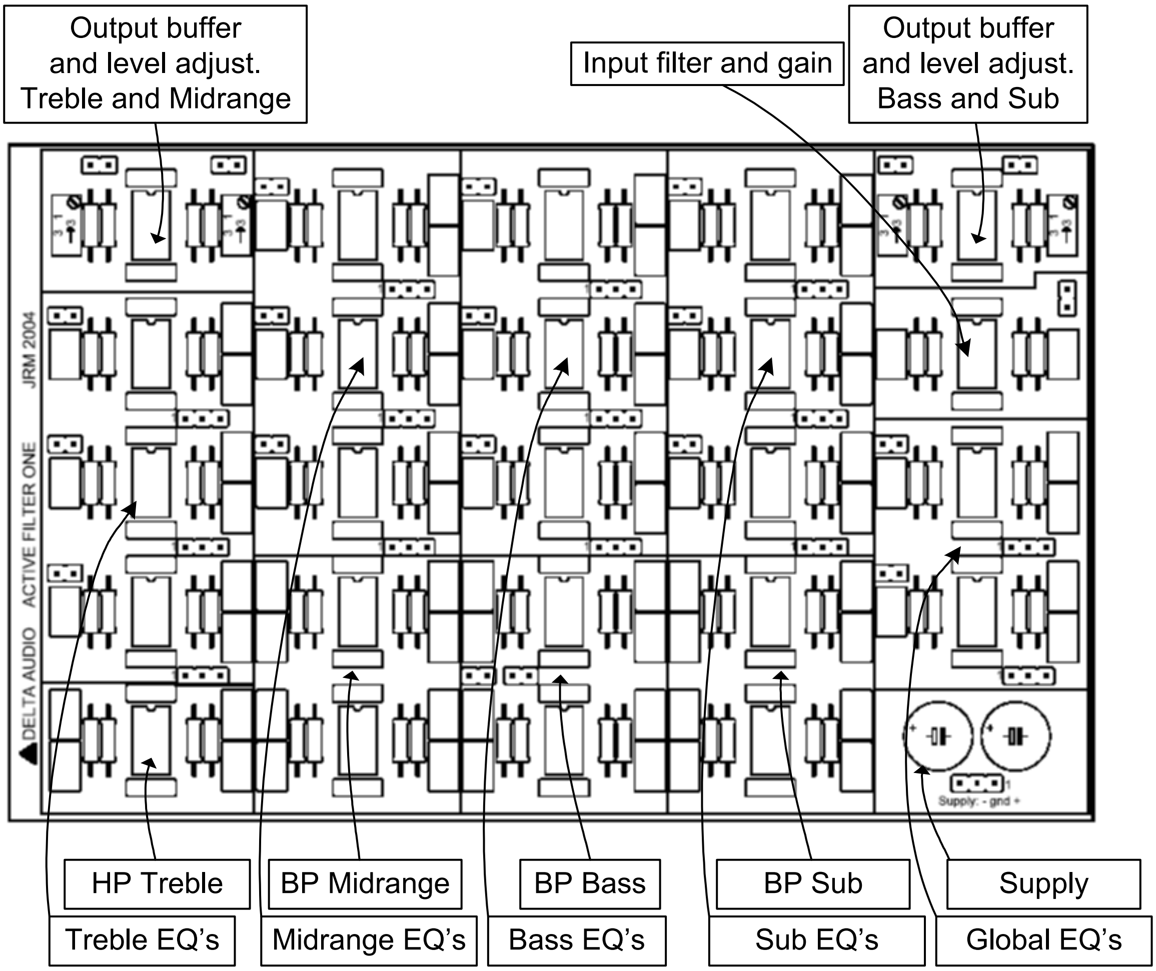


Figure 2: Layout on the PCB

3 The schematics

Each component has a three digit designator. The first two digits refer to the section of the schematic, and the last digit refers to the component number in that section. An example could be C041, which is C1 in section four.

If a section is used more than once like the HP, LP and EQ’s, C1 in one section will be in the same place as C1 in a similar section. This means that the transfer functions derived can be reused by changing the number in the component designator to refer to the section where the component is used.

When reading this chapter, please keep a copy of the schematics in hand for easy reference.

3.1 Input filter and buffer

The input filter has three major functions in the active filter.

1. Set the input impedance.
2. Cut of HF noise signals.
3. Boost the signal to improve the signal to noise ratio.

The input filter and buffer are made by the components around IC1A, and the opamp IC1. When building this active filter you do not need to design this stage of the filter unless your source’s signal amplitude is different from around 1 V RMS.

I have already designed a stage that suits signal levels of 1 V, so all you need is, to mount the components specified in the parts list, and you are ready to design the rest of the filter.

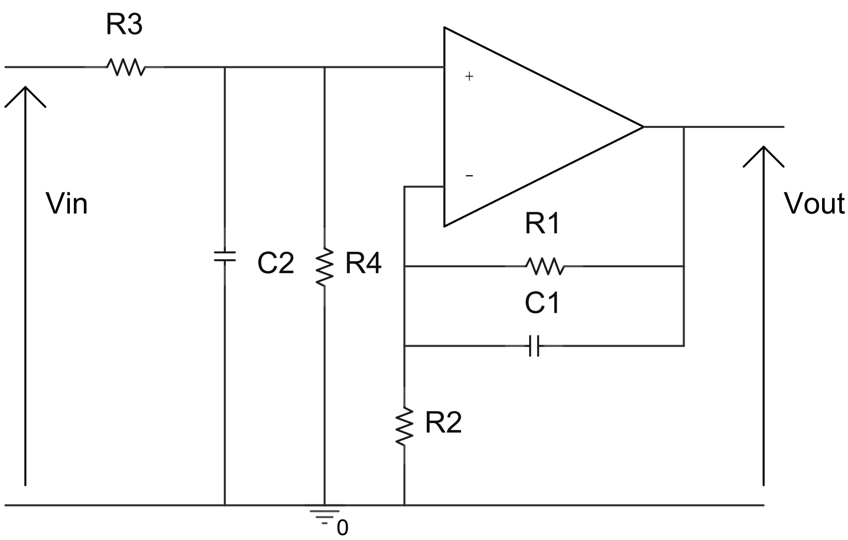


Figure 3: Input buffer and gain

3.2 The BP filter in each section

In all sections (apart from the treble) there is need for band pass (BP) filter. The filter is used to remove unwanted frequencies in a given part of the audio band. There are designated circuits for making BP filters, but since this PCB is intended for the ultimate analog filter versatility, I have employed a HP and LP filter to form the BP function instead.

A BP filter is described by two things:

- f_c cut off frequency HP and LP
- Q

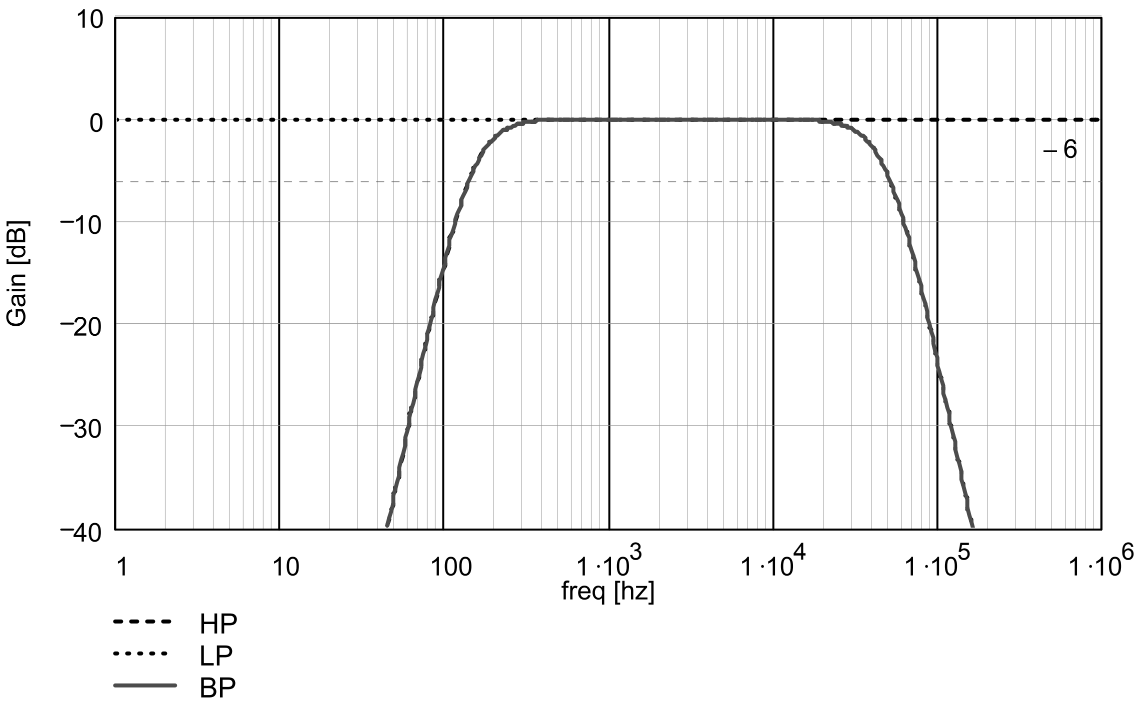


Figure 4: BP section made by a HP and LP in series

Figure 4 Shows the frequency response when a series connection using a HP and LP 4.th order filter is made.

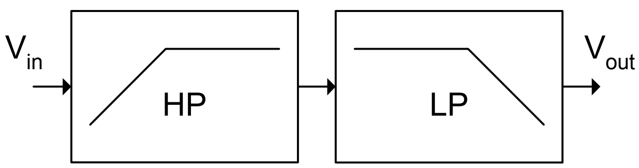


Figure 5: Series connection of HP and LP

3.2.1 The Sallen and Key HP filter

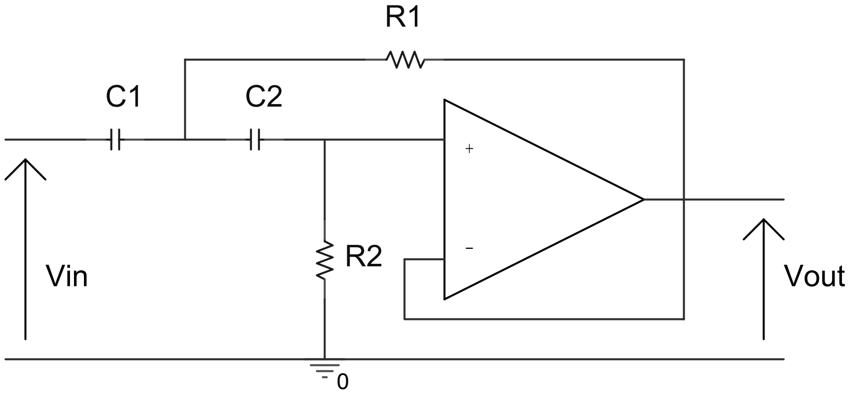


Figure 6: Sallen and Key HP section

To design a filter the describing parameters are the Q and cutoff frequency f_c.

$$Q = \frac{1}{\frac{C_1 + C_2}{R_2 \cdot C_2 \cdot C_1} \cdot \sqrt{C_1 \cdot R_1 \cdot C_2 \cdot R_2}}$$

$$f_c = \frac{1}{2\pi \cdot \sqrt{R_1 \cdot R_2 \cdot C_1 \cdot C_2}}$$

It's a good idea to choose the caps C_1 and C_2 because resistors have more different values to choose from. This will also enable the DIY builder to measure the caps in hand, and use the specific values in an attempt to maximize the accuracy of the filter.

This will result in the following design equations.

$$R_1 = \frac{1}{2 \cdot \pi \cdot f_c \cdot (C_1 + C_2) \cdot Q} \qquad R_2 = Q \cdot \frac{C_1 + C_2}{2 \cdot \pi \cdot f_c \cdot C_2 \cdot C_1}$$

If you decide to have $C_1 = C_2 = C$ the following design equations can be used:

$$R_1 = \frac{1}{4 \cdot \pi \cdot f_c \cdot C \cdot Q} \qquad R_2 = \frac{Q}{C \cdot \pi \cdot f_c}$$

3.2.2 The Sallen and Key LP filter

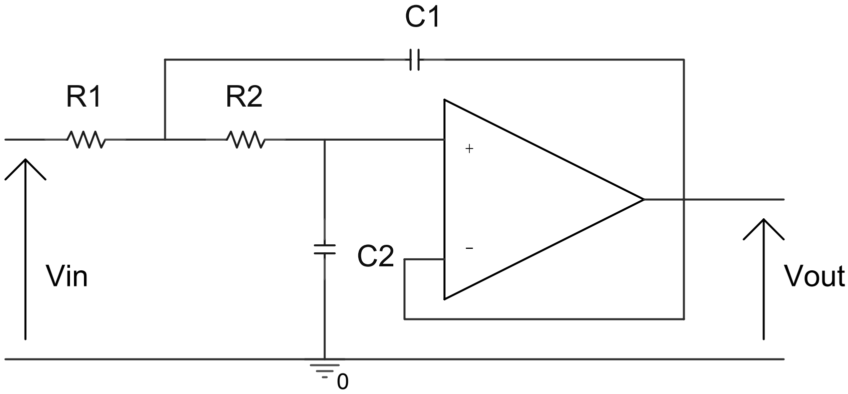


Figure 7: Sallen and Key LP section

To design a filter the describing parameters are the Q and cutoff frequency f_c.

$$Q = \frac{1}{\frac{R_1 + R_2}{R_1 \cdot R_2 \cdot C_1} \cdot \sqrt{R_1 \cdot R_2 \cdot C_1 \cdot C_2}}$$

$$f_c = \frac{1}{2\pi \cdot \sqrt{R_1 \cdot R_2 \cdot C_1 \cdot C_2}}$$

Unfortunately it's not possible to use two capacitor values as a starting point for a LP design. Therefore the resistor values can be chosen and the capacitors values worked out later.

This results in the following design equations.

$$C_2 = \frac{1}{2 \cdot \pi \cdot f_c \cdot (R_1 + R_2) \cdot Q} \qquad C_1 = Q \cdot \frac{R_1 + R_2}{2 \cdot \pi \cdot f_c \cdot R_1 \cdot R_2}$$

If you decide to have $R_1 = R_2 = R$ the following design equations can be used:

$$C_2 = \frac{1}{4\pi \cdot R \cdot Q \cdot f_c} \qquad C_1 = \frac{Q}{R \cdot \pi \cdot f_c}$$

Another better option with $R_1 = R_2 = R$, is to choose either C_1 or C_2. Then calculate R from this known cap, and finally find the last capacitor with the found R. This might be a good idea since caps don't come in many different values.

Witch ever gives the most usable result will do nicely.

For a more detailed look at the Sallen and Key filter, please refer to the specific document called Sallen_And_Key_ver_1.0.pdf found on the webpage for Active filter ONE

3.3 How equalizers work

An equalizer is used to manipulate the gain in a given part of the frequency band. The width of the audio band affected by the EQ depends on the Q of the equalizer.

The simplest equalizer, makes a notch, and is just a resistor in series with an LC series connection like this:

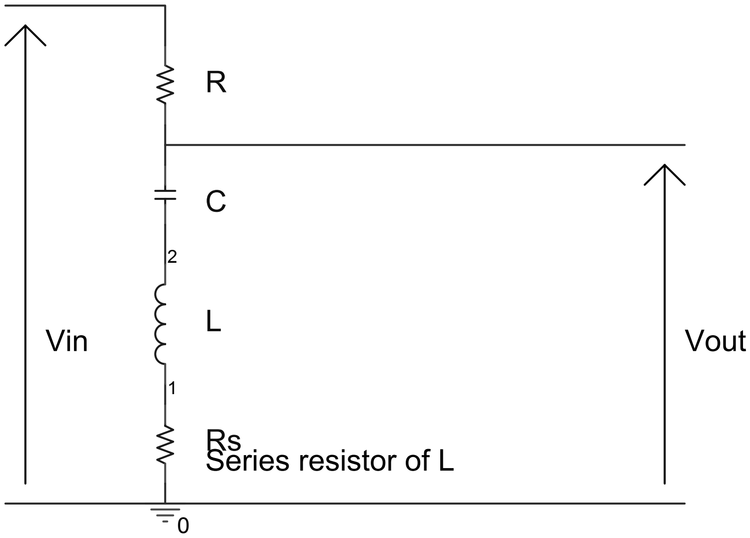
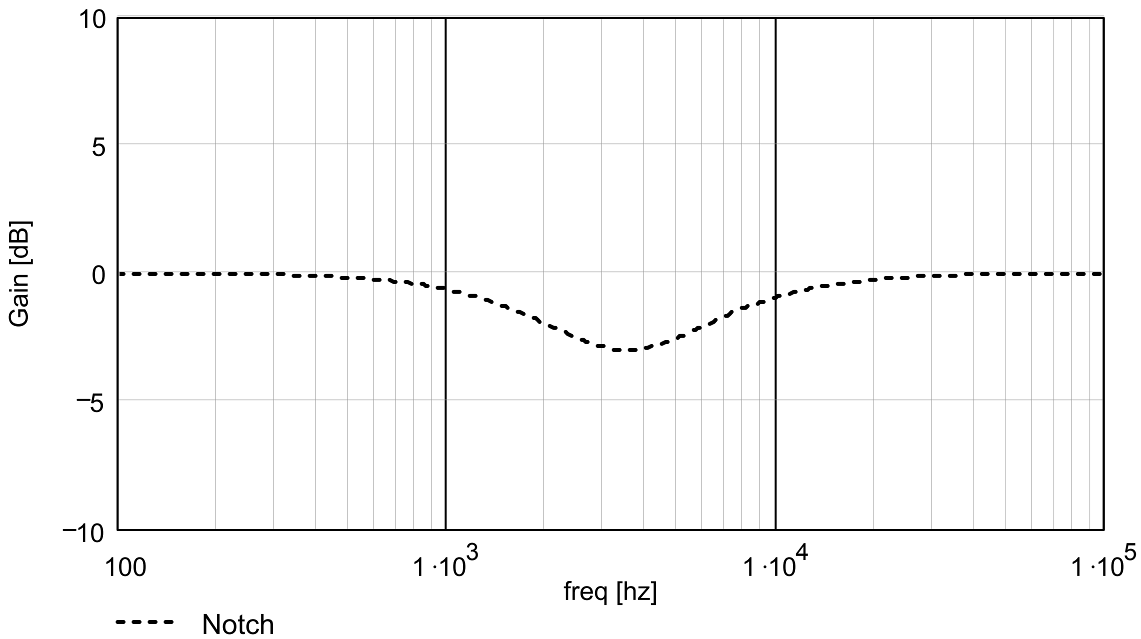


Figure 8: Simple notch EQ



It's easy to imaging what happens at the resonance frequency of the LC circuit, the circuit becomes a simple voltage divider between R and Rs. Of cause the notch filter needs to be buffered so it's not loaded in any way.

When making a peak, the same LC circuit as Figure 8 is used in the feedback path of an opamp.

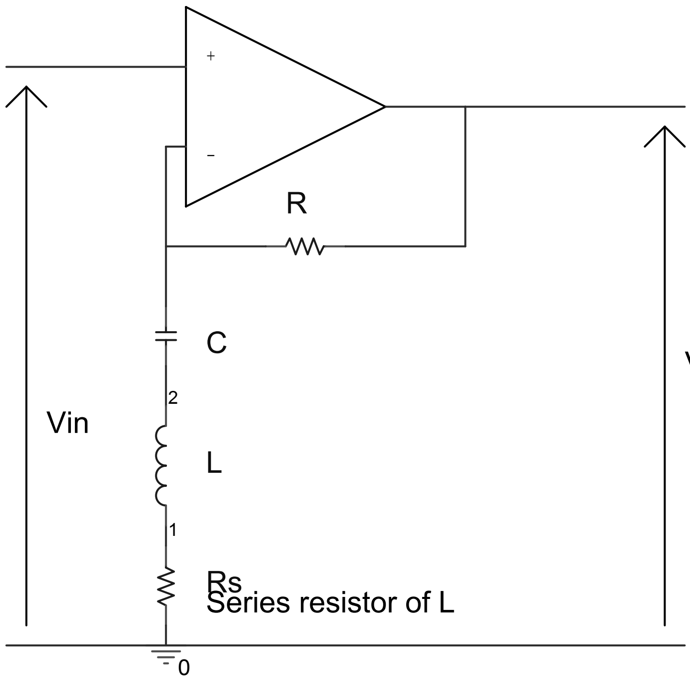
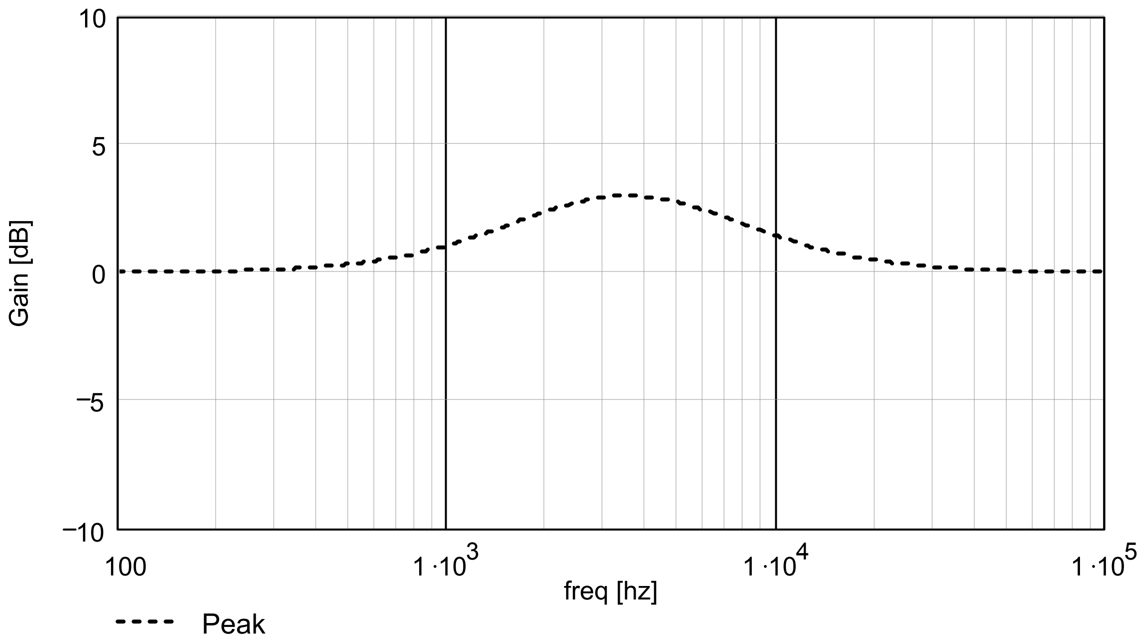


Figure 9: Simple peak EQ



The way that the circuits in Figure 8 and Figure 9 work are really great for EQ's, unfortunately inductors are difficult and expensive to make in the exact value needed for a specific EQ.

Therefore a way to work around the inductor is needed, and this is exactly what the next chapter is about.

3.3.1 The Gyrator vs. The active inductor

There are at least two ways of making a capacitor act like an induction. One is called the gyrator, and one is called the active inductor. In the following the two circuits will be examined, and one will be chosen for this filter's EQ sections.

3.3.1.1 The Gyrator

The gyrator makes a nice RCL circuit. It gives symmetrical impedance (on a logarithmic scale) around the resonance frequency.

The symmetrical impedance makes it ideal for an equalizer.

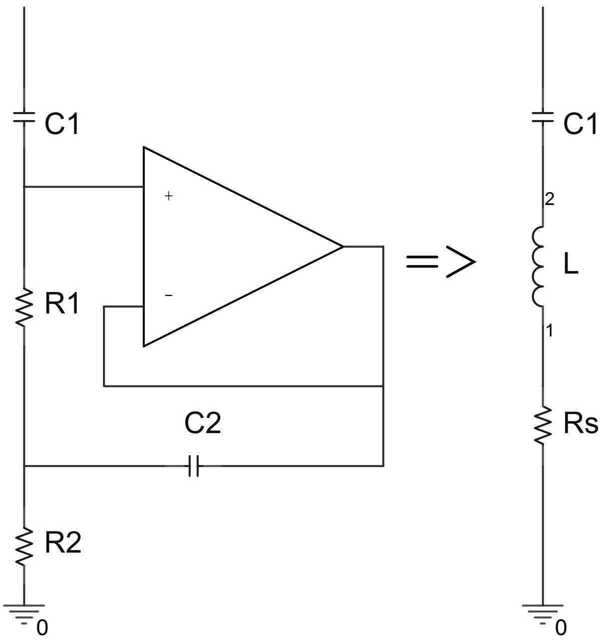


Figure 10: The gyrator

$$f_{res} = \frac{1}{2\pi\cdot\sqrt{C_1\cdot C_2\cdot R_1\cdot R_2}} \qquad R_s = R_1 + R_2 \quad L = C_2\cdot R_1\cdot R_2$$

3.3.1.2 The active inductor

The active inductor is actually just a Sallen and key HP filter drawn slightly differently. It too functions as a series connection between a capacitor and an inductor. The major difference is that the inductor has a parallel resistor. This parallel resistor makes the impedance of the circuit NON-symmetrical around the resonance frequency, thus making the circuit less suited for EQ's.

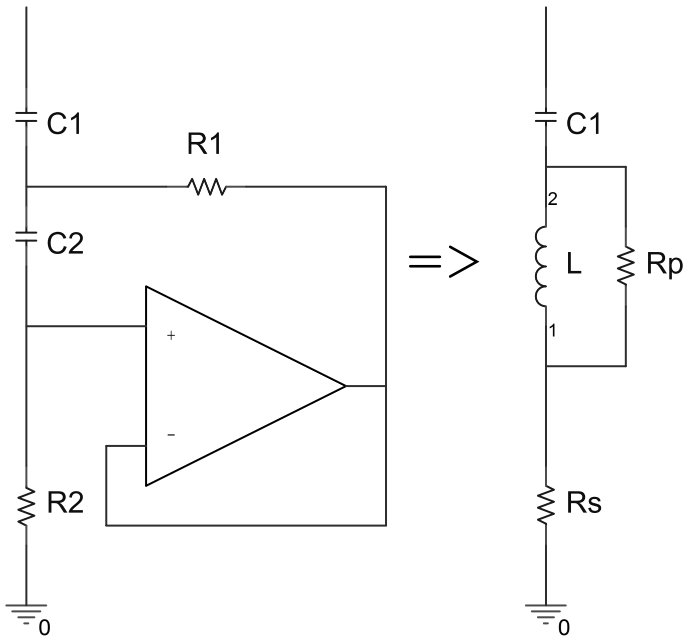


Figure 11: The active inductor

$$L = C_2 \cdot R_1 \cdot (R_2 - R_1) \quad R_s = R_1 \quad R_p = R_2 - R_1$$

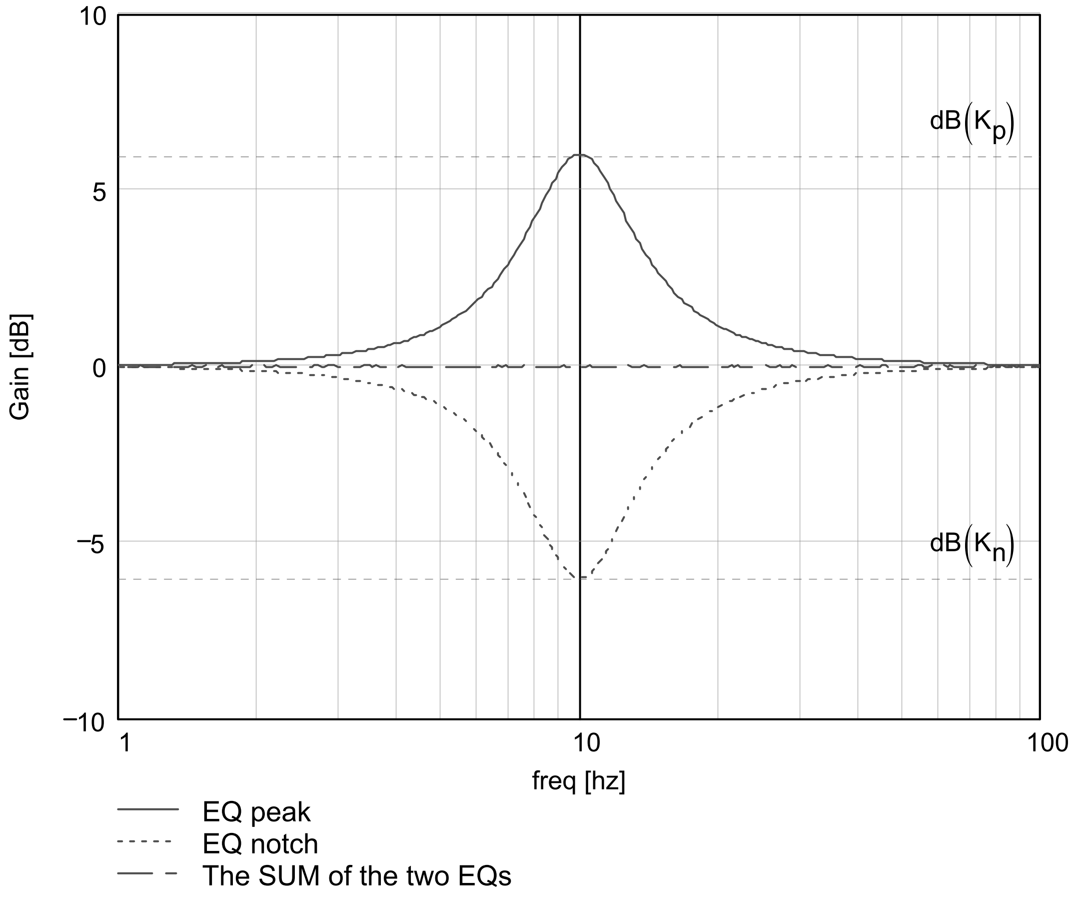
Because of this parallel resistor, the gyrator is preferred to implement the inductor in the EQ sections of this filter.

3.3.2 The mathematical EQ

The EQ is described by a transferfunction as any other second order system. There is however a minor change to the way that it is written.

$$EQ_{peak}(s) = \frac{s^2 + \frac{K_p \cdot \omega_o}{Q} \cdot s + \omega_o^2}{s^2 + \frac{\omega_o}{Q} \cdot s + \omega_o^2} \qquad EQ_{notch}(s) = \frac{s^2 + \frac{\omega_o}{Q} \cdot s + \omega_o^2}{s^2 + \frac{\omega_o}{Q \cdot K_n} \cdot s + \omega_o^2}$$

K_p and K_n are the gains of the EQ at their center frequency ω_o. Their frequency response can look like this:



The two EQs have identical Qs of 2 and f_o = 10 Hz. K_p = 2 and K_n = 0.5. The SUM of the two filters is 0 dB witch means that two EQ sections can cancel each other out, if they share Q and f_o. The gain in dB must be the same for the two (Notch is negative of cause and peak is positive)

3.3.3 The notch

When designing a notch, the filter is specified by three things:

- f_o
- Gain @ f_o called $Gain_{fo}$
- Q

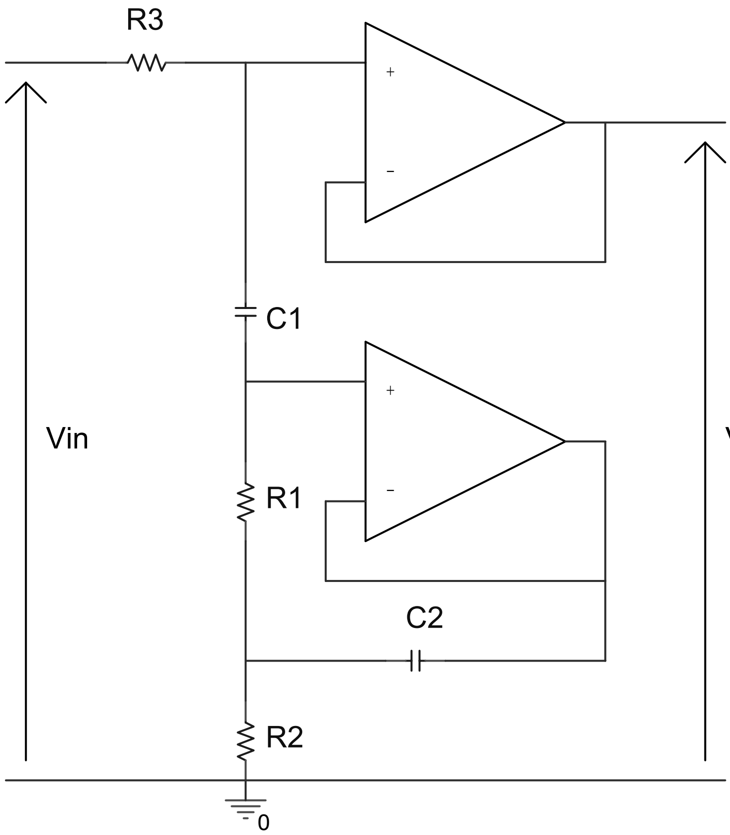


Figure 12: EQ that makes a notch, buffered

The transferfunction for the circuit is like this:

$$H_{notch}(s) = \frac{s^2 + \left(\frac{1}{C_2 \cdot R_2} + \frac{1}{C_2 \cdot R_1}\right) \cdot s + \frac{1}{C_1 \cdot C_2 \cdot R_1 \cdot R_2}}{s^2 + \left(\frac{R_3}{C_2 \cdot R_1 \cdot R_2} + \frac{1}{C_2 \cdot R_2} + \frac{1}{C_2 \cdot R_1}\right) \cdot s + \frac{1}{C_1 \cdot C_2 \cdot R_1 \cdot R_2}} = \frac{s^2 + \frac{\omega_o}{Q} \cdot s + \omega_o^2}{s^2 + \frac{\omega_o}{Q \cdot K} \cdot s + \omega_o^2}$$

With $R_1 = R_2 = R$ it simplifies to

$$H_{notch}(s) = \frac{s^2 + \frac{2}{C_2 \cdot R} \cdot s + \frac{1}{C_1 \cdot C_2 \cdot 2R}}{s^2 + \frac{R_3 + 4}{2 \cdot C_2 \cdot R} \cdot s + \frac{1}{C_1 \cdot C_2 \cdot 2R}} = \frac{s^2 + \frac{\omega_o}{Q} \cdot s + \omega_o^2}{s^2 + \frac{\omega_o}{Q \cdot K} \cdot s + \omega_o^2}$$

The simplified transferfunction makes it possible to write down some equations that can be used to find the components needed in a given EQ, K = Gain_{fo}.

$$\omega_o = 2\pi \cdot f_o \qquad \omega_o = \frac{1}{\sqrt{C_1 \cdot C_2 \cdot 2R}} \qquad \frac{\omega_o}{Q \cdot \mathrm{Gain}_{fo}} = \frac{R_3 + 4}{2 \cdot C_2 R}$$

From the voltage divider it is found that

$$\mathrm{Gain}_{fo} = \frac{2 \cdot R}{2 \cdot R + R_3}$$

When designing, start by selecting a value for C_1

This result is the following design equations.

$$R = \frac{1}{4 \cdot C_1 \cdot Q \cdot \pi \cdot f_o} \qquad R_3 = \frac{1}{2 \cdot C_1 \cdot Q \cdot \pi \cdot f_o} \cdot \frac{1 - \mathrm{Gain}_{fo}}{\mathrm{Gain}_{fo}} \qquad C_2 = 4 \cdot Q^2 \cdot C_1$$

When evaluating the values found you might find it necessary to recalculate the values with a new start value for C_1 to obtain practical component values.

3.3.4 The peak

When designing a peak, the filter is specified by three things:

- f_o
- Gain @ f_o called $Gain_{fo}$
- Q

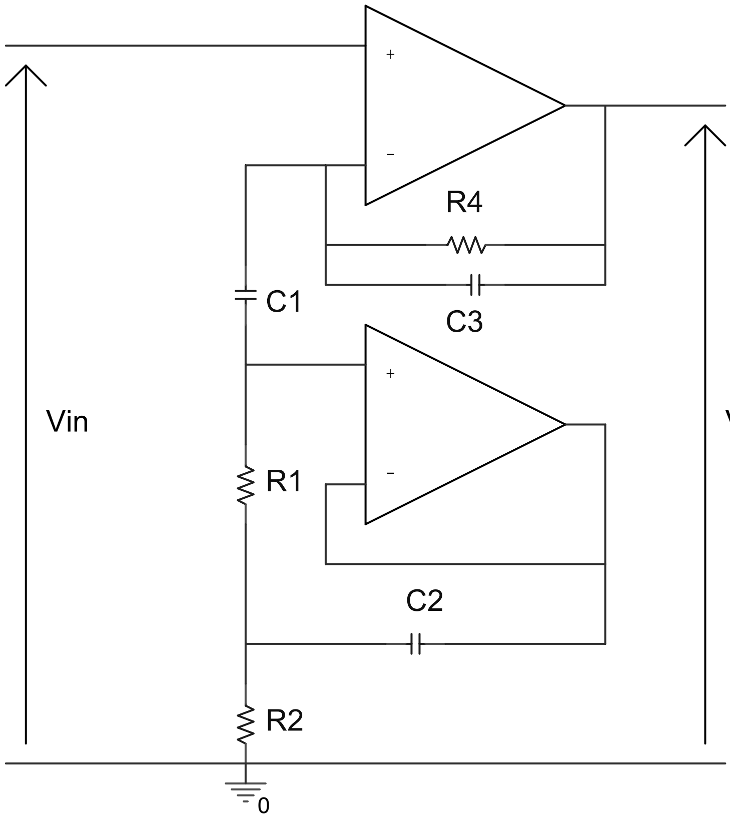


Figure 13: EQ that makes a peak

The transferfunction for the circuit is like this:

$$H_{peak}(s) = \frac{s^2 + \left(\frac{1}{C_2 \cdot R_2} + \frac{1}{C_2 \cdot R_1} + \frac{R_4}{C_2 \cdot R_1 \cdot R_2}\right) \cdot s + \frac{1}{C_1 \cdot C_2 \cdot R_1 \cdot R_2}}{s^2 + \left(\frac{1}{C_2 \cdot R_2} + \frac{1}{C_2 \cdot R_1}\right) \cdot s + \frac{1}{C_1 \cdot C_2 \cdot R_1 \cdot R_2}} = \frac{s^2 + \frac{K \cdot \omega_o}{Q} \cdot s + \omega_o^2}{s^2 + \frac{\omega_o}{Q} \cdot s + \omega_o^2}$$

With $R_1 = R_2 = R$ it simplifies to

$$H_{peak}(s) = \frac{s^2 + \frac{R_4 + 4}{2 \cdot C_2 \cdot R} \cdot s + \frac{1}{C_1 \cdot C_2 \cdot 2 \cdot R}}{s^2 + \frac{2}{C_2 \cdot R} \cdot s + \frac{1}{C_1 \cdot C_2 \cdot 2 \cdot R}} = \frac{s^2 + \frac{K \cdot \omega_o}{Q} \cdot s + \omega_o^2}{s^2 + \frac{\omega_o}{Q} \cdot s + \omega_o^2}$$

Again the simplified transferfunction gives us a set of equations.

$$\omega_o = 2\pi \cdot f_o \qquad \omega_o = \frac{1}{\sqrt{C_1 \cdot C_2 \cdot 2 \cdot R}} \qquad \frac{\omega_o}{Q} = \frac{2}{C_2 \cdot R}$$

From the gain formula of the non inverting opamp it’s found that

$$Gain_{fo} = \frac{2 \cdot R + R_4}{2 \cdot R}$$

Start by selecting a value for C_1

This results in the following design equations.

$$R = \frac{1}{4 \cdot f_o \cdot \pi \cdot C_1 \cdot Q} \qquad R_4 = \frac{Gain_{fo} - 1}{f_o \cdot 2 \cdot \pi \cdot C_1 \cdot Q} \qquad C_2 = 4 \cdot Q^2 \cdot C_1$$

When evaluating the values found you might find it necessary to recalculate the values with a new start value for C_1 to obtain practical component values.

C_3 is there to ensure stability for the opamp. It should be selected carefully so it does not interfere with the operation of the EQ. There will be more about C_3 in the design example.

4 What components to use

The following section is a guide to choose your components for your active cross over. Please note the following table to make sure your components fit in the drilled PCB

Components	Hole size
Opamps	0.8 [mm]
Capacitors	0.8 [mm]
Resistors	0.8 [mm]
Jumpers / connectors	1.1 [mm]
Mounting holes	3.2 [mm]

Table 1: Holes in the PCB

4.1 Opamps

Several opamps can be used for the crossover. The PCB layout is made for dual opamps with standard 8 pin DIL casing. I recommend low noise types. I have made a list with some of the types I like to use.

Type	Approximate Price US$
MC33078P	1
NE5532	1
LM833N	2
OPA2134	3

Table 2: List of opamps

If you don't want to spend too much money on opamps I recommend using the OPA2134 for buffer and treble section and the LM833N or MC33078 for the rest. However it's completely up to the designer of the crossover to choose the opamps, since there are so many different types to choose from I have no chance of testing them all. If you are in doubt of what type to get, feel free to contact me for help.

4.2 Capacitors

I prefer polypropylene (PP) capacitors for signal processing, but polyethylene (PE) can also be used. The PCB has room for box capacitors sized 5x7 [mm]. The pitch distance is 2 modules (about 5 mm). To get a good accuracy between the wanted frequency response and the final result please only use 5% or better tolerances. A cheap way of obtaining good tolerances is by means of a capacitor meter to measure a bunch of cheaper 10 % capacitors. I use a Monacor CM-200 it ranges from 200 pF to 20 mF, witch is all I need for my projects.

The caps for decoupling the supply (Designators C801–C824 and C901–C924) are boxed 100 nF caps sized 2.5x7 [mm] with a pitch distance of 2 modules (about 5 mm)

The lager caps for supply decoupling (Designators C900 and C800) are your favorite electrolytic 100µF cap. The maximum diameter is about 10.5 [mm] and a pitch distance of 5 [mm]

4.3 Resistors

The PCB is made for ¼ watt resistors. To get a good accuracy between the wanted frequency response and the final result please use 1% or better tolerances.

5 Getting started with filter design

When starting a new filter design carefully think about what you need the filter to be able to do. Use the block diagram as a guide to find out how many sections you require, and determine the different f_c's, gains and Q's you need.

Note all the info you have, including output level of the filter (what is your amps sensitivity/gain) and the output voltage of your source.

Output level of source [V]	Input sensitivity of power amps			
	SUB	Bass	Midrange	Treble amp

EQ's	Gain @ fo [dB]	fo [Hz]	Q
SEQ1			
SEQ2			
SEQ3			
BEQ1			
BEQ3			
BEQ3			
MEQ1			
MEQ2			
MEQ3			
TEQ1			
TEQ2			
TEQ3			

Global EQ's	Gain @ fo	fo [Hz]	Q
QEQ1			
QEQ2			

HP and LP	HP fc freq [Hz]	Q HP	LP fc freq [Hz]	Q LP
SUB				
BASS				
Midrange				
Treble			---------------------	---------------------

Having gathered all the info needed, we are ready to proceed to the design of the active crossover.

Be sure to keep the tables for later reference.

6 Designing a two way cross over with two EQ's

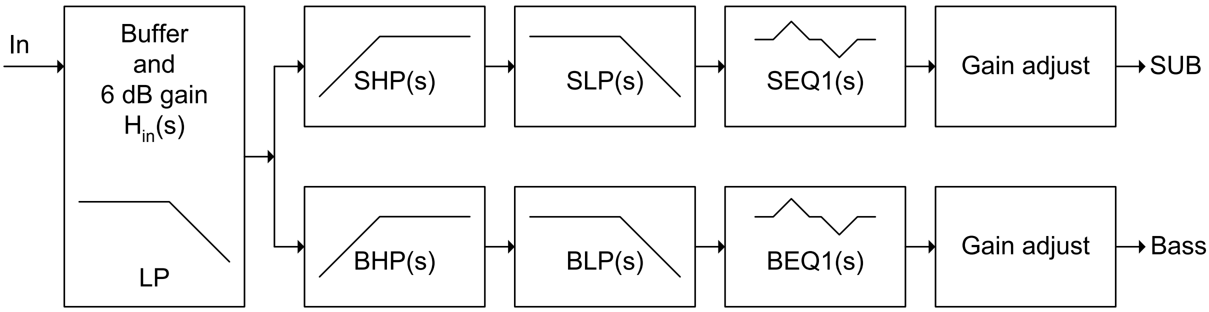
This chapter gives an example on how the PCB module can be used. It features a two way crossover using two EQ, a peak and a notch.

6.1 Making a plan

The following data are given for a design used to drive a subwoofer with a set of normal speakers.

We would like to have a Linkwitz Riley response so a Q of 0.707 is needed when two identical 2.order filters in series are used to form a 4.th order function.

The linkwitz Riley means that there will be no "bumps" in the freq response where the two filter sections (HP and LP) overlap each other in the frequency band.



Output level of source [V]	Input sensitivity of power amps			
	SUB	Bass	Midrange	Treble amp
1 V RMS	1 V	1.3 V	------------------	------------------

HP and LP	HP fc freq [Hz]	Q HP	LP fc freq [Hz]	Q LP
SUB	10	1.2	140	0.707
BASS	140	0.707	50000	0.707

EQ's	Gain @ fo [dB]	fo [Hz]	Q
SEQ1	2	25	0.9
BEQ1	-0.8	1000	0.6

6.2 Calculating the component values

Using the formulas given in the schematics section the following components can be found:

INPUT

$R_{001} = 10 \times 10^{3}$

$R_{002} = 10 \times 10^{3}$

$R_{003} = 100 \times 10^{0}$

$R_{004} = 10 \times 10^{3}$

$C_{001} = 100 \times 10^{-12}$

$C_{002} = 100 \times 10^{-12}$

SUB HP

$R_{031} = 15 \times 10^{3}$

$R_{032} = 82 \times 10^{3}$

$R_{041} = 15 \times 10^{3}$

$R_{042} = 82 \times 10^{3}$

$C_{031} = 470 \times 10^{-9}$

$C_{032} = 470 \times 10^{-9}$

$C_{041} = 470 \times 10^{-9}$

$C_{042} = 470 \times 10^{-9}$

SUB LP

$R_{051} = 3.6 \times 10^{3}$

$R_{052} = 3.6 \times 10^{3}$

$R_{061} = 3.6 \times 10^{3}$

$R_{062} = 3.6 \times 10^{3}$

$C_{051} = 470 \times 10^{-9}$

$C_{052} = 220 \times 10^{-9}$

$C_{061} = 470 \times 10^{-9}$

$C_{062} = 220 \times 10^{-9}$

BASS HP

$R_{101} = 3.6 \times 10^{3}$

$R_{102} = 7.3 \times 10^{3}$

$R_{111} = 3.6 \times 10^{3}$

$R_{112} = 7.3 \times 10^{3}$

$C_{101} = 220 \times 10^{-9}$

$C_{102} = 220 \times 10^{-9}$

$C_{111} = 220 \times 10^{-9}$

$C_{112} = 220 \times 10^{-9}$

BASS LP

$R_{121} = 10 \times 10^{3}$

$R_{122} = 10 \times 10^{3}$

$R_{131} = 10 \times 10^{3}$

$R_{132} = 10 \times 10^{3}$

$C_{121} = 440 \times 10^{-12}$

$C_{122} = 220 \times 10^{-12}$

$C_{131} = 440 \times 10^{-12}$

$C_{132} = 220 \times 10^{-12}$

SUB EQ1

$R_{071} = 75 \times 10^{3}$

$R_{072} = 75 \times 10^{3}$

$R_{073} = 0 \times 10^{0}$

$R_{074} = 36 \times 10^{3}$

$C_{071} = 47 \times 10^{-9}$

$C_{072} = 150 \times 10^{-9}$

$C_{073} = 100 \times 10^{-12}$

BASS EQ1

$R_{141} = 56 \times 10^{3}$

$R_{142} = 56 \times 10^{3}$

$R_{143} = 12 \times 10^{3}$

$R_{144} = 0 \times 10^{0}$

$C_{141} = 2.2 \times 10^{-9}$

$C_{142} = 4.7 \times 10^{-9}$

$C_{143} = 100 \times 10^{-12}$

Output Level adjust

$R_{292} = 4.7 \times 10^{3}$

$R_{293} = 100 \times 10^{0}$

$R_{302} = 4.7 \times 10^{3}$

$R_{303} = 100 \times 10^{0}$

R291 and R301 could be a 22 kohm multi turn pot

10^3 means kohm, 10^-9 means nF and 10^-12 means pF

Please note that these are not the only component values that will give the wanted filter.

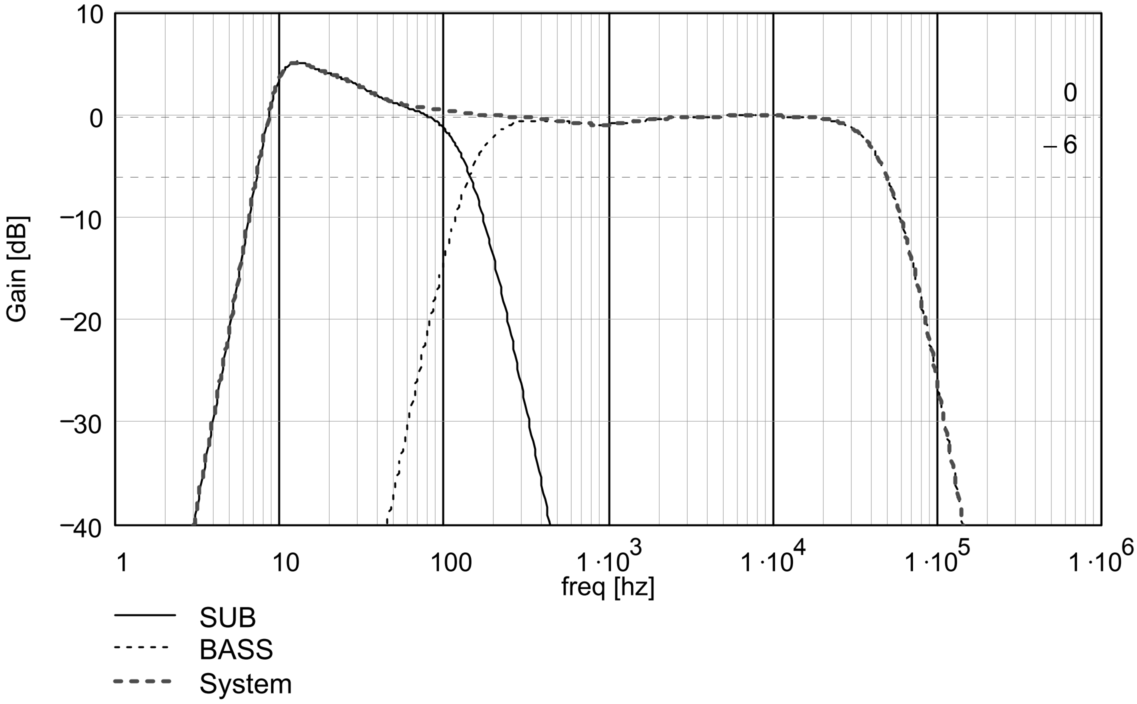


Figure 14: Resulting freq response

7 Construction details

This chapter is a detailed description of construction assembly and test of the filter. When a filter design is ready, and all the components have been worked out and double checked, please proceed with the assembly of the filter.

7.1 Connections

There are six external connections on the PCB

- Supply
- In
- Sub
- Bass
- Midrange
- Treble

7.1.1 Supply, Signal in and out

In order to get good performance of the filter, a well regulated power supply of +-15V is needed. The current needed differs with the number of opamps used, but a good estimate is 0.75 Ampere per fully populated filter PCB.

The supply connection is Pin

1. + 15V
2. Gnd
3. -15V

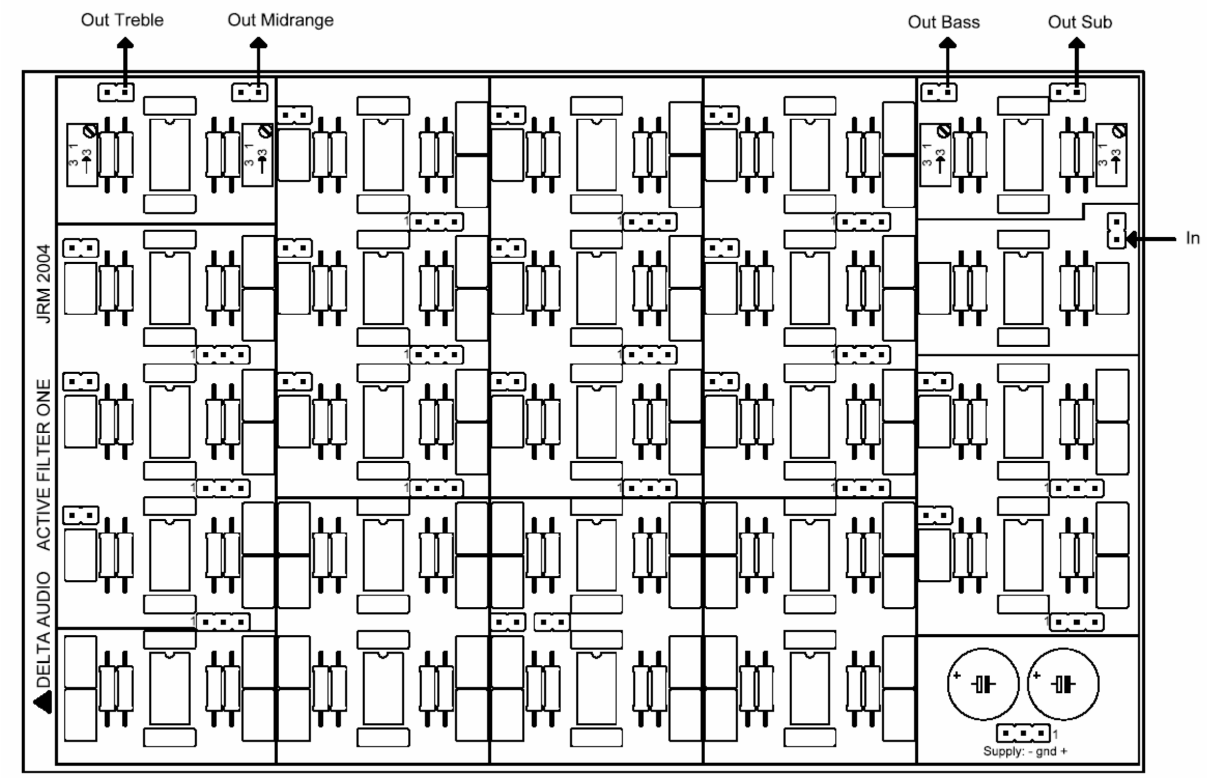
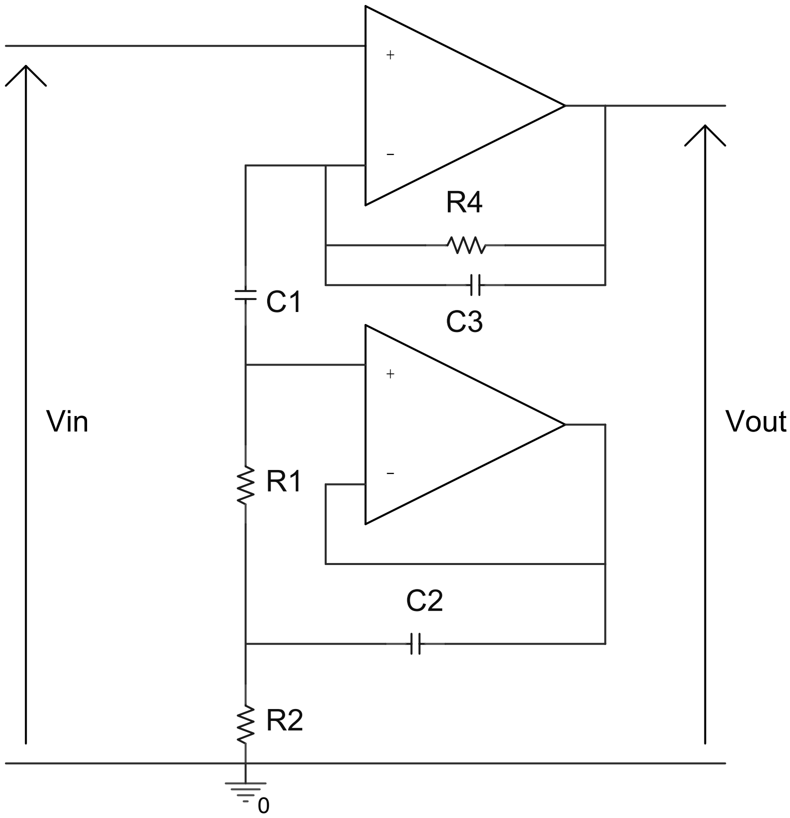


Figure 15: External connections

7.2 Stability cap Cxx3

If an EQ section is used as a peak you must ensure that the opamp doesn't oscillate. This is done by means of a cap C_{XX3} across the top resistor in the feedback loop. The size of the cap is determined by the frequency of the peak. At low frequencies a larger cap can be used, but the higher the frequency is, the smaller the cap must be to ensure that the operation of the equalizer remains undisturbed. I have generally used 10 pF because then the frequency of the peak can be chosen freely, without worrying about weird behavior of the EQ caused by the cap. If in doubt contact me, and I will help you.



8 Test and adjustment

When all the needed components and jumpers for a project are mounted on the PCB, it's time for double checking the components and the testing.

Start by connecting a turned off +- 15 V supply to the module, making sure that the polarity is right. Slowly adjust the voltages while watching the current meter on the supply, this must newer exceed a current of a few hundred milliamps. If the current is higher, turn off the power supply and start fault finding.

When everything is ok, connect a source to the input and a scope or RMS voltmeter to one of the outputs. Adjust the source to a frequency where the channel you test has 0 dB gain. Measure the input voltage and write it down. Then adjust the potentiometer, while monitoring the output, until the same value is measured on the output as on the input.

Do this for all the channels that you have used, when done your filter is ready to be used with amps and speaker that have the same sensitivity.

If you have amps with different sensitivity you must measure the output voltage of the amp while it's connected to the filter. Be careful when you do this, some amps have very high voltage capabilities that can cause serious electrical shock – and in worst case it can kill you.

9 Parts list

The parts list doesn't include a lot because most of the filter components are dependant on the design of the filter. Using the included values for the input filter and output level adjust most designs will be ok. If in doubt of what to use, please don't hesitate to contact me.

Component	**Value**	**Note**
IC1- IC24	MC33078P LM833N OPA2134 NE5532 …your choice	Make sure that the opamp can run on a +- 15 V supply. Use IC sockets in case you want to change opamps later
C800 + C900	100µF 25V	Pitch distance = 5[mm] Diameter = 10.5 [mm]
C801→C824+C901→C924	100 nF 63V	Pitch distance = 5 [mm] Boxed 2.5x7 [mm]
R001	10 kohm ¼ W	
R002	10 kohm ¼ W	
R003	100 ohm ¼ W	
R004	10 kohm ¼ W	
C001	100 pF	
C002	100 pF	
R291+R301+R311+R321	22 kohm multi turn pot	
R292+R302+R312+R322	10 kohm ¼ W	
R293+R303+R313+R323	100 ohm ¼ W	