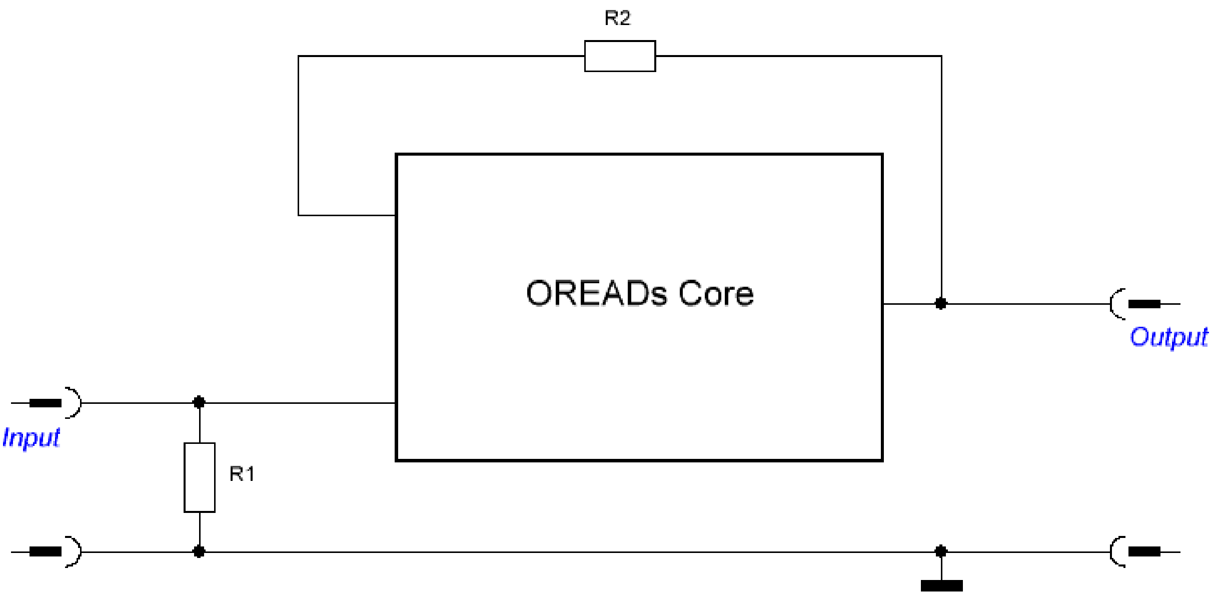
A guide

on the way to my own MM phono (equalizer) preamplifier

§1



In the first step, we define R1 equal to R2, for example 150kΩ.

The actual amplifier has two completely equal inputs; we see them on the left side of the large rectangle. Hidden behind these inputs is the so-called difference point; the amplifier (in the core) forms a difference from what it literally sees at its inputs. It processes this difference internally and forwards it to its output in a proportionally enlarged form, shown here on the right side.

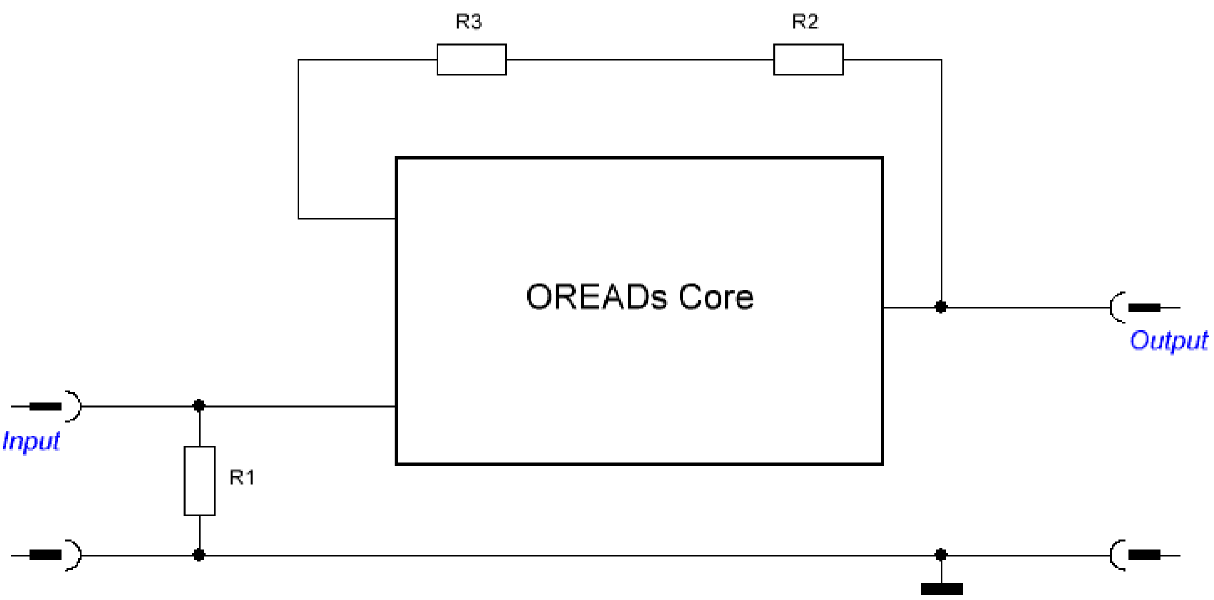
At the moment it sees two identical states at its inputs, the difference is zero, i.e. its output will also have this zero. And that's exactly what you want as a good designer. Two resistors of different sizes would represent a difference and would inevitably lead to a corresponding response from the amplifier. The output would carry an undesirable dc-voltage.

The perspective taken is also called static observation. It is necessary to distinguish between static and dynamic. When it comes to the initial dimensioning of the component values of the actual discrete amplifier, our view is static.

Another distinction concerns the small difference between static and stationary. But that definitely takes us too far away from where we started, this introduction.

In the next step, we divide the resistor R2 into two parts, which together correspond unchanged to R1 in value, i.e. R1 = $R2_{new}$ + R3.

§2



With the following rule it is quite simple: "R2 divided by R3 equals 11.78 with sufficient accuracy".

R2/R3 = 11,78

R3 = R1 / (1 + 11,78)
= 11,737kΩ

R2 = 11,78 * R3
= 138,263kΩ

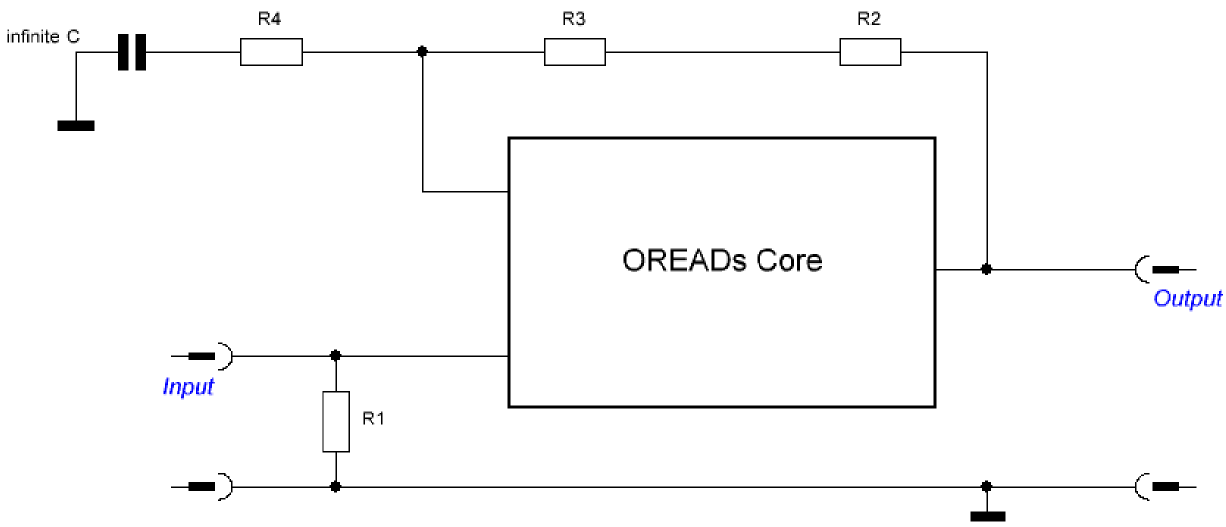
Of course, the positions of both real resistors are interchangeable, the only important thing is that their series connection corresponds in value to that of R1.

It is not yet clear why we should proceed in this way, but at the end of the journey the view becomes clearer. Something is definitely missing: namely the amplification. Our ideal core is fully feedbackcoupled and represents a so-called impedance converter, it decouples the left side from the right side. The amplification factor is 1 or 0dB.

As long as the differential voltage is zero everything is fine. The upper branch, which we give the name negative input, essentially looks at its own output and this must now obediently follow what is happening at the other input.

The returned voltage must be reduced to a fraction (that will automatically correspond to our input signal), then there is a amplification, because the difference between the two inputs must be zero.

§3



What level of signal amplification do we want? Unfortunately, we are only allowed to express a limited wish in this respect, because ultimately a typical frequency-dependent course of this amplification should be achieved. For the time being, we therefore assume a linear factor of 10 * 100 = 1000 or 60dB.

1000 = [R4 + (R3+R2)] / R4
= 1 + (R3+R2) / R4

R4 = 150kΩ / 999
= 150,15Ω

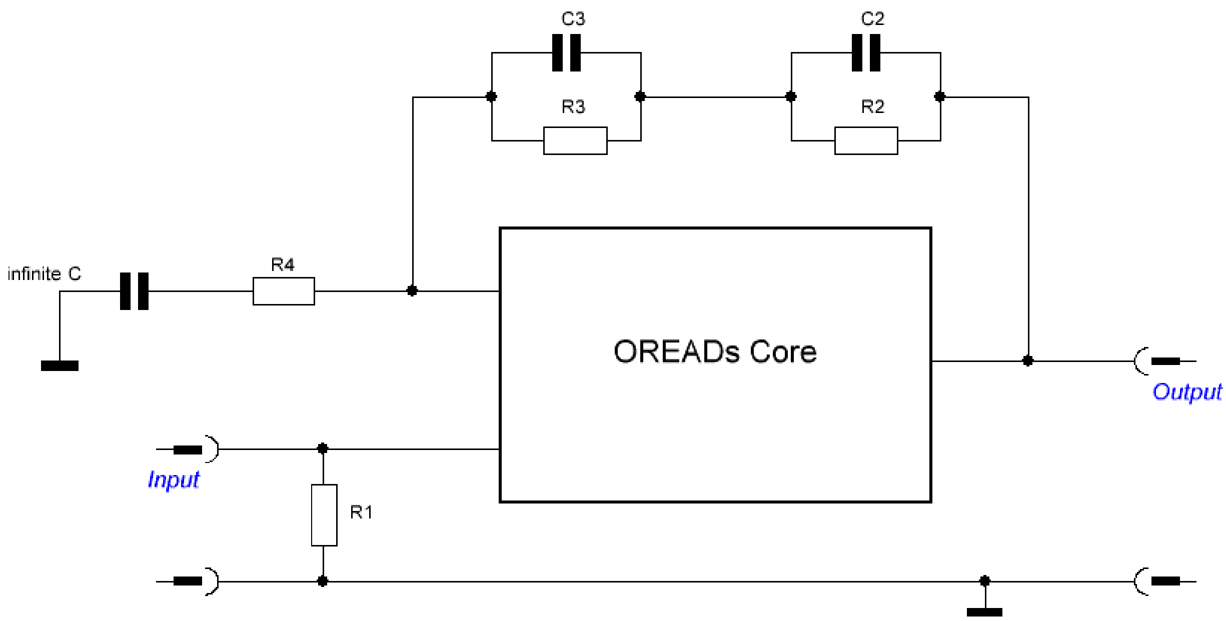
This factor is explained by the fact that the so-called non-inverting amplifier circuit can only amplify but not attenuate - this is definitely a disadvantage compared to the second form, the inverting circuit. However, both types also differ fundamentally in other areas. But the exact discourse takes us too far here.

The factor 1000 will be explained later.

By inserting the R4 we realize a voltage divider and the series feedback. The infinitely large (inserted) series capacitor ensures that our voltage divider only works dynamically, for alternating voltages. DC voltages are not amplified, but are transmitted with a maximum factor of 1. They appear at the output (yes) and the capacitor charges to this value (and block).

In the following step, we will take care of the linear distortion and reset it in accordance with the RIAA-Recommendation.

§4



As the signal frequency increases, the gain decreases from 1000 to 10, i.e. +60dB at the beginning of the audible frequencies to +20dB at the end of the audio band.

1000/10 = 100, +40dB

With the appropriate ratio of C2 to C3 and the products C3*R3 & C2*R2, the factor 100 is set for a signal frequency of 1kHz. Now, a typical voltage of 5mV at the input of the equalizer leads to an output voltage of 0.5V.

Above a signal frequency of 2kHz, the gain factor will fall linearly, with every tenfold increase by a factor of 10, i.e. the slope falling at -20dB per frequency decade. We recognize this from the complex transfer function or purely intuitively. But in this case our factor cannot be smaller than 1, i.e. not < 0dB. And we should move this point on the X-axis, the frequency, far away from the audio range to the right. Around 200kHz is already so good that we can forego a correction that would otherwise be necessary. This is the short explanation of the 1000, see §3.

C2/C3 = 3.6 is the aforementioned quotient. However, it does not yet help us from the perspective we have adopted. We need the so-called time constants, and the rest is an easy calculation.

C3 = 75µsec / R3
= 0,000075sec / 11,737kΩ = 6,39nF

C2 = 3,183msec / R2
= 0,003183sec / 138,263kΩ = 23,021nF

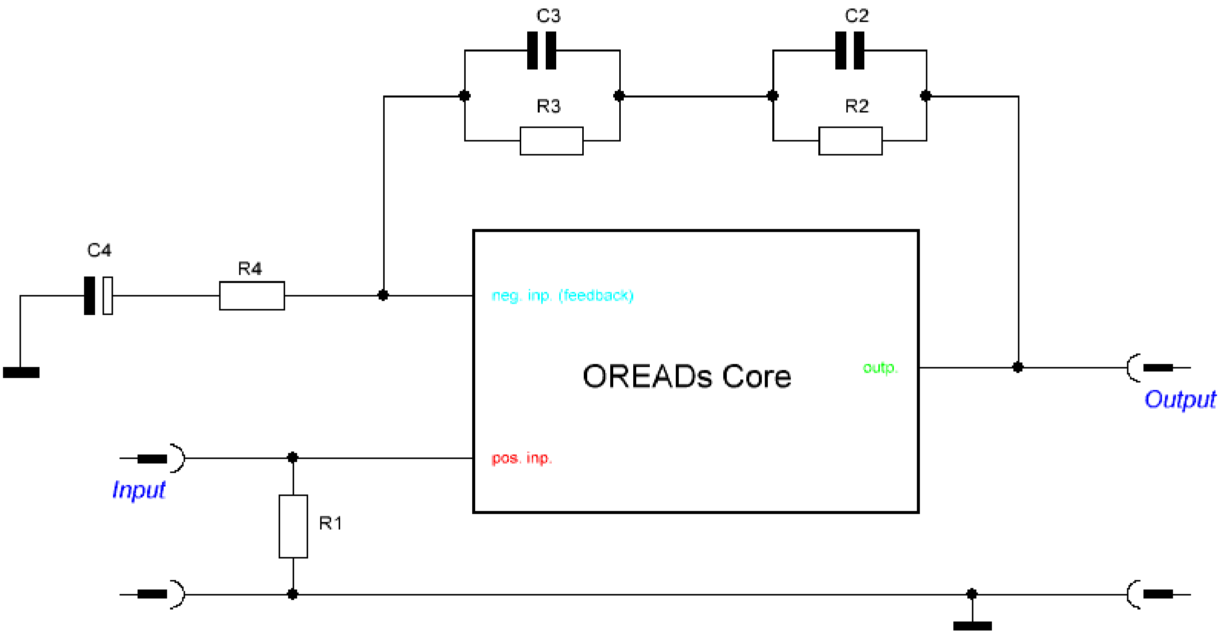
We calculate (R3||R2) * (C3+C2), this product must correspond to 318.186µsec (as you can see from the numerator of the H(jω) in the headline) quite exactly.

If we have already seen the descriptive means of amplitude frequency response and phase frequency response, in short the frequency response or Bode plot, then one could say: from 50Hz we fall linearly (constantly) to 500Hz and then take a break until around 2kHz and continue to fall linearly to infinity. In our case, this infinity starts at just under 200kHz. This value guarantees us, among other things, that we can perfectly reproduce sinusoidal oscillations of 20kHz and can even completely reconstruct other time functions.

I think this flood of information has to be digested before mathematics can spread in blessings.

The only task of our infinitely large series capacitor $C_{infinite}$ is to represent an infinitely large reactance for signal frequencies approaching zero. $X(f) = 1 / (\omega * C_{infinite})$ unfortunately, this ideal component does not exist.

§5



We select a suitable electrolytic capacitor and calculate a reference point for the value of C4. As R3+R2 is considerably larger than R4, we only consider the series connection of R4 & C4 and define |X_C4| = R4.

But at what frequency should the amounts be the same? at f<2Hz.

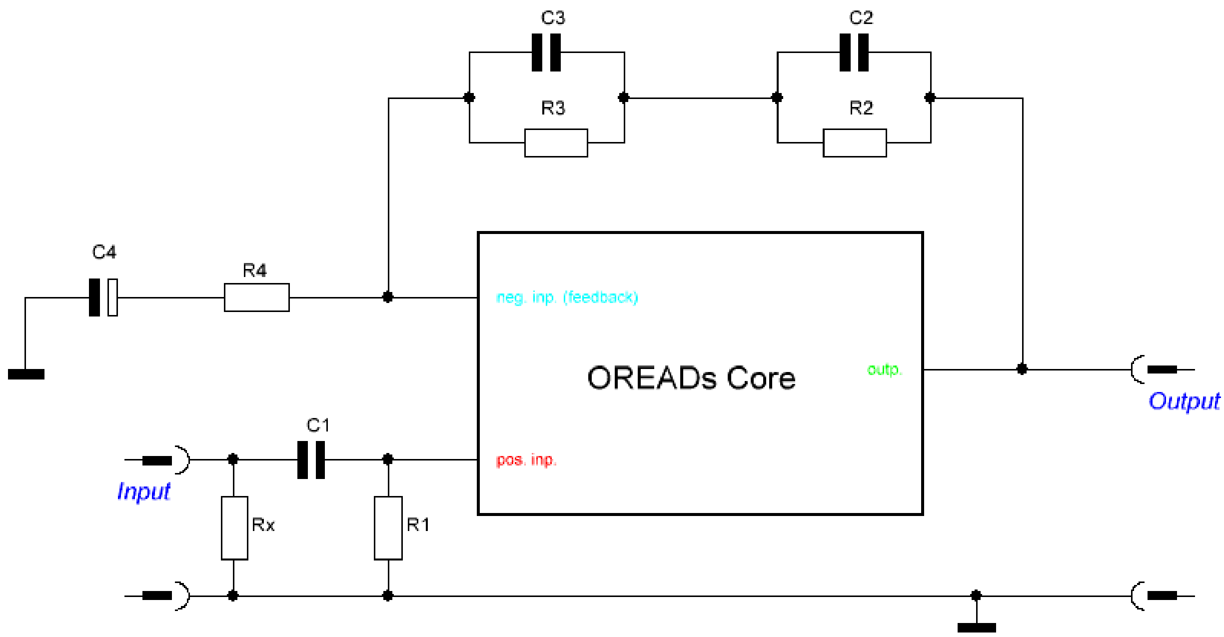
C4 > 1 / (2*PI*f * R4)
   > 530µF

The theoretical requirement f<<1Hz is much better. It would be too far to go deep into the background, but: outside the beginning audible range it doesn't distort and doesn't really change the phase; outside means at least $1/10^{th}$ - and would go beyond the scope of this introduction.

However, it is important to remember that C4 must never determine the lower cut-off frequency „fl“ of the equalizer.

The first-order high-pass filter at the input of our preamplifier is predestined for this.

§6



C1 must not be an electrolytic capacitor now, an electrolytic capacitor would have to be dimensioned differently than a foil type at this point (see §5; as a signal coupling C at least 1/100th).

We should set the cut-off frequency at 5Hz and calculate a reference point:

C1 > 1 / (2*PI*5Hz * 150kΩ)
> 212,21nF

The influence on the phase rotation and thus also on so-called group delay distortions is negligible, but we have still set a lower limit frequency. In many cases this is good for the noise, but less so with the phono equalizer, but in return we attenuate the deepest disturbances a little.

Rx (and Cx, not used in OREAD) is known to be determined by the preferred pickup (from the manufacturer's specifications), with Rx = 68.45kΩ the system would see a termination of 47kΩ in parallel Cx if it could see into the input.

Remark:
All six steps or paragraphs assume that the core behaves like an ideal operational amplifier. The search is therefore for a practical ideal that can be used for the respective application.

MM's correct termination results in a 2nd order low pass function with an upper cutoff frequency between 15kHz and 20kHz.