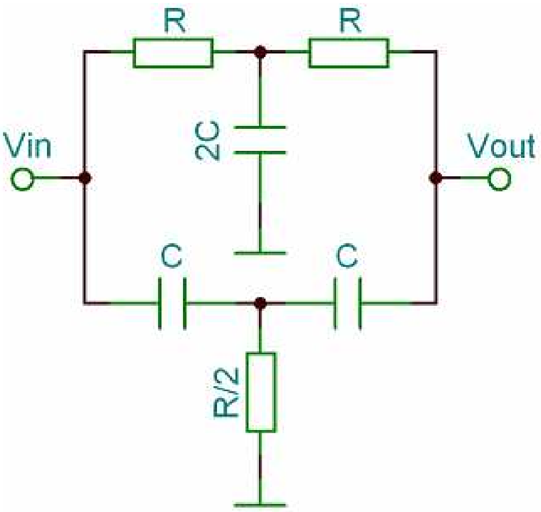
Electronics II. laboratory

Active filters and oscillators

Theory

TT filter

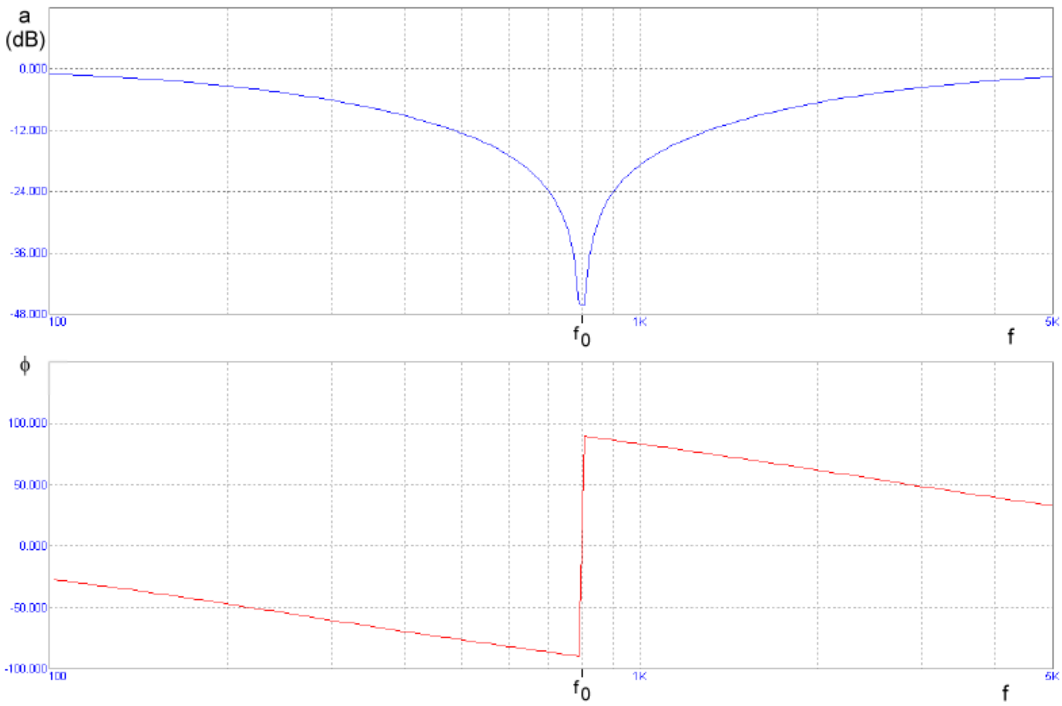


1. TT filter

The double-T or TT filter is a passive RC circuit. When connected in series, it acts as a band stop filter (BSF).

This filter has a very different phase plot in ideal and real cases (while the amplitude plots remain similar).

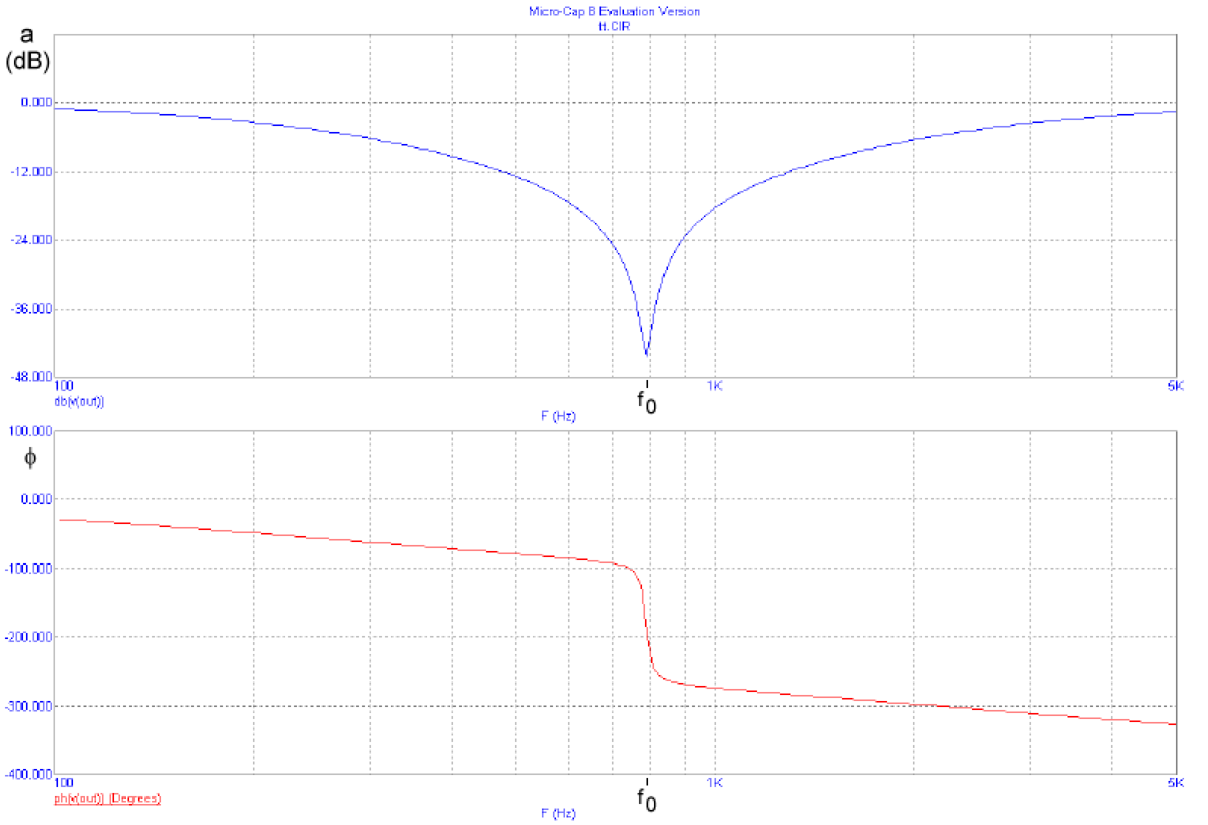
In ideal case all R and C values are exact. Then the Bode plot looks like this:



2. Ideal TT filter Bode plot approximation

At resonance frequency f_0 the output ideally is zero (minus infinite dB). In the image it is not, because it is from a simulation using discrete frequency and time steps etc.

In real case the R and C values have some tolerance (variation), that is, their values are off by a few percents (for the resistors, it's usually max 1%, for the capacitors it can be more). Then:



3. Bode plot approximation of non-ideal TT filter

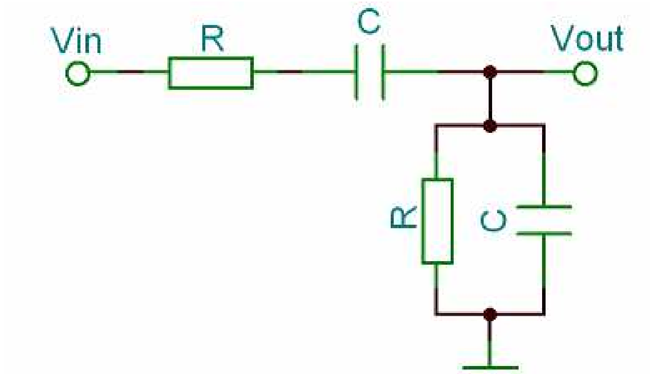
In ideal case, around f_0 the phase jumps from -90° to +90° and at f_0 is at 0°.

In non-ideal case, it goes from -90° to -270° and is at -180° at f_0.

The amplitude graphs stay similar. For ordinary filters, therefore they behave similarly.

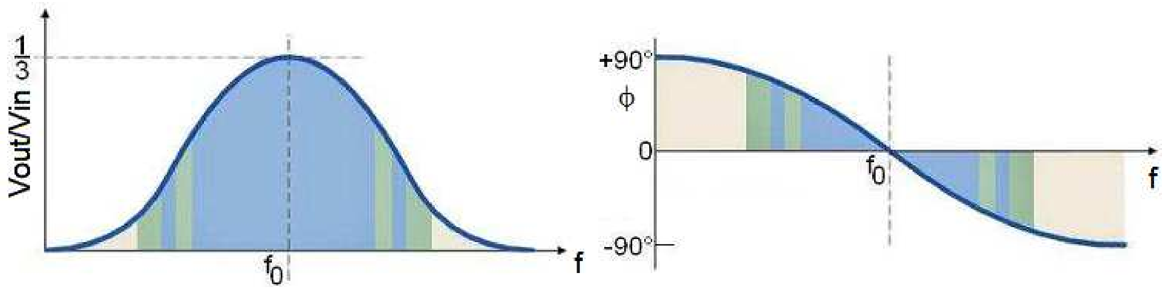
When building oscillator from feedback amps, the phase diagram does matter. The phase condition is that the whole loop should have 0+n·360° phase. This is realized by the TT doing -180° at f_0 and connecting to the inverting input of opamp gives another -180°. The two together gives then 360° and so a positive feedback. That's an example when feedback to the inverting input doesn't cause negative, but positive feedback.

Wien-bridge



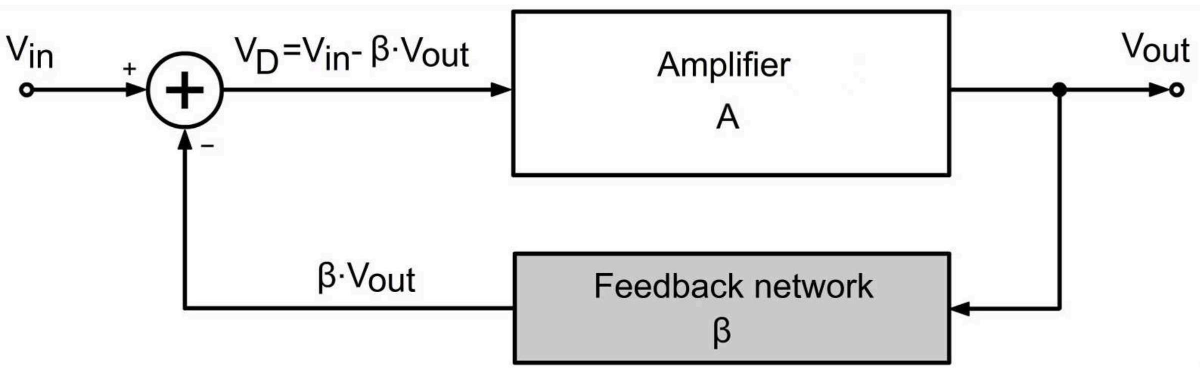
4. Wien-bridge

The Wien-bridge is also created from RC sub-circuits. At low frequencies the series C blocks the signal. At high frequencies the parallel C shorts the signal. As such the circuit will have a band pass filter behaviour.



5. Wien-bridge Bode-plot

On feedbacks



6. General block diagram for explaining feedback

The figure shows the classical model of a feedback amplifier. In this a fraction of the output signal is subtracted from the input. (In some models it is added, but subtraction is more often used and is often more practical. The end result is the same with opposite sign.)

If B is positive, this gives a negative feedback. The opamp circuits from Electronics I. are good examples for this. (They also show why subtraction is useful - because then B can be simply the voltage divider formula of the resistors.) In negative feedback, the resulting gain is less than A.

If B is negative (or has 180° phase shift) then we get positive feedback , ie. a fraction of the output is added to the input. The gain at the end will be greater than A.
If the feedback is "strong enough", then the output will increase without bounds, or at least until the power supply voltage stops it, the gain is practically infinite. For normal amplifiers,

we need to avoid positive feedback. It can usually appear from unwanted capacitances and inductances, especially with multi-stage amplifiers. Homework: connect a loudspeaker and a microphone to an amplifier (your PC will do it, just enable the speaker to output the boosted mic signal) and put mic and speaker close to each other. Then run before others in your household come investigating.

One category of oscillators are created by positive feedback amps. If the feedback network B behaves as a band pass filter, with amplitude being maximal at f_0 and phase being 180° at f_0, then the positive feedback can only appear at f_0. Thus a sine wave at f_0 will appear at the output. The interesting point is, that after the output has stabilised, we can disconnect the input and the output will remain, as the feedback supplies now the input. (Of course the circuit does have a DC power supply to gain energy from.) What is even more interesting is that we don't have to supply an f_0 input signal at all. If we have some noise (such as thermal noise or electromagnetic noise/interference received), which contains a large spectrum including f_0, then even this very small power can be enough to start the oscillator (because of the infinite gain). The filter will filter out f_0 from the noise and suppress the rest. The L and C components in the circuit can also provide a starting impulse transient if the circuit is started by L and C being un-energised.

So in ideal case the conditions of oscillation:

- The amplitude of the loop's total gain is unity: $|A \cdot B|=1$.
  This is hard to realise. If it is slightly less, the oscillation will start but then soon decrease and stop. If it is greater than one, the output voltage will increase and then will be limited by the power supply and we get a square wave instead of sine wave. In such cases non-linear amplitude limiting components have to be included (parts with curved characteristics, such as diodes, to not cut the signal sharply). And/or filters.
- The total phase of the loop should be $\varphi(A \cdot B)=0 + n \cdot 2\pi$. It is also important how the phase changes in vicinity of f_0.

Tips for simulation:

These circuits can be hard to simulate in software. Normally noise is turned off in simulating software, but many can enable such a function. Also, often they start simulation by finding a DC operating point, which doesn't allow the L and C parts to produce a transient upon charging up. Then the simulated circuit will probably not start. Look for options such as "zero initial condition", turn off "find DC operating point" or similar to force the circuit to start without calculating DC operating point. This often helps. Also it is possible usually to setup an AC generator to create simulated noise or an impulse (an impulse will contain a very wide spectrum if it is thin enough).

The simulation software usually call time simulation "transient analysis" (even if analysing periodic signals). This is usually done by stepping time in very small discrete steps and calculating changes from previous state. This results in certain limitations.
If the time step is greater than the time a transient effect has or the period of a signal, then you would see something totally different (eg. a DC instead of AC or a different AC).
This time step unfortunately can not always be directly changed. The time duration of the total transient analysis, however, seems to influence it. Also, do set the total time to be larger than the effect you want to see. If you have a periodic signal of 1kHz then setting transient analysis time to 1us (often a default) will give no meaningful results.

Laboratory exercises

1. TT filter

1.1

Calculate f_0 using R=20k and C=10nF. The formula is the same as for the standard RC-filter:

$$f_0 = \frac{1}{2\pi RC}$$

1.2

Measure and plott the Bode amplitude plot (v_{out}/v_{in} versus f) of the TT circuit (without the amp, only the RC circuit). No power supply is needed for this. Remove all jumpers from the circuit as well. Connect the input signal between the GND and one of the pins named T and measure output between GND and the other pin named T.

Use cc. 10Vpp sinewave as input. Measure V_{in} and V_{out} at all frequencies in the table below. (With a good function generator you can quickly check that its output amplitude is constant in all frequencies and therefore you need to measure it only once.) Don't forget to graph the results in the lab report.

You can measure either peak to peak or RMS values (but make sure to use the same for all measurements). It is recommended to use the voltmeter in AC mode (measuring RMS), as it is easier than constantly having to adjust the oscilloscope.

Find the real f_0 by finding the frequency where the output is minimal. Write this frequency into the f_0 cell in the table (and measure the output here as well). In the lab report, compare the measured and calculated f_0 values.

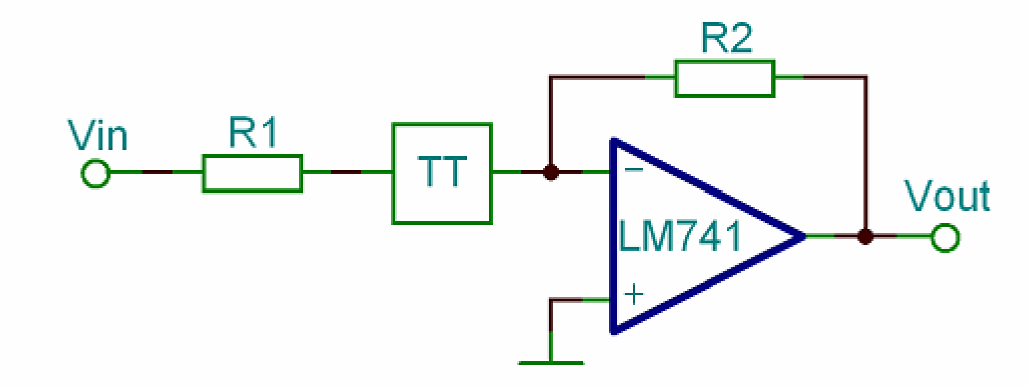
f (Hz)	50	100	200	300	400	500	600	700	750	770	f0
V_{in} (V)											
V_{out} (V											

f (Hz)	820	850	900	1k	1.1k	1.2k	2k	3k	4k	10k
V_{in} (V)										
V_{out} (V										

Find the 3dB points by taking the v_{out} at f_0 as 0dB (and finding the frequencies at which the output is 1.4 times of the output at f_0). (The output at f_0 is very small and therefore has a relatively high level of noise, try to average it out.)
Find the quality factor Q:

$$Q = \frac{f_0}{B} = \frac{f_0}{f_{upper} - f_{lower}}$$

2. Active band stop filter



7. Active band stop filter

Connect jumpers 2, 4, 5, 8 to create the active band stop filter circuit.
Connect ±15V power supply (+15V to pin "+Ut", −15V to pin "−Ut" and ground to pin GND). Setup 20mA current limit on both sides.

The Bode plot of this circuit should look similar to figure 3. The differences will come from the frequency dependency of the opamp itself (which has a low-pass filter characteristic).

The gain is (as an inverting amplifier)

$$A_V = -\frac{Z_2}{Z_1}$$

where Z_1 is now made up of the TT and R_1, Z_2 is R_2. The TT has maximum impedance at f_0, therefore the band stop characteristic. R_1 is needed because at high frequencies the TT would have very little impedance (which loads the function generator).

2.1

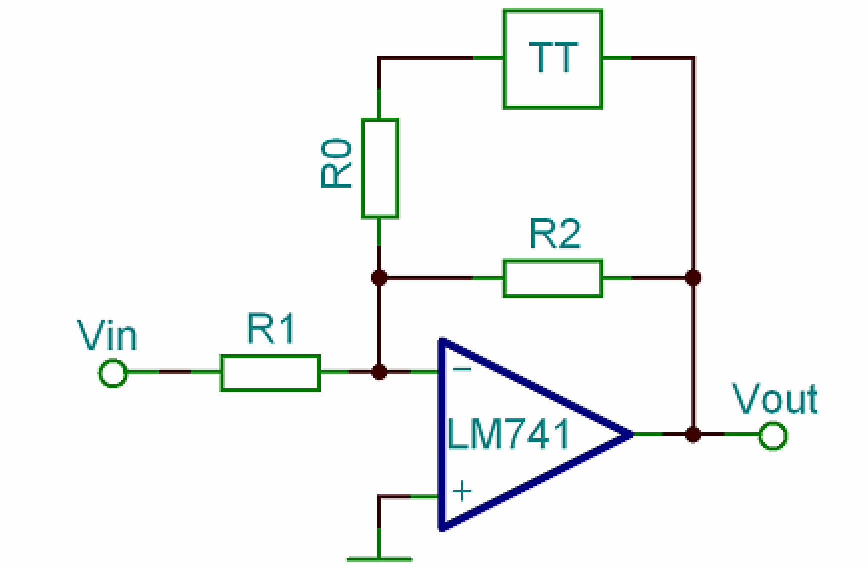
Connect the function generator to the input pin V_{in} (U_{be}). Measure output on V_{out} (U_{ki}).
First connect circa 5kHz sinewave to the input and setup the amplitude such that the output is not distorted (check on the scope).

Measure v_{out}/v_{in} plot. Use the same frequencies as in exercise 1.2.

Find the 3dB points and Q as well.

3. Active band pass filter

Use jumpers 1, 3, 5, 8 to create the active band pass filter. Keep the power supply and input-output connections from the previous circuit.



8. Active band pass filter

R_1=4.7kΩ, R_2=75kΩ, R_0=10kΩ

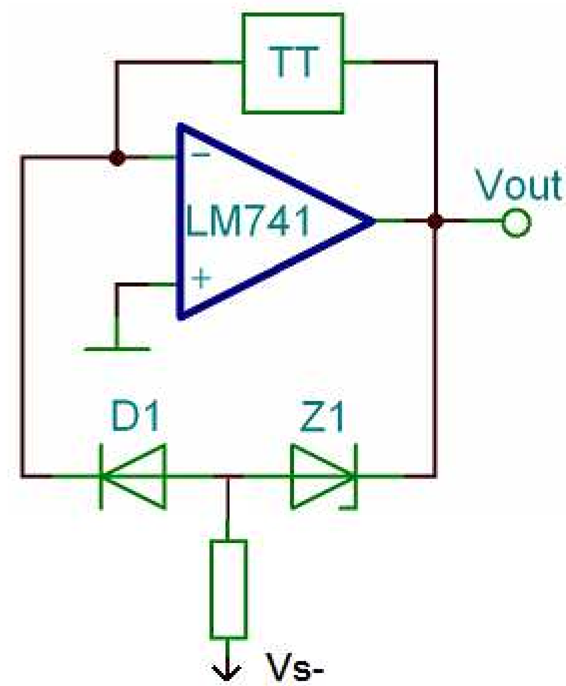
Here the TT filter is in the feedback, which is similar to parallel connecting it, therefore the circuit will show a band pass characteristic (as now Z_2 is made up of the TT and R_0 and R_2 in the gain formula). R_2 is needed because around f_0 the TT has too high impedance and thus the gain would be too high, overdriving the amplifier.

3.1

Use cc. 0.5Vpp sinewave as input. (Check again so the output is not distorted.)
Measure v_{out}/v_{in} plot again using the same table.

Find the -3dB points (use the value at f_0 as 0dB).

4. Oscillator with TT



9. Oscillator with TT and with amplitude limiting diodes

Connect jumpers 3, 5, 7 to create the TT oscillator circuit. This circuit needs the same power supply but does not need an input!

The amplitude requirement is hard to meet exactly, therefore nonlinear elements (here the diodes) are often used to limit the amplitude (without it becoming squarewave-like). The output voltage peak should therefore be approximately the sum of the forward and Zener voltages.

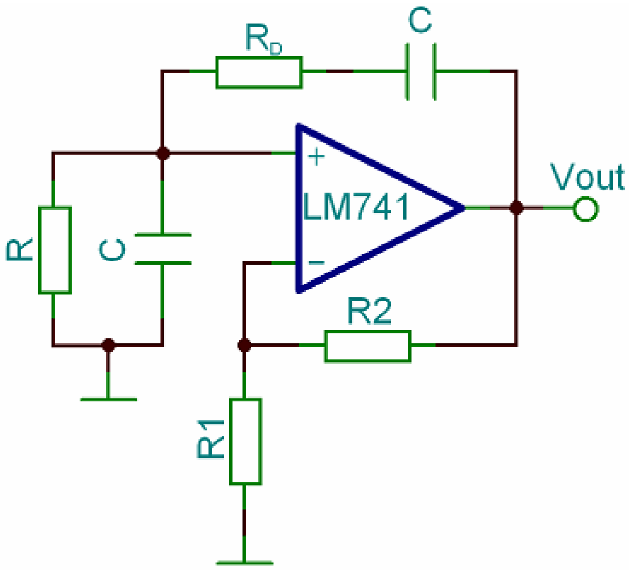
4.1

Check the output waveform. Measure the output frequency and compare with calculated f_0.

Measure the peak values of the output. Compare it with the calculated value.

Remove jumper 7. This removes the diodes from the circuit. Examine the output waveform.

5. Wien-bridge oscillator



10. Wien-bridge oscillator

R=7.15kΩ, C=22nF

This oscillator uses the Wien-bridge instead of the TT filter. The Wien-bridge has zero phase at f_0 and therefore is connected to the non-inverting input. The amplitude gain of the bridge is 1/3 at f_0 and therefore a gain of 3 is created on the amplifier to compensate and fulfill the oscillation requirement:

$$\frac{R_1 + R_2}{R_1} = 3$$

Amplitude limiting is not used here.

The resonance frequency formula is still the same:

$$f_0 = \frac{1}{2\pi RC}$$

(But the R and C values are different.)

Connect jumpers 6 and 9 to create this circuit and connect the double power supply.
(Remember, input is not needed for oscillators.)

5.1
First short-circuit the pins named RD and check the output waveform.

5.2
Then connect a decade resistor (ask your teacher for it) between points named RD. The decade resistor acts like a variable resistor, you can set the value numerically by switches (the sum of the values of the switched on resistors gives the total resistance).

Start turning on resistors on the decade resistor until the output signal is closest to a sine wave (increase until the output disappears and then go back). Write down the RD value here. (It should theoretically be 7.15k, equal to the other R – more correctly, RC products for the two parts of the bridge should be equal).

Measure the output signals frequency and amplitude. Compare with calculated values.

6. Entry test questions

1. Draw the TT filter schematic. Write the resonance frequency formula.
2. Draw the Bode plot of an ideal TT filter.
3. Write the formula for quality factor and name the components.
4. Draw an active band stop filter circuit (use block for TT).
5. Draw a Wien-bridge and give the resonance frequency formula.
6. What is an oscillator? What are the oscillation criteria?