Muğla Sıtkı Koçman University

EEE-2004 – Circuit Analysis II

DESIGN PROJECT

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Cascaded Active Low-Pass Filter as Double Integrator

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Introduction

Filters

Filters are a type of signal processing electrical circuits that remove, i.e filter out unwanted frequency from an applied signal and/or enhance wanted ones.

Based on the frequency range being discarded filters are classified as high-pass, low-pass, band-pass, band-stop (notch) or all-pass. Where, for instance, a high-pass filter allows frequencies higher than a certain point, but rejects those lower.

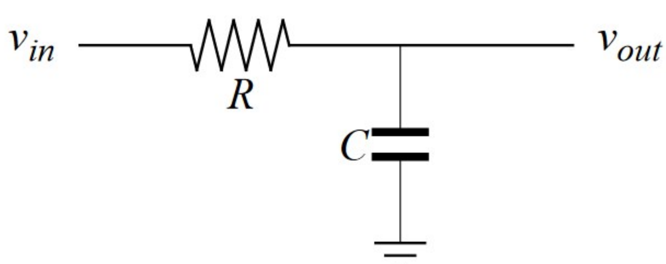
Filters can be passive or active, where passive components like resistors and capacitors etc, form the basis of the former one, while active components like transistors and op-amps make up the latter one. Active filters have the ability to amplify signals and also have the added advantage of being easily adjusted. In particular, op-amps making up filters, are characterized by being modified via external components, such as resistors connected to it.

Active Low-Pass Filter

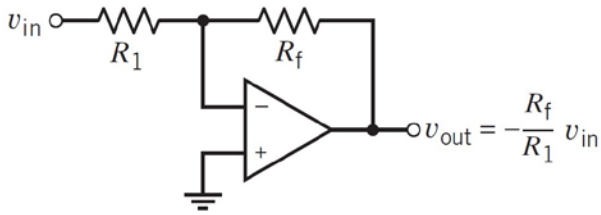
The basic passive low pass filter consists of a resistor and a capacitor, where the output signal is received across the ends of the capacitor. Since,

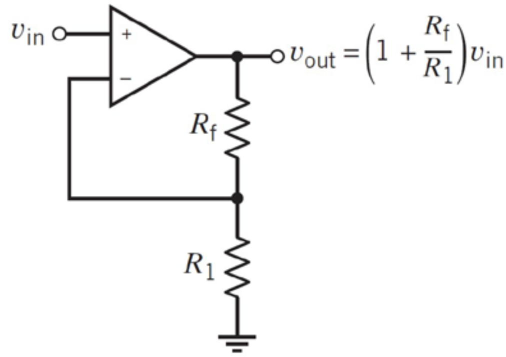
$$X_C = \frac{1}{2\pi f C}$$

The capacitor's impedance decreases with increasing frequency. This low impedance in parallel with the load resistance tends to short out high-frequency signals, dropping most of the voltage across series resistor.



An active low pass filter adds an active component, the op-amp, in the circuit and enables amplification and gain control. The passive filter can either be connected to the inverting or noninverting op-amp amplifier.

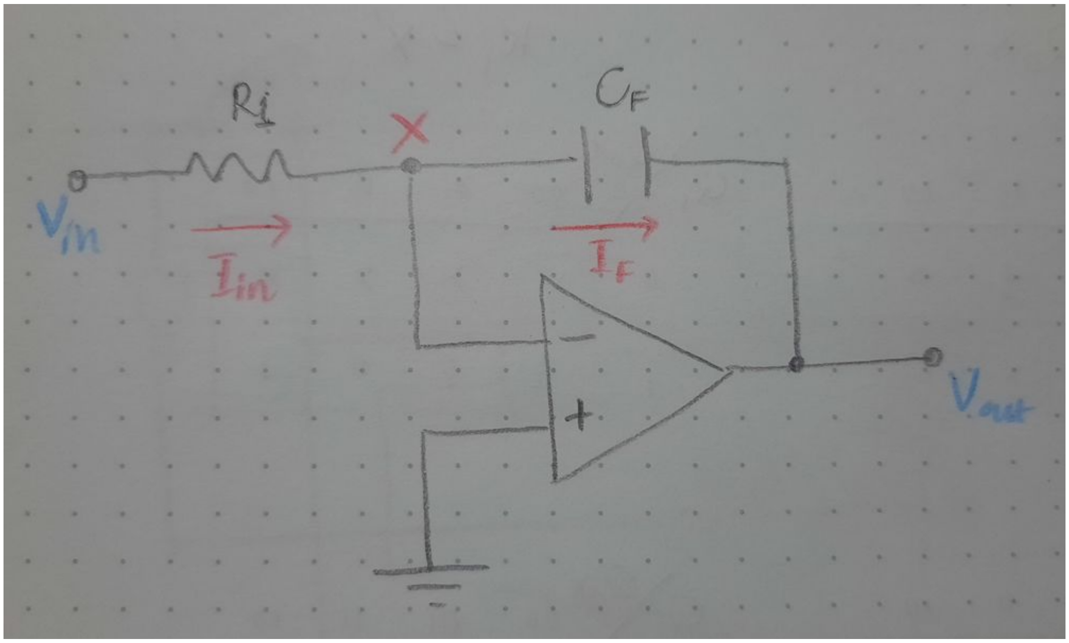




If we take an inverting op-amp amplifier, and add a capacitor to its feedback circuit, we achieve an active low-pass filter. At low frequencies the capacitors reactance X_C is much higher than the feedback resistor R_F, being in parallel, it can be neglected, allowing output gain be the standard inverting formula, $-\frac{R_F}{R_1}$. But at increasing frequencies, where X_C diminishes, reducing the impedance of the parallel combination $X_C||R_F$. The circuit impedance then, is only determined by R_1, across which all the voltage drop occurs, thereby providing no output.

As an Integrator

Let us examine the active low-pass filter.



Node X is at virtual ground, so $V_X = 0$. Current through resistor is I_{in}.

$$I_{in} = \frac{V_{in} - V_X}{R_1} = \frac{V_{in}}{R_1}$$

While current through capacitor is I_F:

$$I_F = C_F \frac{d(V_X - V_{out})}{dt} = -C_F \frac{dV_{out}}{dt}$$

Assuming ideal op-amp:

$$I_F = I_{in}$$

$$-C_F \frac{dV_{out}}{dt} = \frac{V_{in}}{R_1}$$

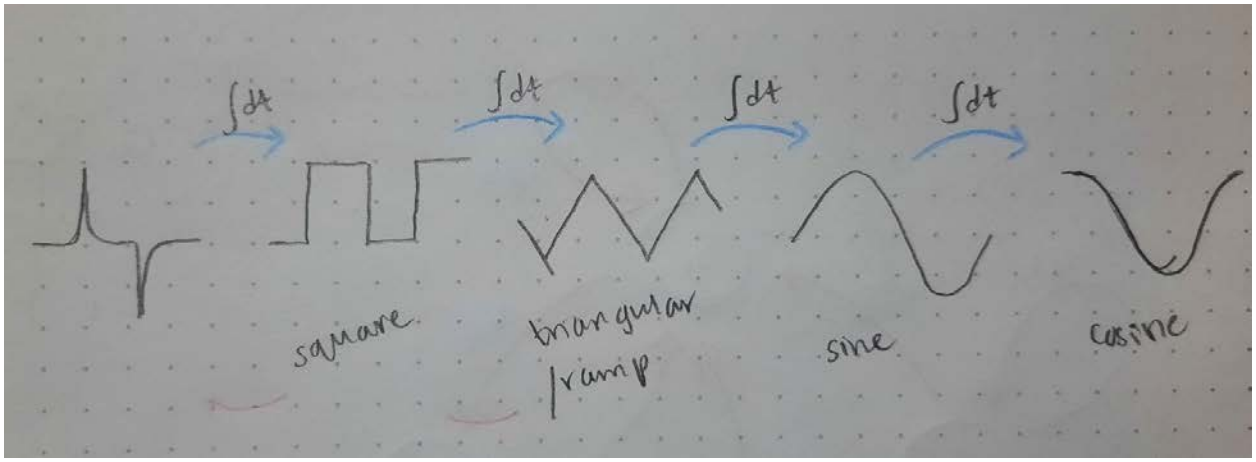
$$-C_F \int \frac{dV_{out}}{dt} dt = \int \frac{V_{in}}{R_1} dt$$

$$V_{out} = -R_1 C_F \int V_{in} dt$$

Hence the signal appearing at the output of the circuit is the integral of the input signal. Or in other words, the low-pass filter acts out the mathematical operation of integration to the applied signal. Where the gain factor is $R_1 C_F = \tau$ the time constant of the RC arrangement.

As Waveform Generator

Now that it can perform integration, an interchange of some common signal waveforms is possible. For example, a square wave into a triangular wave, and this into a sine wave.

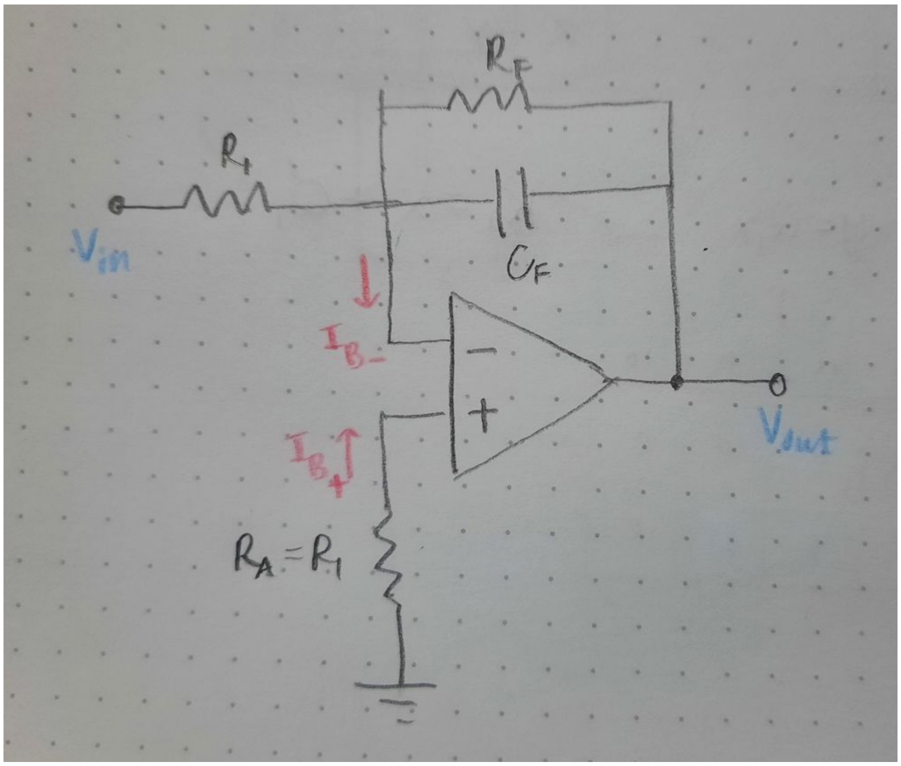


Design

Practical Integrator

In deducing the op-amp integrator formula, we assumed an ideal op-amp. But in a realistic case, there exists input bias currents I_{B-} and I_{B+}. And input offset voltage V_{OS}. These result in the output voltage to drift over time, which evidently interferes with the operation of integration, and as it accumulates the integrator does not remain reliable.

A matching R_A resistor to R_1, effectively manages the effect of the input bias currents. While the introduction of a feedback resistor R_F of large value, limits the ideally infinite DC gain of the op-amp to a finite value. Thus a more practical integrator would be:



Frequency Analysis

Since it is still a low-pass filter, it responds to the effects of varying frequencies, where the gain diminishes at higher frequencies. It is given by:

$$A_V = -\frac{R_F}{R_1}\frac{1}{1 + 2\pi f C_F R_F}$$

The cutoff frequency appears at

$$f_o = \frac{1}{2\pi C_F R_F}$$

Where $f_1 = \frac{1}{C_F R_F}$ represent the lower and $f_2 = \frac{1}{C_F R_1}$ the higher end of the cutoff.

Design

With these in mind, we can design the integrator to operate at designated frequency ranges and adjust its gain.

Let us suppose we have the integrator with values, $C_F = 1\mu F$, $R_F = 10k\Omega$ and we want it to have a gain of $A_V = 1.35$ at $f = 100Hz$, then the resistor R_1 we require would be:

$$A_V = -\frac{R_F}{R_1}\frac{1}{1 + 2\pi f C_F R_F}$$

$$R_1 \cong 1k\Omega$$

As such the frequency at which the gain would drop to 1. i.e. reduce to 0dB would then be:

$$f \cong 150Hz$$

And beyond this point it would start diminishing output signal.

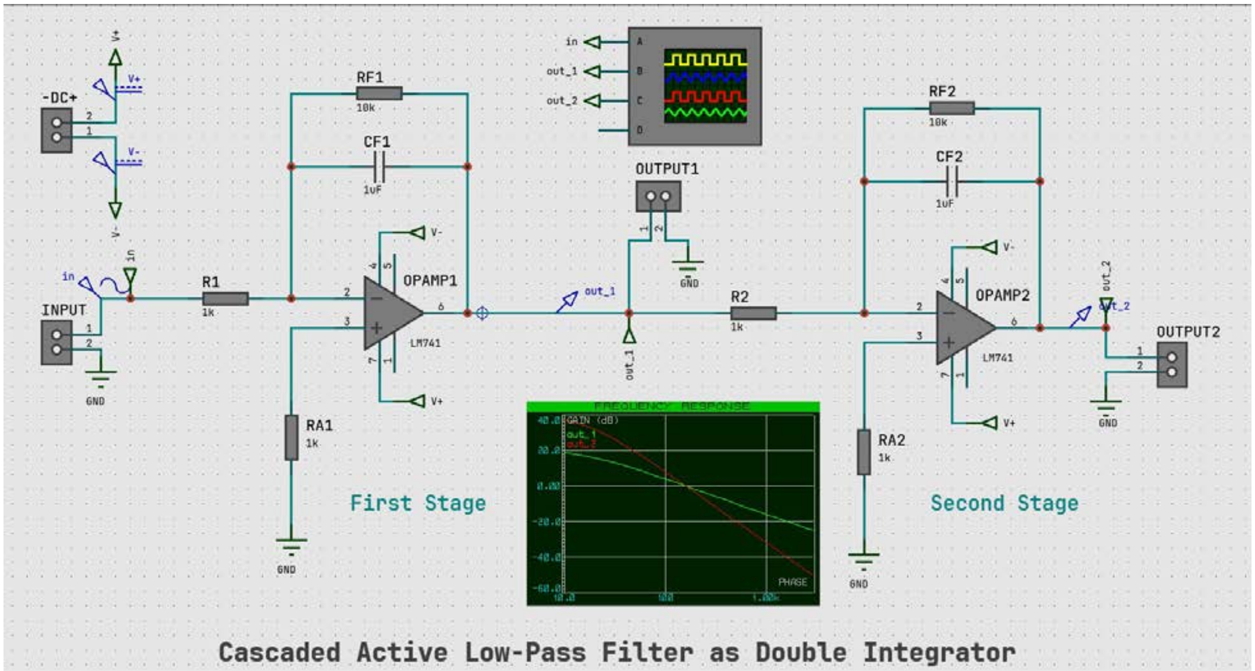
Cascaded Integrators

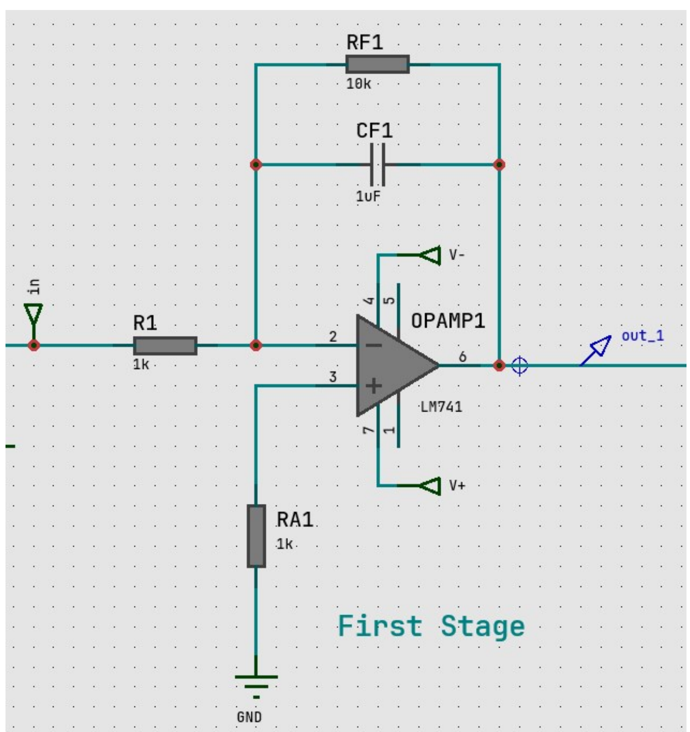
Since the inverting op-amp has high input impedance, a second integrator could be cascaded to the first one, to provide 2nd order integration. The frequency response would remain the same, but the gain will be multiplied.

Simulation

Simulation

Given below is the circuit schematic in the simulation software, complete with connections ready for simulation and also for PCB design

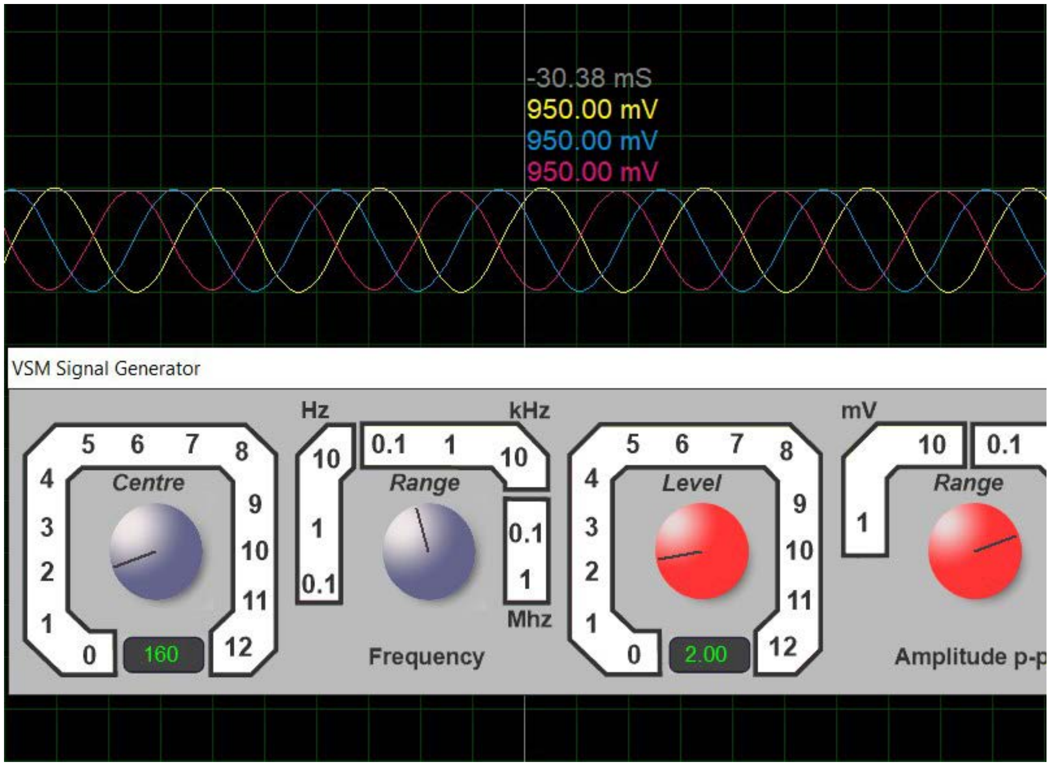




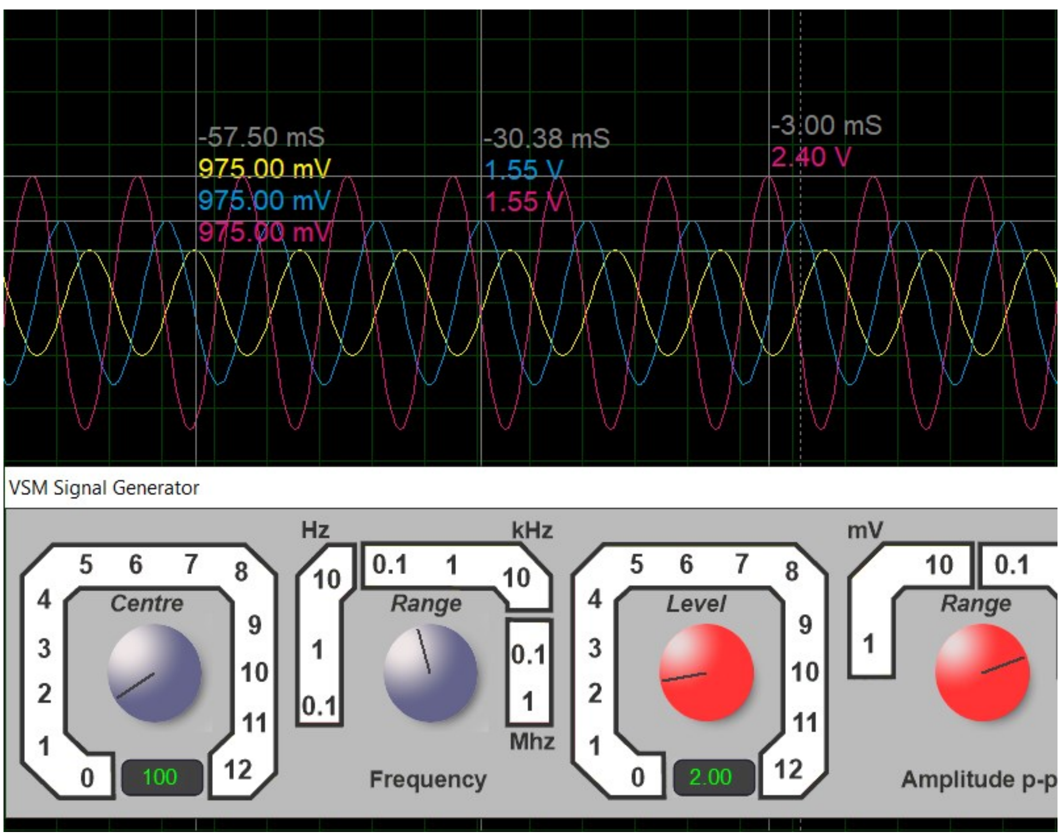
Result

First of all the bode plot confirms our designed values. With the determined resistor values, the gain dimishes to 0dB at $f \cong 150\text{Hz}$. Here, out_1 and out_2 are the output of the first and second integrator respectively.

For another confirmation we see that with an application of input voltage of $2\ V_{PP}$, at this frequency we observe no gain. Here $V_{in} = 1V$, $V_{out1} = 1V$ and $V_{out2} = 1V$.

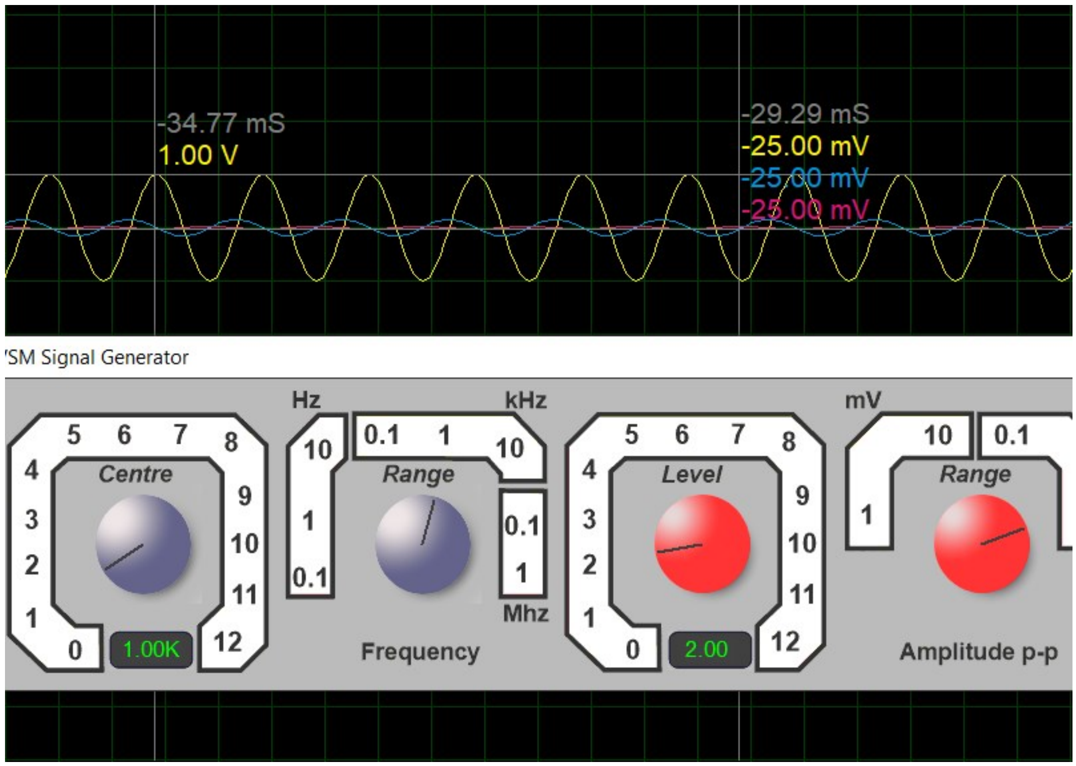


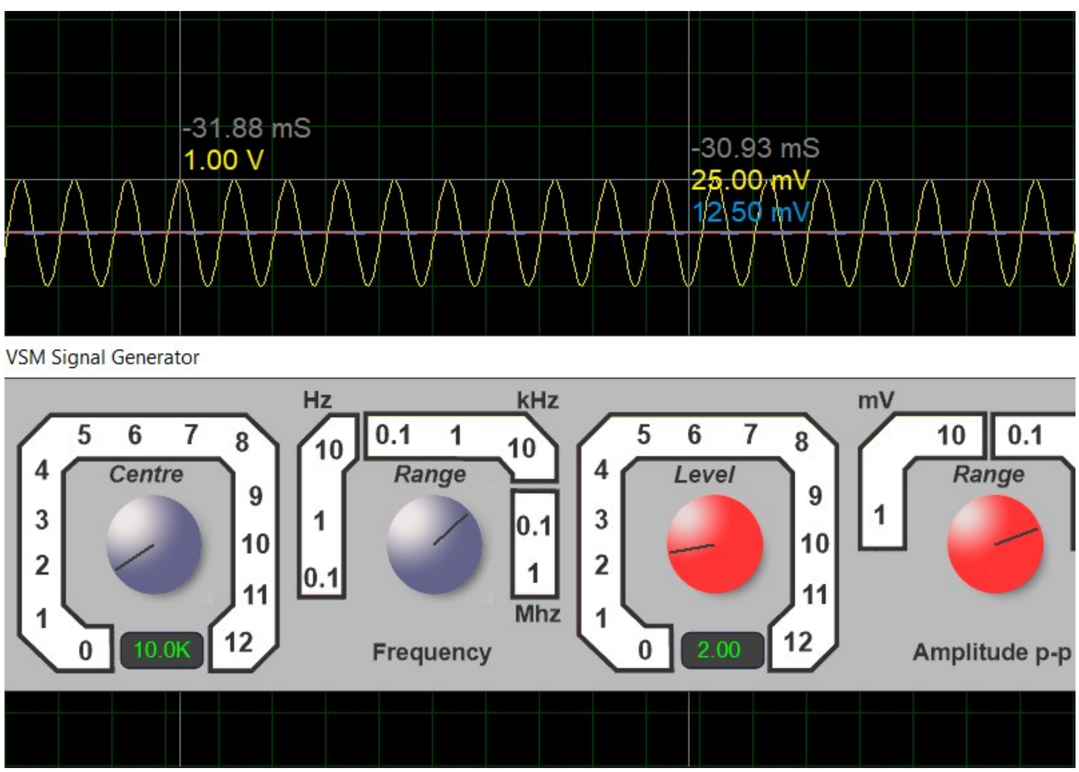
But at 100 Hz, we had designed a gain of 1.35. the simulation shows similar result.



But the gain of 2nd output is greater because its gain is multiplied with that of the previous output.

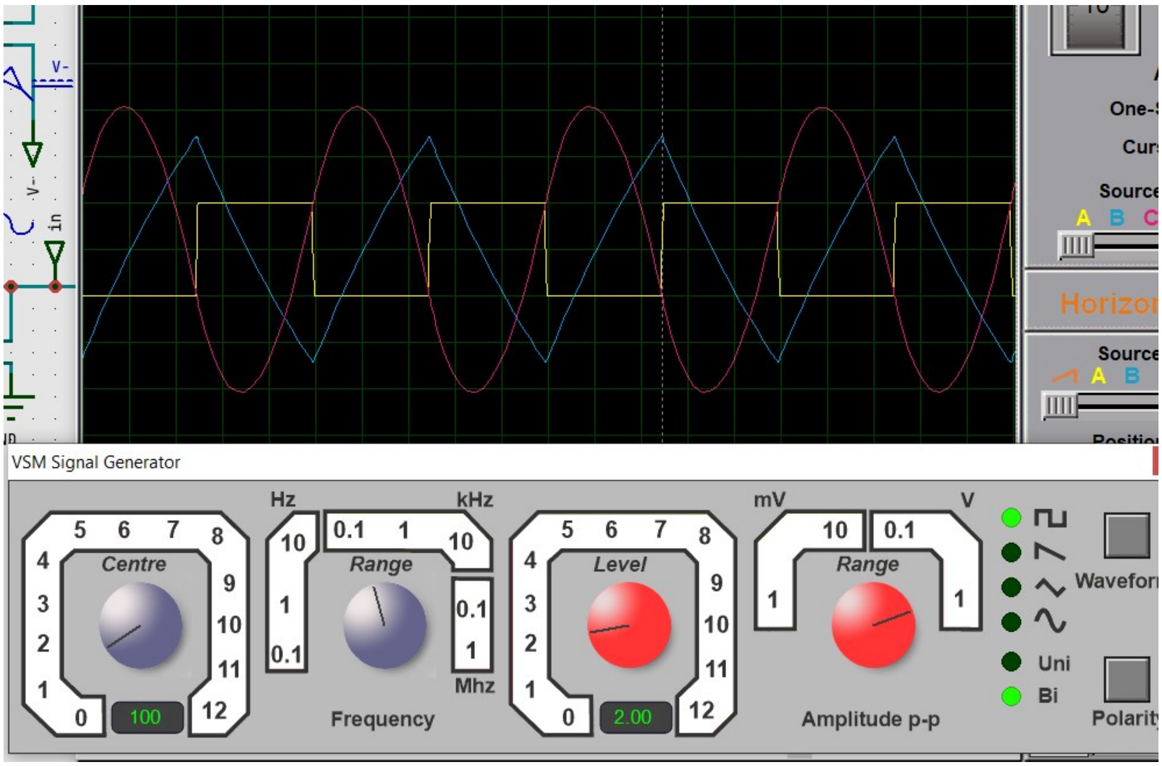
At higher frequencies, the gain diminishes and the circuit behaves as a low-pass filter:



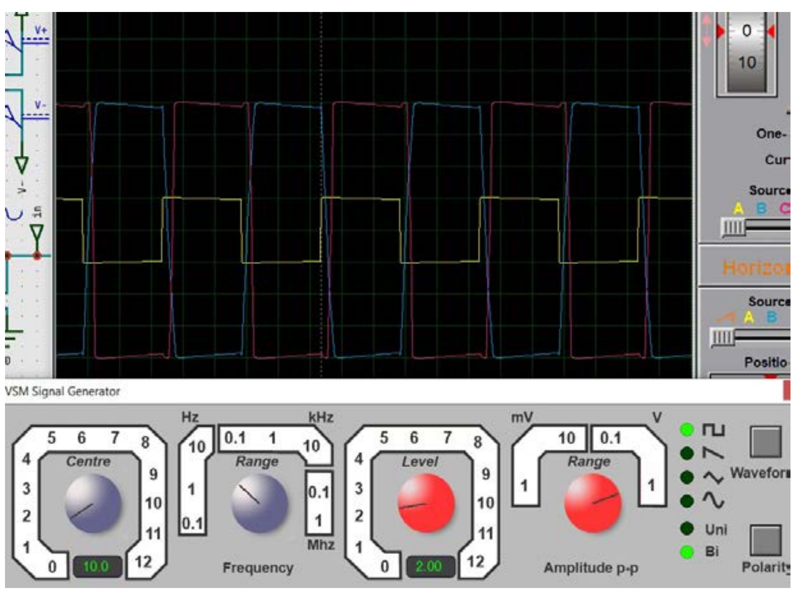


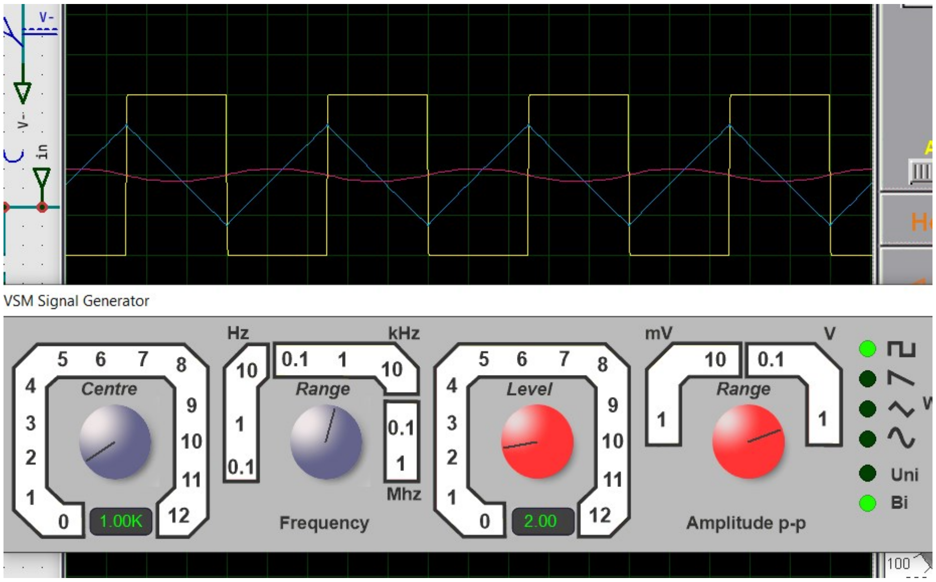
Its effect is an integrator does not yet seem obvious, even though that is being done as well. The fact of the matter is that for sinusoidal inputs, the outputs would be sinusoidal as well, because the integral of a sine waveform is a cosine waveform which further integrates into a sine waveform.

But toggling the input signal to another waveform, in particular a square wave, it would be first integrated into a triangular wave, which further turns into a sine wave.



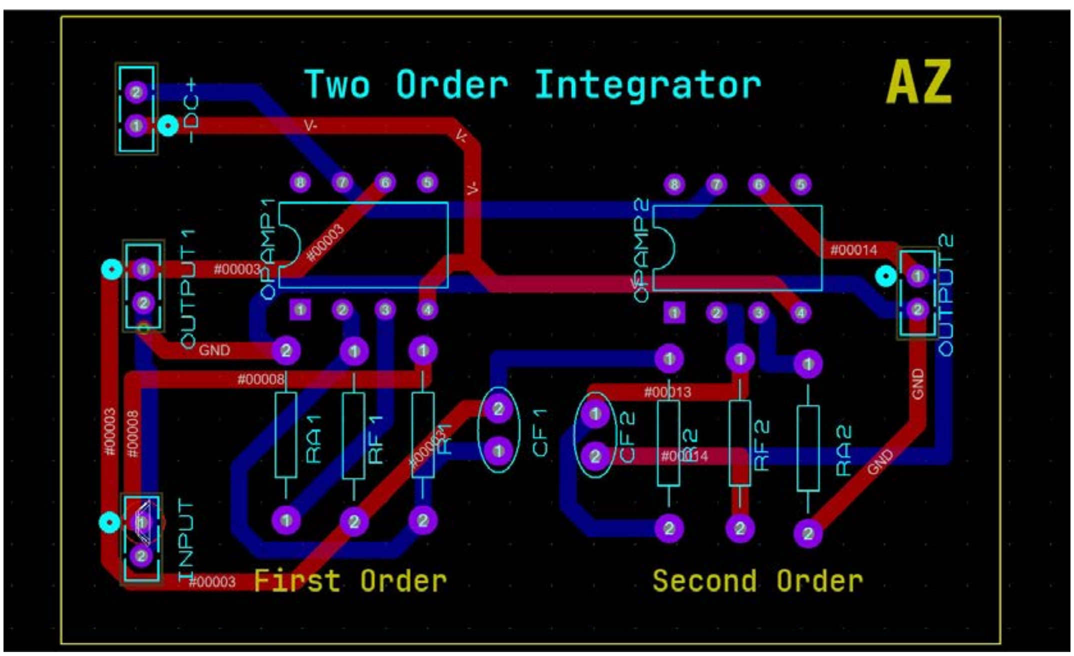
This result though is not too apparent at lower or higher frequencies.



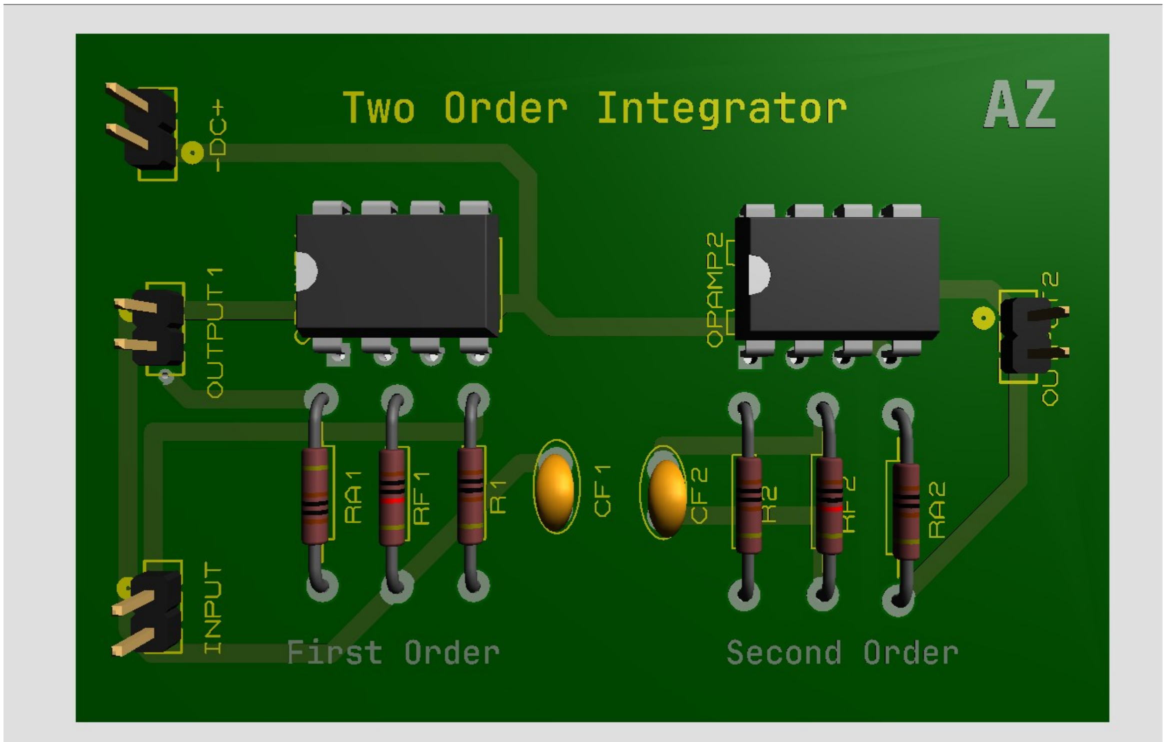


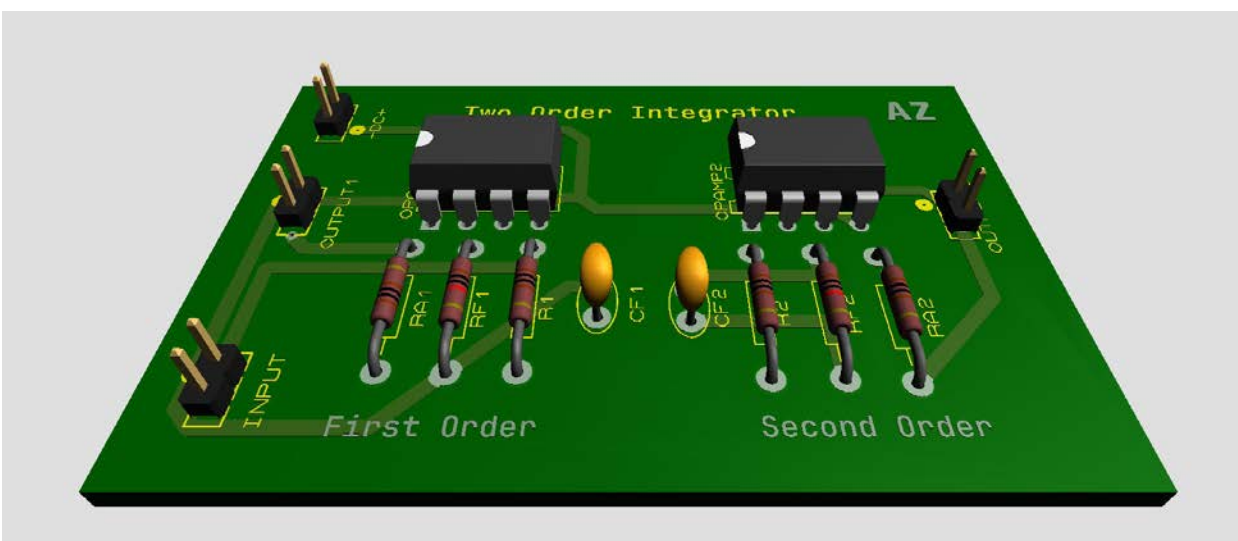
PCB

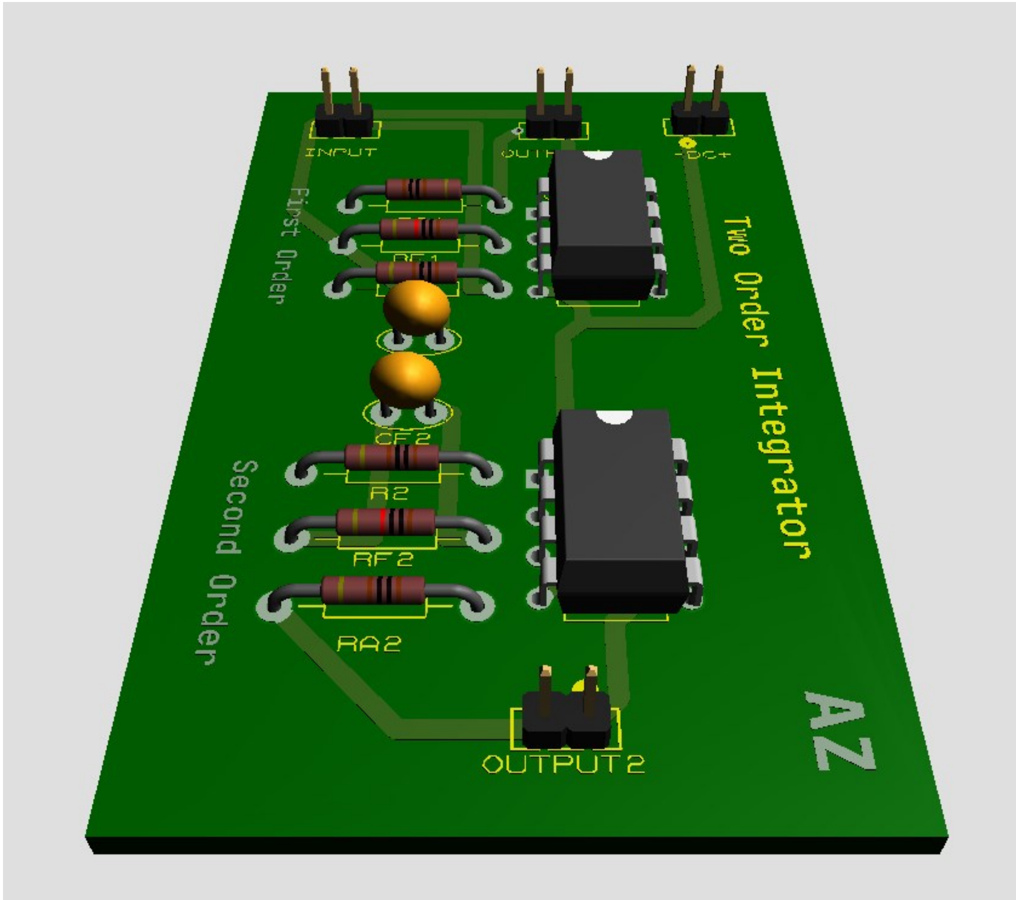
This is the blueprint for the circuit PCB:

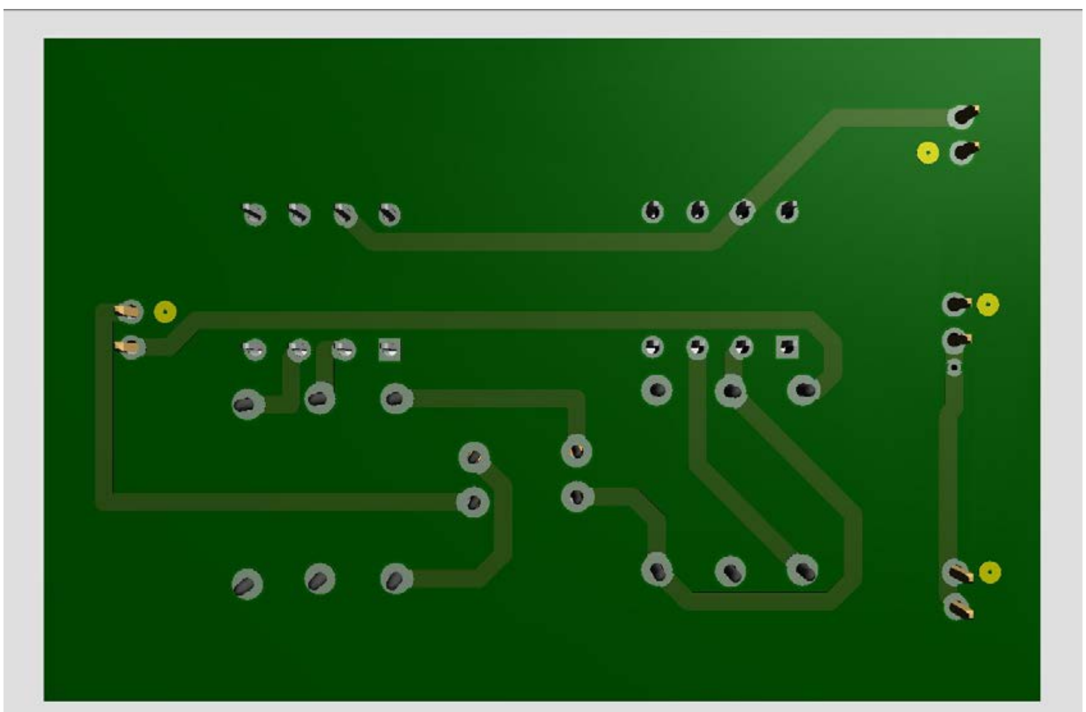


These are the 3D visualizations of the PCB, in different views:

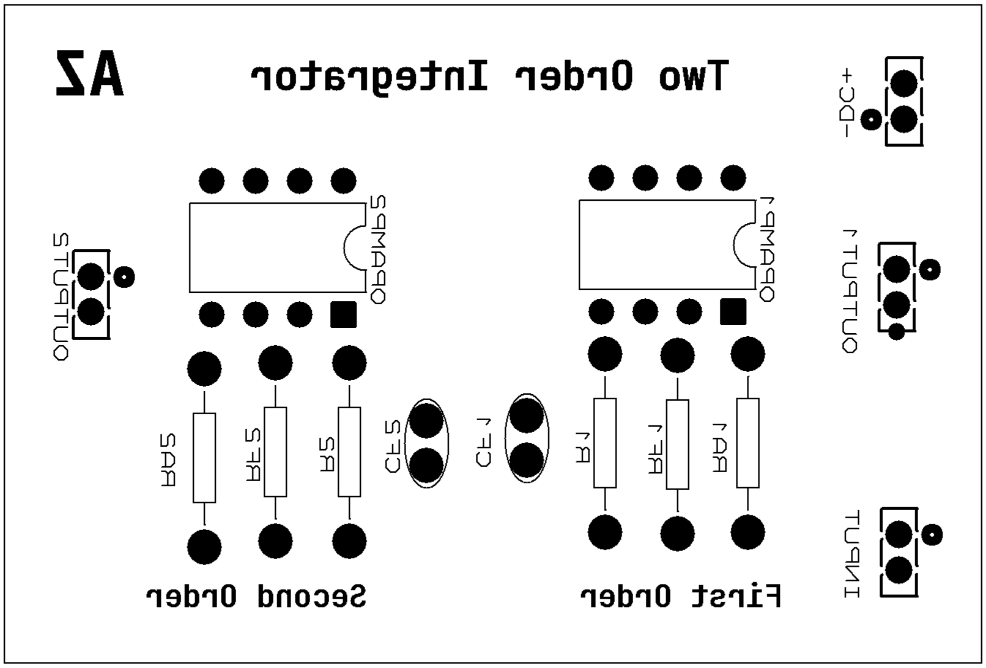


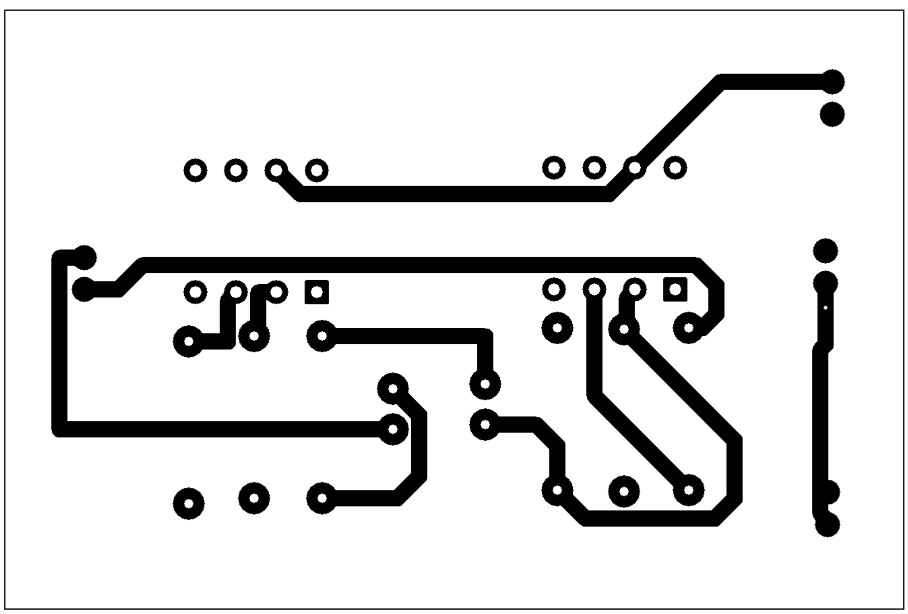


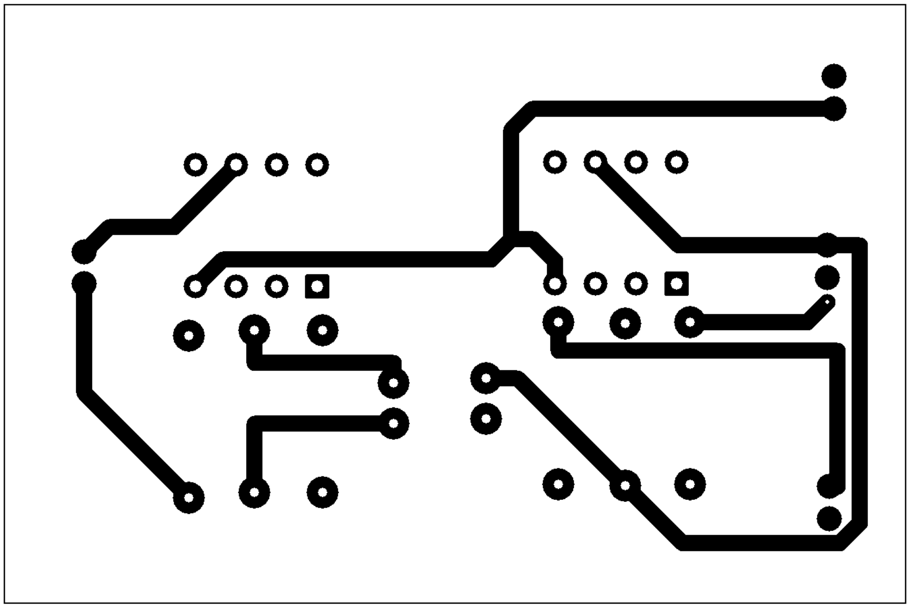

INPUT
First Order
CF2
R2
RF2
RA2
Second Order
Two Order Integrator
OUTPUT2
AZ



PCB Print-Out



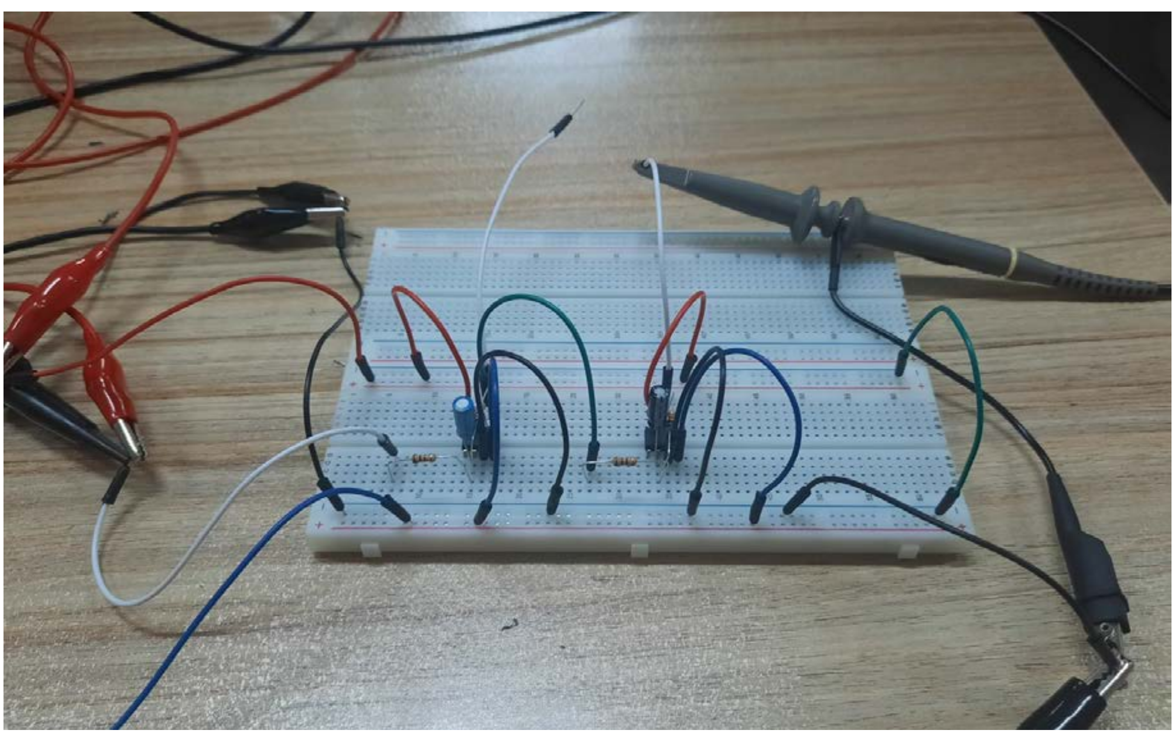




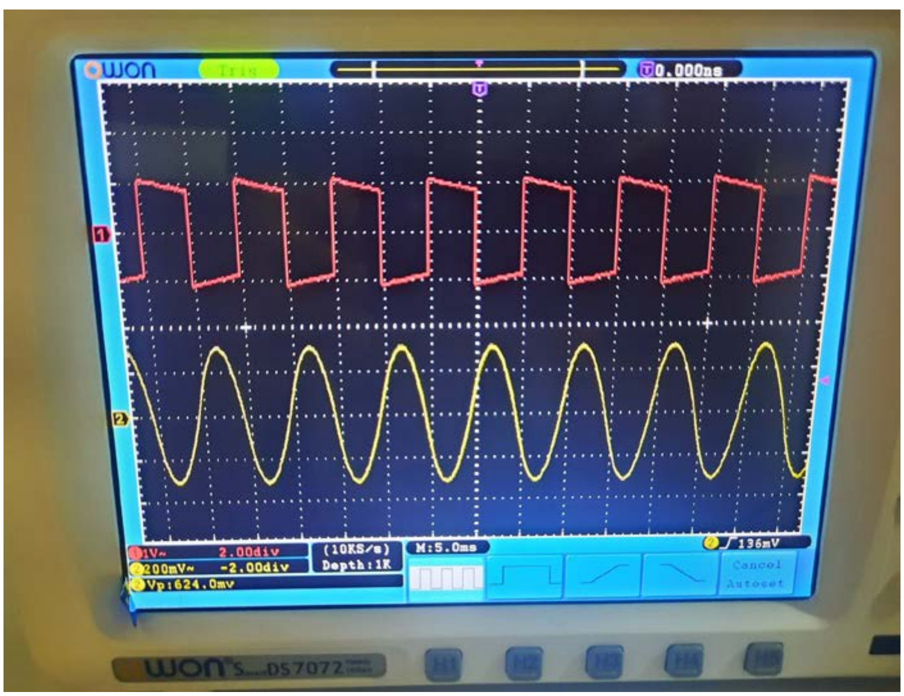
Back and front side respectively.

Experimental Results

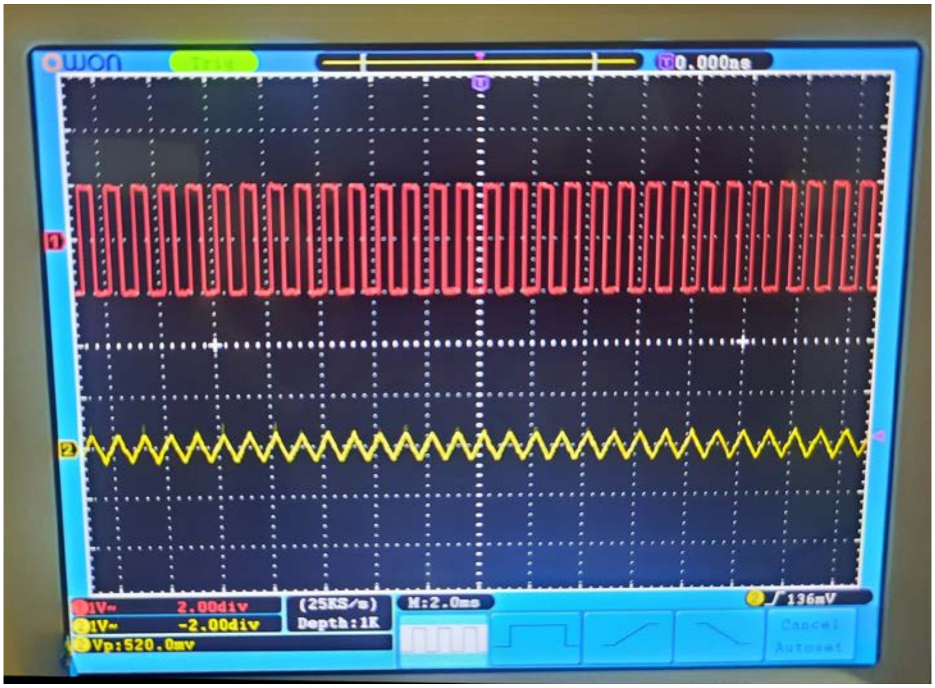
Circuit on breadboard:

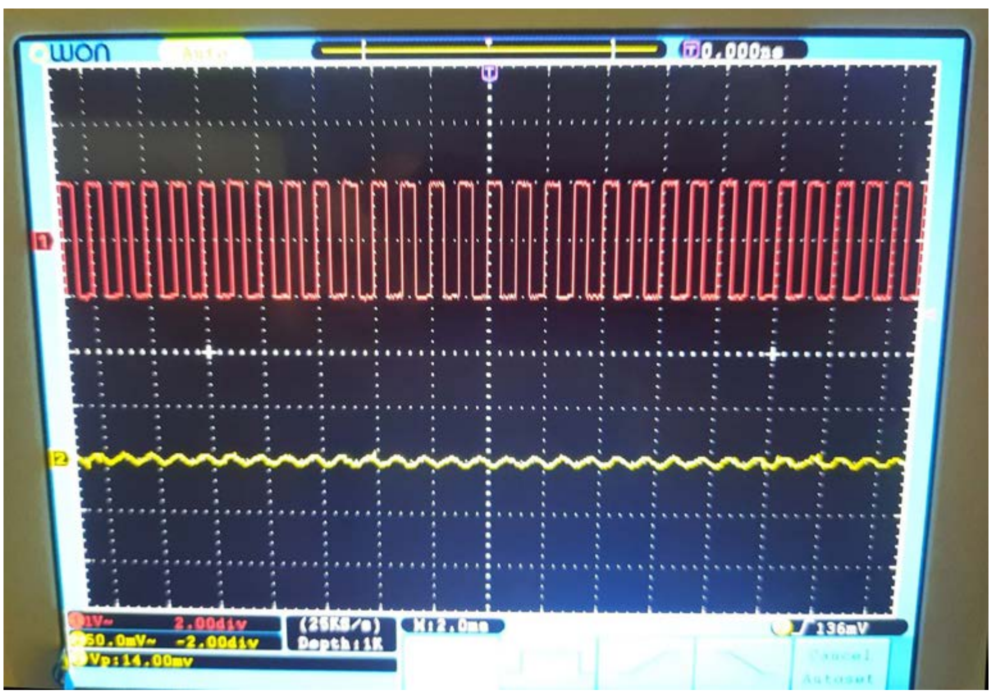


Input and Output signal on the oscilloscope show its operation as an integrator. The input signal fed is a square wave, while the first output is triangular, and the second one is sine.



Notice at higher frequencies, the output diminishes, thus showing its effect as a low-pass filter.





All files available:

https://github.com/az-yugen/EEE-2002-4-6.-LAB