Operational amplifier examples of application

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1. Introduction

In the exercise the second order filters (low-pass, band-pass and high-pass)) and the rectangular waveform generator are investigated. All circuits are composed of operational amplifiers. Filters and generator are implemented inside the Field Programmable Analog Array (FPAA) chip from Motorola. FPAA chips are used for rapid implementation of basic analog projects. Motorola chip was built using the switched capacitor (SC) technique. The SC technique is used to replace the resistors by circuits consisting only of MOSFET switches and capacitors. The switched capacitor circuits and operational amplifiers allow implementing analog signal processing (e.g. amplification, filtration) in standard CMOS technologies. The Motorola FPAA chip is programmed by 6864 electronic switches trough a software EasyAnalog. The EasyAnalog window displays a schematic of the chip with subcircuits (gain stages, filters, rectifiers, oscillators, integrators, etc.) displayed as icons. The user is allowed to select and place the subcircuit icons and connect them together interactively.

2. FPAA short description

Figure 1 shows a block diagram of the Motorola MPAA020 FPAA chip. The device contains an array of 20 identical CMOS analog core cells arranged in a 5 x 4 matrix which can be programmed individually or in combination to perform a variety of analog functions. Each cell (Fig. 4) contains an operational amplifier, a comparator, 5 capacitors (each with an 8-bit programmable value) and a number of routing switches. Each core cell is capable of a number of primitive functions such as gain, rectification and first order filtering. Core cells may be combined to create higher level functions such as cascaded biquad filters, data converters and control functions. Local interconnect is provided to allow a core cell to directly communicate with the cells adjacent to it. The majority of signals are usually routed this way to reduce cross talk. Global interconnect runs as pairs of wires throughout the whole array, both vertically and horizontally between each cell. Each cross-over of the global interconnect has a set of switches, to allow signal to move from horizontal to vertical global tracks.
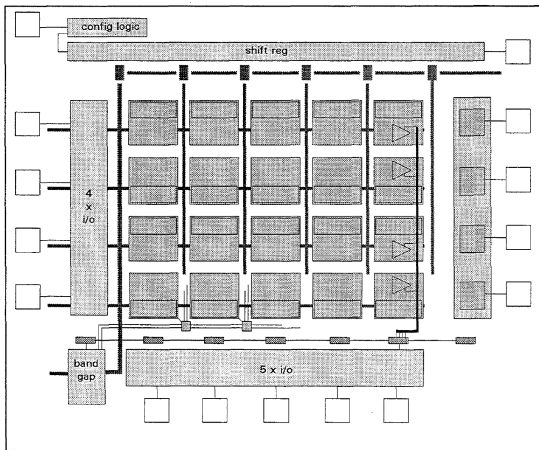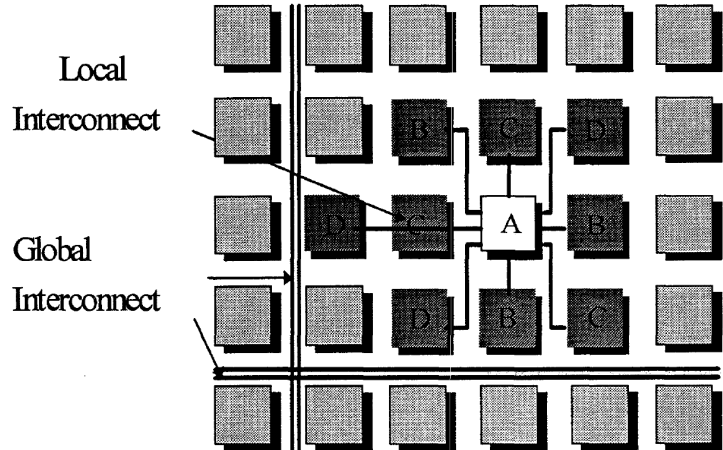

Fig. 1. Array Overview

Fig. 2. Interconnect patterns.

Around the periphery of the array there are 13 I/O cells which provide buffering. These are bi-directional analogue cells (Fig. 3), and may be used as unity gain buffers or may be bypassed to allow direct I/O.
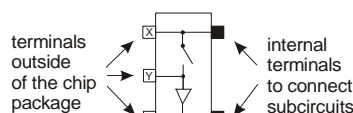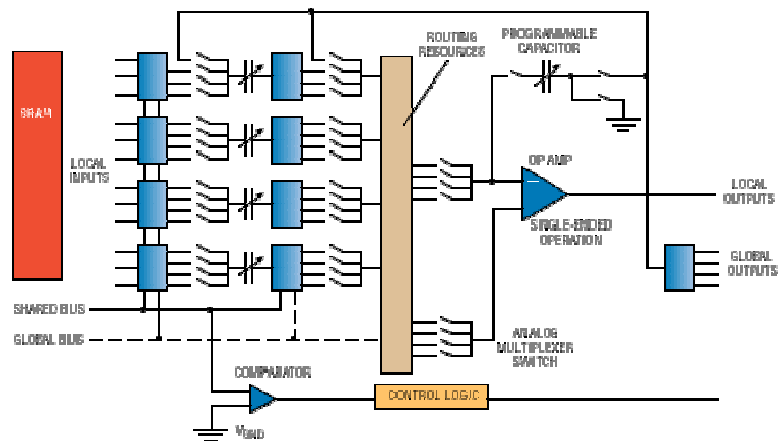

Fig.3. I/O cell

Fig.4. Switched capacitor core cell

At the heart of each cell is a switched-capacitor op amp that serves as the core programmable element. The cell also includes a comparator, capacitor arrays, switches, and SRAM (see Fig. 4). Within the cell, switches are configured to achieve the desired values of capacitance in the input and feedback paths of the op amp. Switch configurations are stored in the SRAM.

Switches programming capacitors values are static and are set once during the programming. The system also uses a dynamic switches to periodically (at a specified frequency) switch the capacitors, which makes the capacitor with the switches acts as a resistor. Connections within and between the cells are configured via an RS-232 interface and PC computer. Under typical operating conditions, the recommended maximum signal frequency is 200 kHz, with 1 MHz specified as the maximum clock rate for internal sampling. The device handles input signals from 0.5 V up to V_{DD}–0.5 V, and output loads of 100 pF maximum capacitance and minimum 1 kΩ resistance. Harmonic distortion is 0.1% at 1 kHz. The chip is supplied from a single +5V power supply. Signal processing uses the signal ground, which is generated by dividing the supply voltage. Thus the potential of signal ground relative to earth is + 2.5V. Voltages of all the signals refer to the signal ground, hence the greatest signal amplitude is 2.5V.

3. EasyAnalog software

The software EasyAnalog, takes care of all decisions made regarding capacitor values and signal flow paths. The software has a built-in library of analog subcircuit (called Macro) with information about how to switch connections in one or two adjacent core cells in order to implement the specified macro. Thus, user does not need to take care of the settings for each switch, and operates only on the symbolic blocks representing elementary circuits. The user must select the macro from the library, connect them with "wires" and set the parameters of each macro. Then the finished project needs to be transmitted from the PC via RS232 to the FPAA. Finally, the project must be tested by measuring.

The software window displays (Fig. 5) the floor plan of the FPAA, containing the array 20 free cells to be programmed. Around them (left, right, and bottom) is 13 I/O cells with buffers. Vertical and horizontal double lines arranged between the cells are wires for global interconnection between distant cells.

In order to program a new project you should:
- select and place subcircuits (macros);
- define the parameters of the subcircuits;
- connect subcircuits and I/O using wires;
- transfer the project to the FPAA chip;
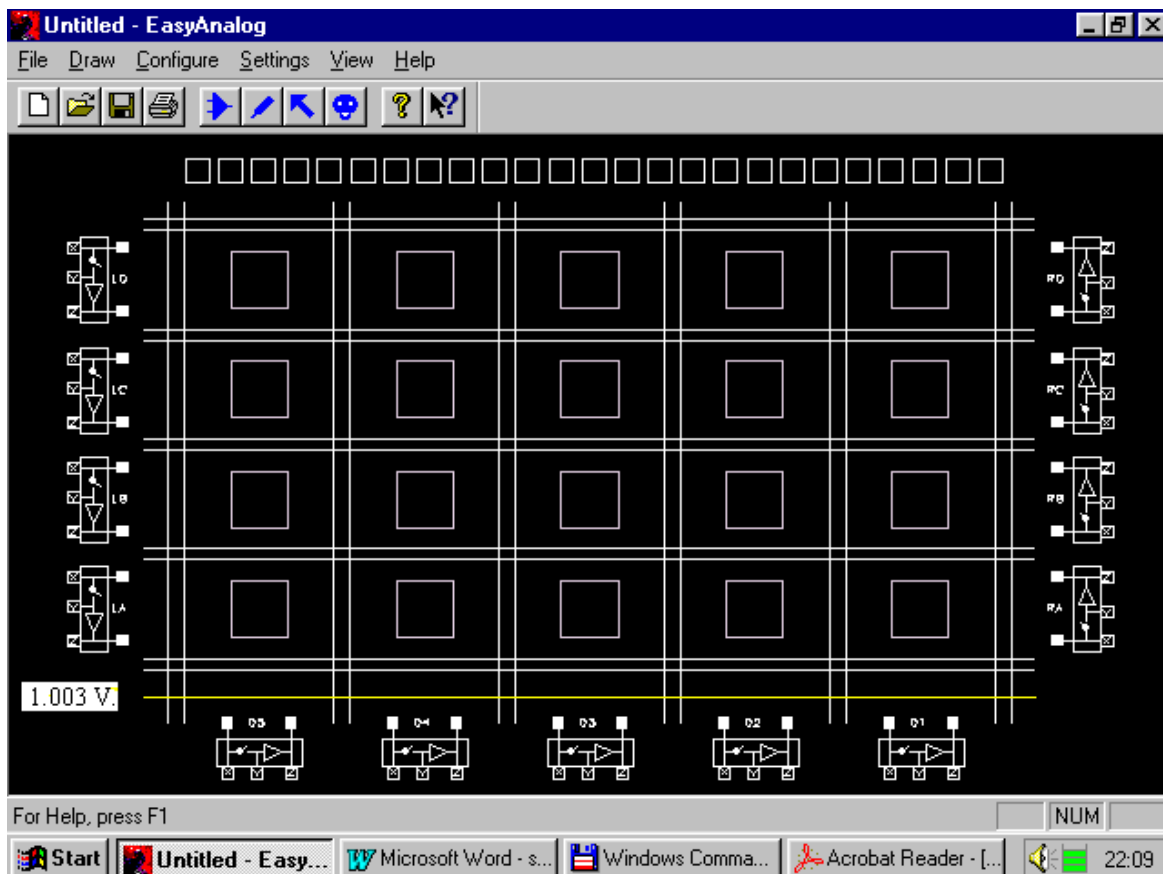- check the operation of the project.

Fig. 5. A view of "Easy_Analog" program window after turn on.

To select macro click the mouse on the symbol ![arrow], then the macro selection window will appear (Fig. 6).
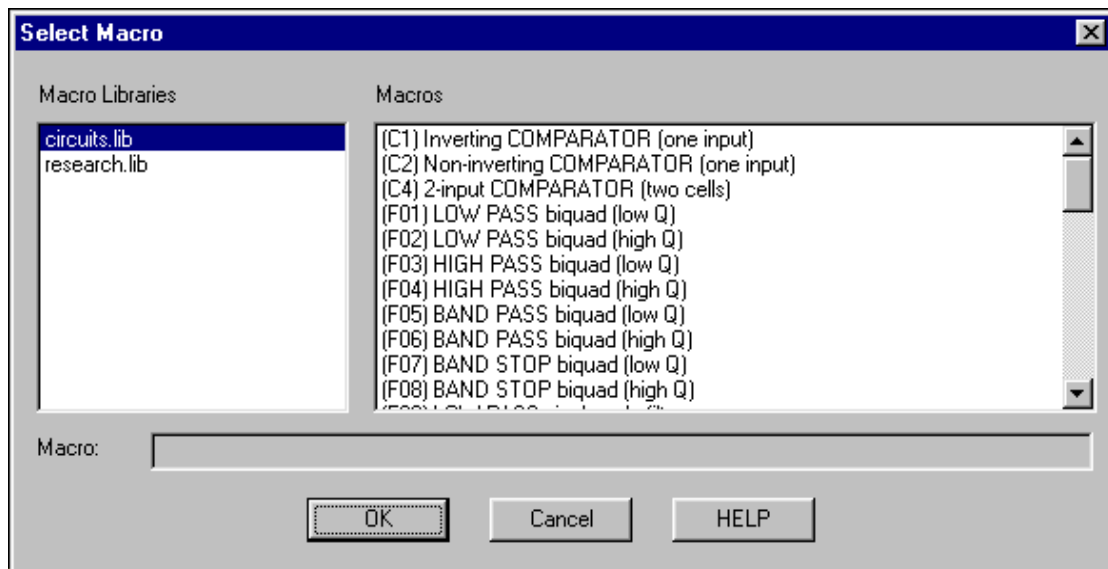
Fig. 6. Selection window of the macro from library

The selected macro you can put any number of times in FPAA, positioning the mouse pointer in the selected area and click the left mouse button. Fig. 4 shows an example of arrangement of the inverting amplifier in the lower left zone of the FPAA.
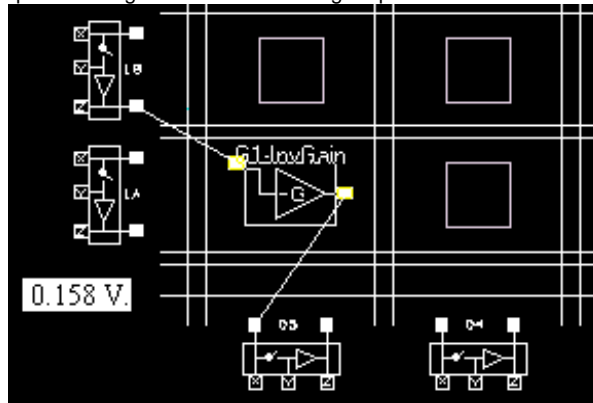

Rys. 7. Example of the subcircuit location

To enter the drawing connections mode, press the "w" key or left-click on the symbol [icon], then the cursor changes shape. By pressing and holding the left mouse button, you can connect macro pins with inputs, outputs and vertical and horizontal buses. You can also connect wires together at the point of intersection of vertical and horizontal global lines. If the connection is possible, the element changes its color to blue when it is under the cursor. Some connections can not be realized due to the lack of switches which have already been used to create a specified macro. In the drawing connections mode, you can also edit parameters of individual macro The window containing the parameters of the macro opens when you place the cursor over a selected macro and click the right mouse button.
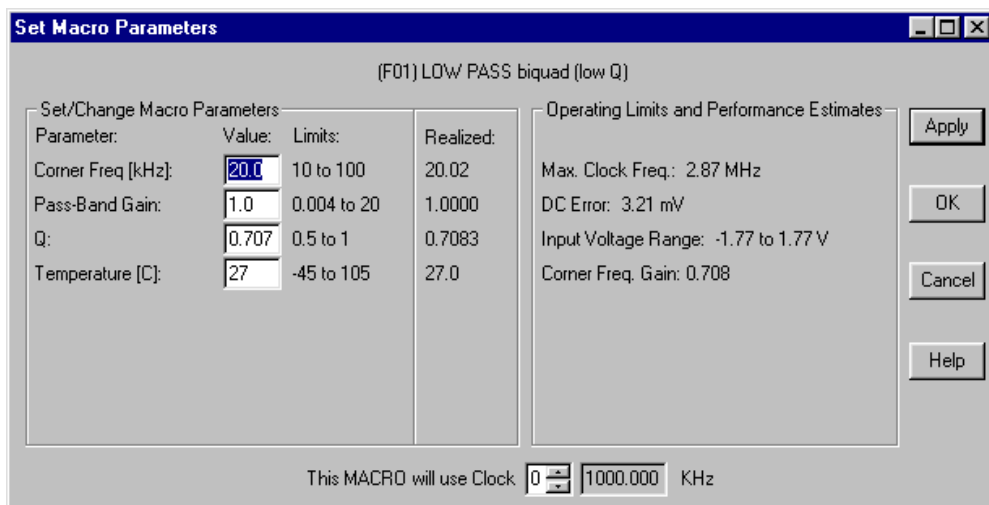
Rys. 8. View of the window for setting the macro parameters

By default, when you turn the program all buffers placed in I/O blocks are disabled. To enable them in drawing connections mode, click the left mouse button on the selected buffer. In a similar way you can change the settings of the switches in the I/O block.

If you want to delete the connection or selected macro you need to enter the delete mode by clicking the icon [icon].
After completion of the project, we can save it to disk (command **File/Save**), and then upload it to the chip using the command **Configure/Write configuration data to Serial Port** (Ctrl + W).

4. Macro description

In this exercise are used macros shown in Table 1.
On the outside of the FPAA chip there are delivered terminals:
inputs:
    terminal Y of LA cell
    terminal Y of LC cell
outputs:
    terminal Z of RA cell
    terminal Z of RC cell
The signals must be carried out through the buffers. Due to this inside the chip use the following terminals:
inputs:
    terminal Z of LA cell
    terminal Z of LC cell
outputs:
    terminal X of RA cell
    terminal X of RC cell
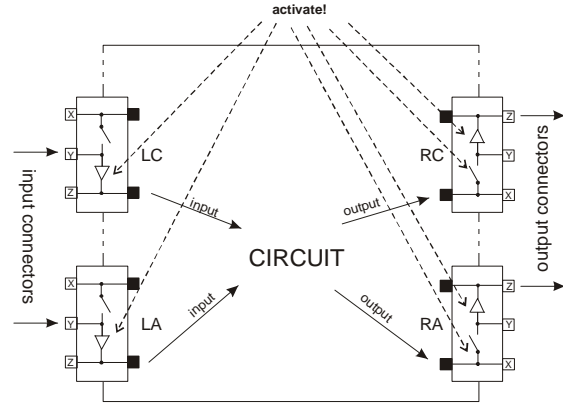**Note! You must activate the following items:**
- in cell LA and LC: buffers (buffers red, switches open);
- in cell RA and RC: buffers and switches ((buffers red, switches closed).

List of parameters of macro   Table 1

Macro Type	Description	Symbol
G1 *Inverting Gain Stage*	Inverting amplifier with a gain of: $\dfrac{V_{out}}{V_{in}} = -G$ (1) The gain G can be set (see example window of Fig.5) in the range of 0.004 to 20 V/V.	
G4 *Inverting Gain Stage with offset compensated in one phase*	The amplifier operates as G1, but additionally in the phase ϕ_2 of the clock signal amplifier compensates the offset voltage. Gain adjustment range is: 0.004-20.0V/V.	
G6 *Inverting Sum Amplifier*	The summing amplifier of the two signals, each of these signals is scaled by a factor respectively G1 and G2 in accordance with the formula: $V_{out} = -G_1 V_{in1} - G_2 V_{in2}$ (2) Gain adjustment G1 and G2 are: 0.004-20.0V / V.	
F1 *Low-Pass biquad low Q*	Transfer function: $$\frac{V_{out}(s)}{V_{in}(s)} = -\frac{4\pi^2 f_0^2 G}{s^2 + \dfrac{2\pi f_0}{Q}s + 4\pi^2 f_0^2}$$ (3) Corner frequency f_0 is defined as the frequency at which the signal gain is equal to GQ [V/V], where G is the gain in the passband and Q is the quality factor of the filter. Setting range: f_0: 8.0-100kHz; G: 0.004-20.0 V/V; Q: 0.5-1.0	
F2 *Low-Pass biquad high Q*	Transfer function: $$\frac{V_{out}(s)}{V_{in}(s)} = -\frac{4\pi^2 f_0^2 G}{s^2 + \dfrac{2\pi f_0}{Q}s + 4\pi^2 f_0^2}$$ (4) Corner frequency f_0 is defined as the frequency at which signal gain is equal to GQ [V/V], where G is the gain in the passband and Q is the quality factor of the filter. Setting range: f_0: 10.0-100kHz; G: 0.004-20.0 V/V; Q: 1.0-20.0	
F3 *High-Pass biquad low Q*	Transfer function: $$\frac{V_{out}(s)}{V_{in}(s)} = -\frac{Gs^2}{s^2 + \dfrac{2\pi f_0}{Q}s + 4\pi^2 f_0^2}$$ (5) Corner frequency f_0 is defined as the frequency at which signal gain is equal to GQ [V/V], where G is the gain in the passband and Q is the quality factor of the filter. Setting range: f_0: 8.0-100kHz; G: 0.004-20.0 V/V; Q: 0.5-1.0	
F4 *High-Pass biquad high Q*	Transfer function: $$\frac{V_{out}(s)}{V_{in}(s)} = -\frac{Gs^2}{s^2 + \dfrac{2\pi f_0}{Q}s + 4\pi^2 f_0^2}$$ (6) Corner frequency f_0 is defined as the frequency at which signal gain is equal to GQ [V/V], where G is the gain in the passband and Q is the quality factor of the filter.	

	Setting range: f_0: 8.0-100kHz; G: 0.004-20.0 V/V; Q: 1.0-20.0		
F5 *Band-Pass biquad low Q*	Transfer function: $$\frac{V_{out}(s)}{V_{in}(s)} = -\frac{2\pi f_0 \frac{G}{Q} s}{s^2 + \frac{2\pi f_0}{Q} s + 4\pi^2 f_0^2}$$ (7) Corner frequency f_0 is the filter center frequency, G is the gain in the passband and Q is the quality factor of the filter. Setting range: f_0: 8.0-100kHz; G: 0.004-20.0 V/V; Q: 0.5-1.0		
F6 *Band-Pass biquad high Q*	Transfer function: $$\frac{V_{out}(s)}{V_{in}(s)} = -\frac{2\pi f_0 \frac{G}{Q} s}{s^2 + \frac{2\pi f_0}{Q} s + 4\pi^2 f_0^2}$$ (8) Corner frequency f_0 is the filter center frequency, G is the gain in the passband and Q is the quality factor of the filter. Setting range: f_0: 4.0-100kHz; G: 0.004-20.0 V/V; Q: 1.0-20.0		
O2 *Square Wave Oscillator*	Oscillation frequency: 0.98-250kHz		
C4 *2-Input Comparator*	$$V_{out} = \begin{cases} +2.5V \text{ dla } V_{in1} > V_{in2} \\ -2.5V \text{ dla } V_{in1} < V_{in2} \end{cases}$$ (9)		
M6 Integrator	$$\frac{V_{out}(s)}{V_{in}(s)} = \frac{0.2\pi f_c G}{s}$$ (10) G is the gain for a frequency equal to 1/10 of the clock frequency (clock frequency is: f_{clk} = 1MHz). Setting range: G: 0.01 -20.0 V/V.		
M7 *Inverting Integrator*	$$\frac{V_{out}(s)}{V_{in}(s)} = -\frac{0.2\pi f_c G}{s}$$ (11) G is the gain for a frequency equal to 1/10 of the clock frequency (clock frequency is: f_{clk} = 1MHz) The adjustment range: G: 0.01 -20.0 V/V.		
M1 *Limiter*	The output voltage is limited to the range of $-GV_L$ to GV_L. Where V_L – voltage limit, G – limiter gain, Setting range: V_L: 0.1-2.5V , G: 0.004-20.0 V/V		

5. Measurements

1) Measurements of low-pass, high-pass and band-pass filters: program filter circuits according to the parameters given in Table 3. The 3dB frequency must be found, and any characteristic points. Perform measurements for a single input voltage.

Measuring tasks - filter   Table 3

Macro name	Filter type	set of parameters
F1	*Low-Pass*	Pass-Band Gain= 1 V/V, Corner freq.=10kHz, Q=0.707
F2	*Low-Pass*	Pass-Band Gain = 1 V/V, Corner freq.=10kHz, Q=3
F2	*Low-Pass*	Pass-Band Gain = 1 V/V, Corner freq.=10kHz, Q=10
F3	*High-Pass*	Pass-Band Gain = 1 V/V, Corner freq.=10kHz, Q=0.707
F4	*High-Pass*	Pass-Band Gain = 1 V/V, Corner freq.=10kHz, Q=3
F4	*High-Pass*	Pass-Band Gain = 1 V/V, Corner freq.=10kHz, Q=10
F6	*Band-Pass*	Pass-Band Gain = 1 V/V, Corner freq.=10kHz, Q=3
F6	*Band-Pass*	Pass-Band Gain = 1 V/V, Corner freq.=10kHz, Q=10
F6	*Band-Pass*	Pass-Band Gain = 1 V/V, Corner freq.=10kHz, Q=20

2) Program the following circuit: the rectangular waveform generator (O2) which output signal passes to the input of the limiter (M1) and then to the low-pass filter (F1), as shown in Fig. 9. Circuits parameters are given in Table 4. The output of the limiter (square wave signal) connect to the terminal RAX and the output of the low-pass filter (sinusoidal signal) to the terminal RCX. Redraw the shapes of both waveforms (sine and square-wave) and measure harmonics distortion.

Table 4

Macro name	set of parameters
O2	Oscillator frequency=16kHz
M1	Limiting voltage =0.5V, Gain=1 V/V
F1	Corner frequency=20kHz, Q=707
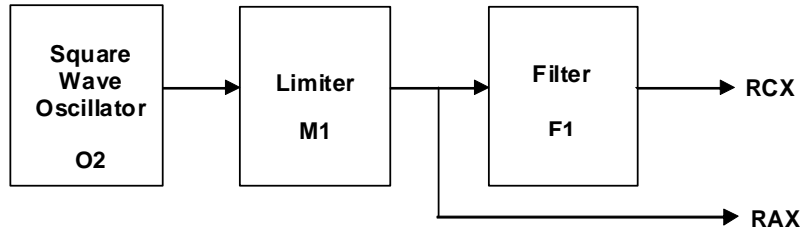
Rys. 9. Generator block diagram.

Redraw the shapes of both waveforms (sine and square-wave) and measure harmonics distortion.

6. Elaboration of the results

For measuring point 5.1 :
  a) Present the results of the measurements for individual filters on common plots.
  b) Compare programmed and measured parameters. It should comment on any discrepancies.
For measuring point 5.2
  Compare achieved measured values of THD with theoretical value and comment on that.

7. Theory

Ad. 5.1) .Amplitude frequency characteristics can be obtained by plotting the magnitude of transfer function using substitution $s = j\omega = j2\pi f$. The transfer functions of each sections and their magnitude are presented in Table 6.

Transfer functions of filters   Table 6

Fliter type	transfer function	magnitude of transfer function
Low-Pass	$$T_{LP}(s) = \frac{-4\pi^2 f_o^2 G}{s^2 + \dfrac{2\pi f_o}{Q}s + 4\pi^2 f_o^2} \quad (12)$$	$$\lvert T_{LP}(j\omega)\rvert = \frac{f_o^2 G}{\sqrt{\left(f_o^2 - f^2\right)^2 + \left(\dfrac{f_o}{Q}f\right)^2}} \quad (13)$$
Band-Pass	$$T_{BP}(s) = \frac{2\pi f_o \dfrac{G}{Q}s}{s^2 + \dfrac{2\pi f_o}{Q}s + 4\pi^2 f_o^2} \quad (14)$$	$$\lvert T_{BP}(j\omega)\rvert = \frac{G\dfrac{f_o}{Q}f}{\sqrt{\left(f_o^2 - f^2\right)^2 + \left(\dfrac{f_o}{Q}f\right)^2}} \quad (15)$$
High-Pass	$$T_{HP}(s) = \frac{Gs^2}{s^2 + \dfrac{2\pi f_o}{Q}s + 4\pi^2 f_o^2} \quad (16)$$	$$\lvert T_{HP}(j\omega)\rvert = \frac{f^2 G}{\sqrt{\left(f_o^2 - f^2\right)^2 + \left(\dfrac{f_o}{Q}f\right)^2}} \quad (17)$$

Where: f_o is *corner frequency,* Q- *quality factor*, G – passband gain

This part of the exercise is to remind the definition of the nonlinear distortion coefficient, its practical measurements and calculations for some signals. This ratio is often called Total Harmonic Distortion (THD).
Each continuous periodic signal can be represented as complex Fourier series

$$v(t) = \sum_{n=-\infty}^{+\infty} c_n e^{j\frac{n\pi t}{T/2}} \text{, where } c_n = \frac{1}{T}\int_{-T/2}^{T/2} v(t) e^{-j\frac{n\pi t}{T/2}} dt \text{ for } n = 0, \pm 1, \pm 2, \dots \text{, T – signal period} \tag{18}$$

Because of the equality $c_n = c_{-n}^*$, a series of double-sided can be replaced following a series of one-sided

$$v(t) = c_0 + \sum_{n=1}^{+\infty}\left[c_n e^{j\frac{n\pi t}{T/2}} + c_n^* e^{-j\frac{n\pi t}{T/2}}\right] = c_0 + \sum_{n} C_n \cos(n2\pi t / T + \varphi_n) \text{ ,} \tag{19}$$

where $c_0 = \dfrac{1}{T}\int_{-T/2}^{T/2} v(t)dt$ and is an average of the signal over one period, $C_n = 2\lvert c_n\rvert$, $\varphi_n = \arg(c_n)$.

A series of coefficients C_n is a series of the amplitudes of subsequent harmonic signal components *v(t)*.
The THD coefficient is defined as:

$$THD = \sqrt{\frac{\sum_{n=2}^{+\infty} C_n^2}{\sum_{n=1}^{+\infty} C_n^2}} = \frac{\sqrt{\sum_{n=2}^{+\infty} C_n^2} \, / \sqrt{2}}{\sqrt{\sum_{n=1}^{+\infty} C_n^2} \, / \sqrt{2}} = \frac{V_{harm-sk}}{V_{sk}} \tag{20}$$

where coefficient C_n are successive amplitudes of concerned harmonic signal..

Dividing the numerator and denominator by the $\sqrt{2}$ show that THD is in fact the ratio of the effective value of harmonics from N = 2 to ∞ in relation to the effective value of the signal.

In this exercise THD value of the square-wave signal and the filtered square-wave signal are measured. For a square-wave with an amplitude V_A coefficients C_n are expressed by equation:

$$C_0 = 0 \, , \; C_n = \frac{2V_A[1-(-1)^n]}{n\pi} \; . \tag{21}$$

The effective value of the square wave is equal to its amplitude:

$$V_{sk} = V_A = \frac{\sqrt{\sum_{n=1}^{+\infty} C_n^2}}{\sqrt{2}} = \frac{\sqrt{C_1^2 + \sum_{n=2}^{+\infty} C_N^2}}{\sqrt{2}} \; , \tag{22}$$

which gives after transformations: $\quad \sum_{n=2}^{+\infty} C_N^2 = 2V_A^2 - C_1^2 = 2V_A^2\left[1 - \frac{8}{\pi^2}\right] , \tag{23}$

thus THD for the square wave is equal to:

$$THD = \frac{\sqrt{\sum_{n=2}^{+\infty} C_n^2} \, / \sqrt{2}}{\sqrt{\sum_{n=1}^{+\infty} C_n^2} \, / \sqrt{2}} = \frac{V_{harm-sk}}{V_{sk}} = \frac{\sqrt{2V_A^2\left[1 - \frac{8}{\pi^2}\right]}}{\sqrt{2}V_A} = \sqrt{1 - \frac{8}{\pi^2}} = 43,52\% \tag{24}$$

THD calculation for the output signal of the low-pass filter should be performed in an approximate way for the first 11 harmonics. Magnitude of the transfer function $\left|T_{LP}(j\omega)\right|$ must be determined using the formula in Tab. 6, substituting $f = nf_g$, where f_g is the frequency of generator oscillation. THD should be approximated by formula (29).

$$THD \approx \sqrt{\frac{\sum_{n=2}^{11}\left[\frac{C_n}{V_A}\left|T(j\omega)\right|\right]^2}{\sum_{n=1}^{11}\left[\frac{C_n}{V_A}\left|T(j\omega)\right|\right]^2}} \tag{25}$$

References
[1] "*EasyAnalog*[TM] Design Software - User's Manual", Motorola Inc. 1997, USA.
[2] A. Guziński, "Linowe elektroniczne układy analogowe" WNT 1992.
[3] A. Filipkowski, "Układy elektroniczne analogowe i cyfrowe", WNT 1978.