Equalizer (Lab 3)

Completed structure

In this lab some parts within the VI shall be built, modified, explained or analyzed. The front panel of the equalizer is shown in Fig. 1 below.

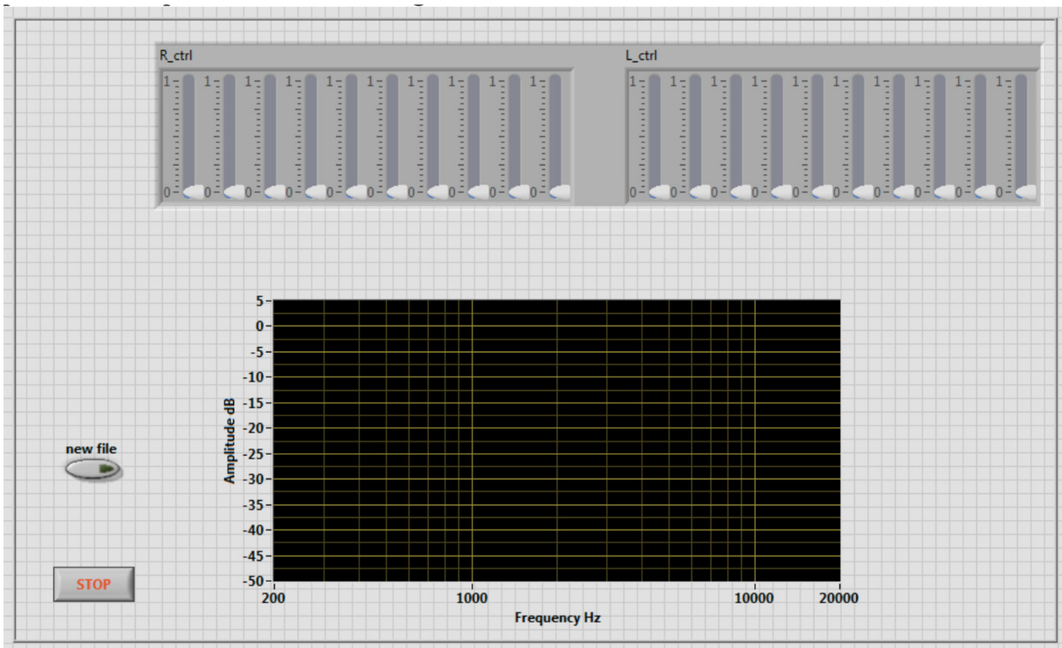

Figure. 1: The equalizer front panel.

The main program is ready to use, but the “spectrum.vi” is empty and must be developed. The main block diagram is shown in Fig. 2.

Functional description

The main VI has 3 states: “read and load a wav-file”, “play wav-file” and “stop soundcard and wait”. When the program starts or the button “New File” is clicked, the state machine enters state “read and load a wav-file”. A pop up window allows you to select a wav-file and the selected file is loaded into the program. The state machine will now go to state “play wav-file”. In this state, the file’s samples are divided into segments that are passed to the sound card output via the equalizer filters one segment at a time. The segment size is set to 4096 samples.

The samples of the segment is also passed to the spectrum module and the resulting frequency content is passed to the spectrum graph on the front panel.

When all samples of the file is played, the state machine goes to state “stop soundcard and wait” and will stay there until “new file” is clicked or the program is stopped. If “new file” is clicked while the file is playing, it will stop the sound card and go to state “read and load a wav-file”ready for a new file selection.

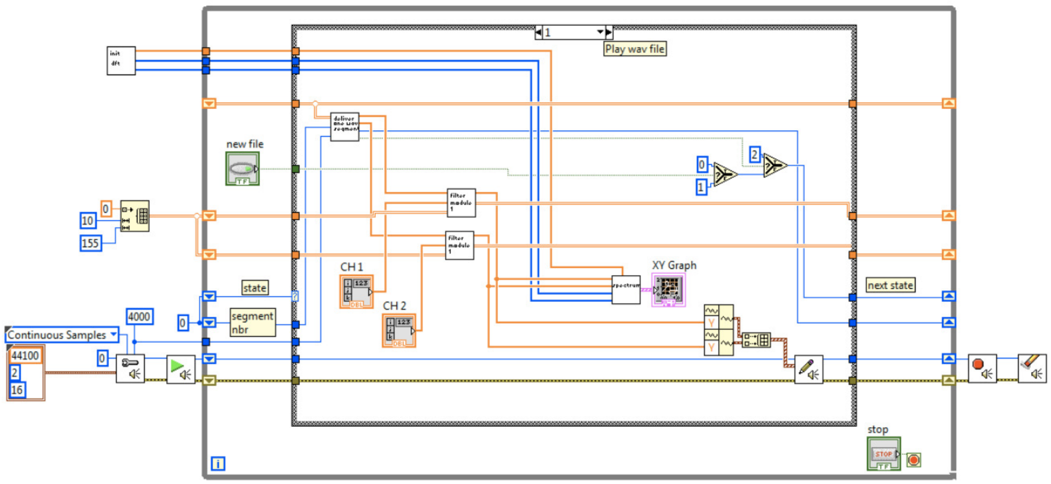

Figure. 2: Main block diagram

The spectrum module

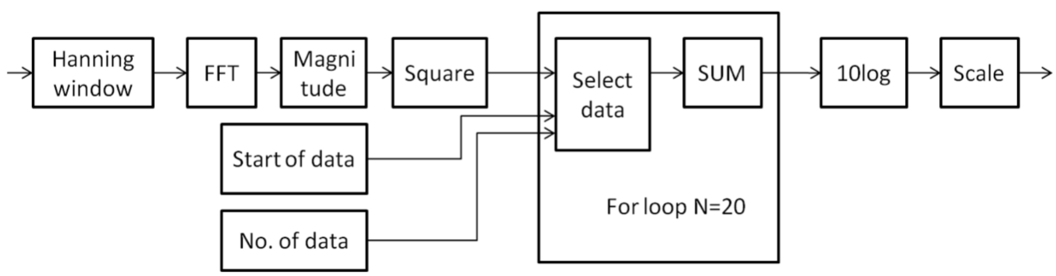

Figure. 3: Spectrum module

Program the spectrum module as indicated by the block diagram above.

- The spectrum (power versus frequency) shall be shown as 20 bars along a logarithmic frequency scale from 200 to 20000Hz, (see Fig 1). The constant array “Center freq” will place the bars correctly along the logaritmic frequency axis.
- The FOR loop will compute the amplitude of one frequency bar per round.
- The frequency range covered by each bar shall be proportional to the frequency. This will be handled by the block “Select data”. In each round this block shall fetch “No. of data” samples starting at “Start of data”.
- The height of a bar shall be proportional to the total power in the frequency span covered by that bar. This is handled by the blocks “Square” and “SUM”.
- The Y- axis shall be in dB from +5 to -60. Use the single test tone produced by test.wav to scale the amplitude of the output to 0 dB.

Filter module

Describe how the filter module (as shown in Fig. 4) is working.

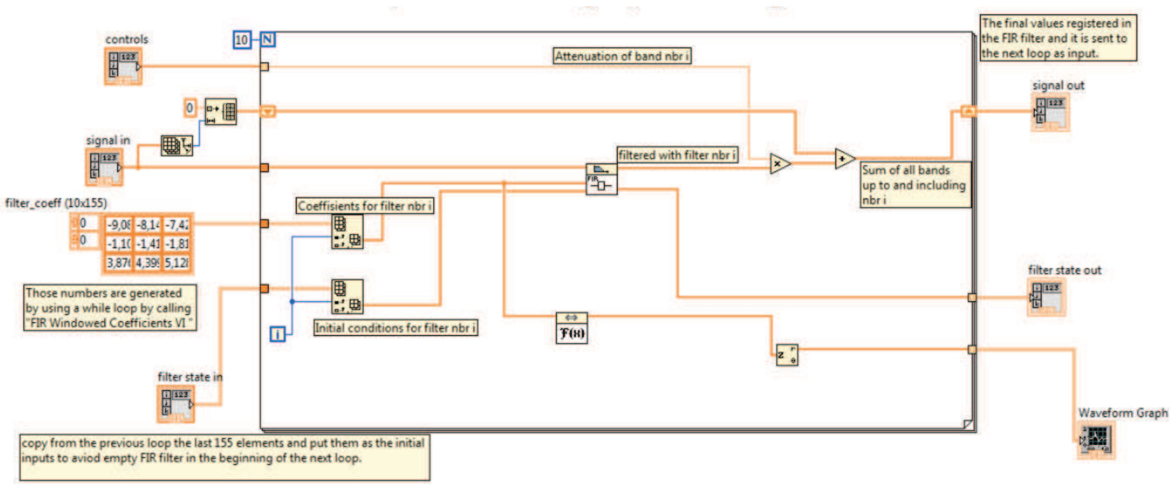

Figure. 4: Filter module diagram

Estimate the total number of multiplications per second performed in the original filter module.

Find the frequency response of each filter.
(Let the input data to the filter be a unit pulse (length 1024), place a DFT plus a complex/polar behind the filter inside the loop and connect the amplitude part to a waveform graph outside the for loop. Another way to show the frequency response is shown in Fig. 4. Why is it correct?)

Try to make a more efficient filter module by combining all impulse responses into one and use this combined response in one filter.

Estimate the total number of multiplications per second performed in this efficient filter module.

What is the purpose of the filter state input and output?

Why is it important that the filters have the same time delay?

Equalizer

Test the complete equalizer both with the original and with the more efficient filter module.