



Noise Filtering with basic Labview Functions

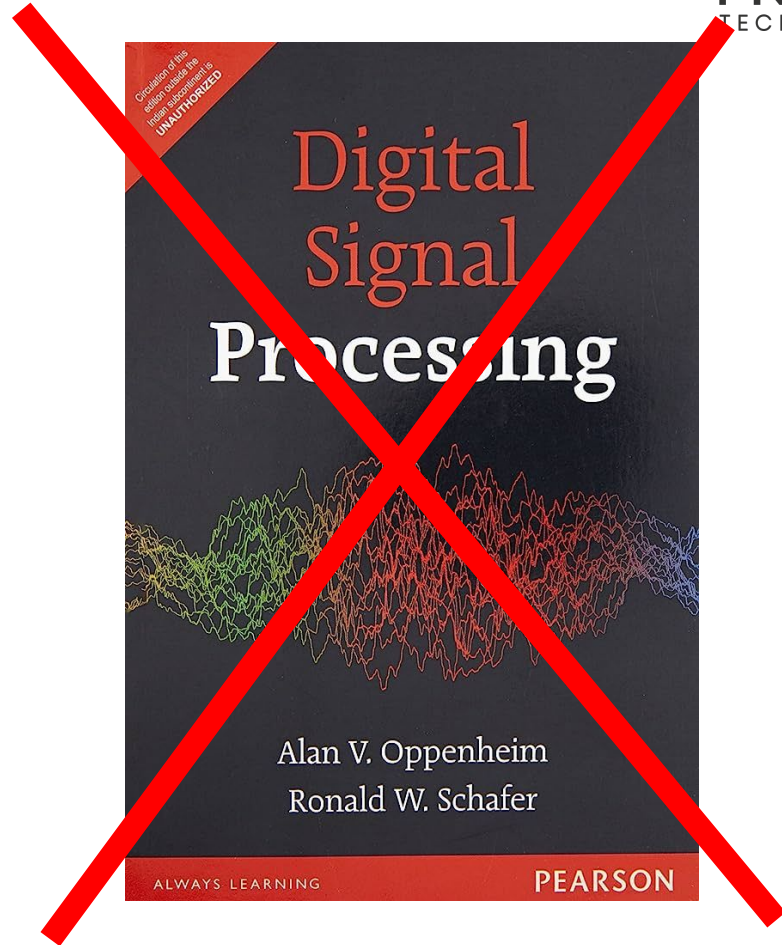
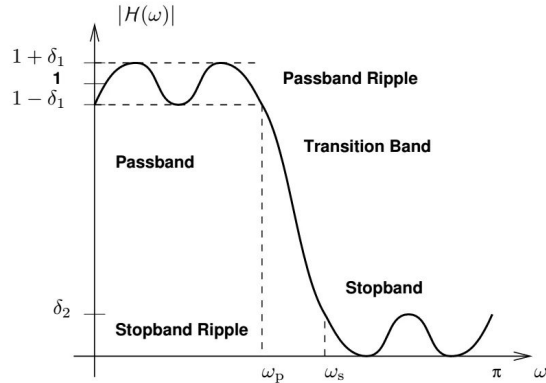
László Balogh

laszlo.balogh@prodsp.hu

BudLUG Reboot, 2023-06-15

Content

- Aim
- Manual Coding
 - Moving Average
 - Exponential Filter
- Built-in Features
- Order Estimation
- Multichannel Filtering

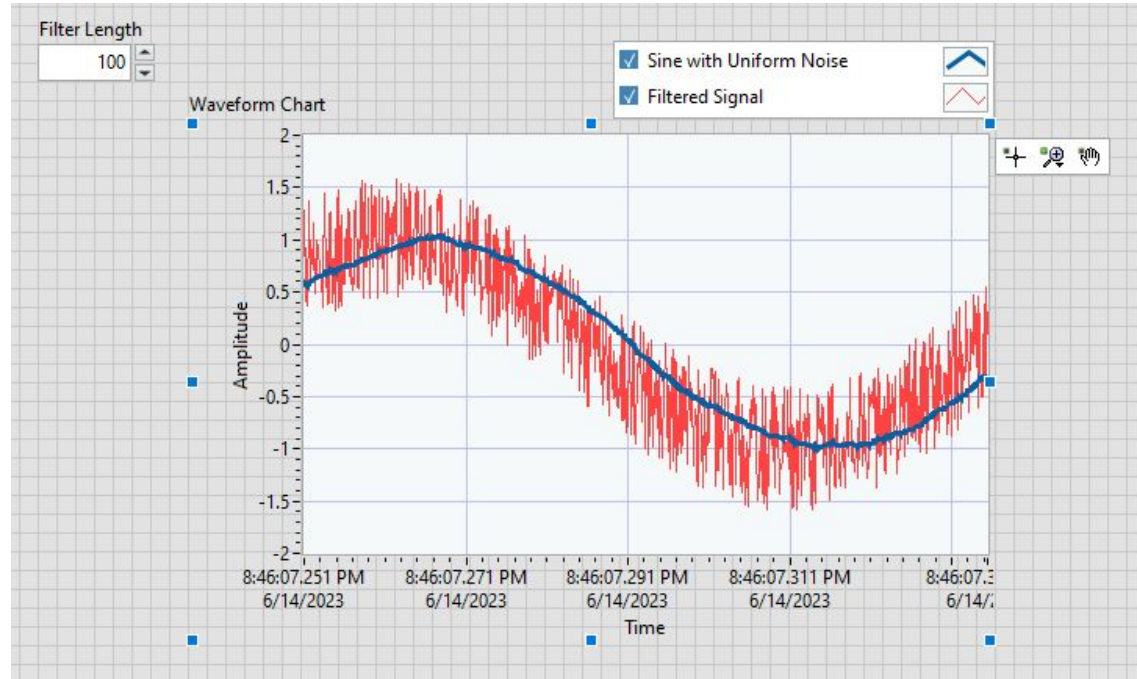


Aim

- Noise filtering for DAQ

Assumptions

- only PC platform
- 1 or more channel
- time domain requirements



Toolset

The screenshot displays the PRODSP Toolset interface. On the left is a vertical navigation menu with the following items: Generation, Unit Test Framework, Measurement I/O, Instrument I/O, Vision and Motion, Mathematics, **Signal Processing** (highlighted in blue), Data Communication, Connectivity, Control & Simulation, Express, Addons, Favorites, User Libraries, Select a VI..., Real-Time, FPGA Interface, TestStand, and Industrial Communications. The main workspace is titled "Signal Processing" and contains several tool icons. A sub-menu titled "Filters" is open, showing a grid of filter tools. The "Filters" sub-menu includes: Wfm Generation, Sig Generation, Digital Filter Design, and Time Frequency Analysis. The "Filters" sub-menu grid contains: Butterworth, Chebyshev, Inv Chebyshev, Elliptic, Bessel, Equi-Ripple LP, Equi-Ripple HP, Equi-Ripple BP, Equi-Ripple BS, Inverse f, Zero Phase ($\Phi=0$), FIR Win Filter, Median Filter, Savitzky-Golay, Mathematical Morphological..., Advanced IIR, and Advanced FIR.

Moving Average

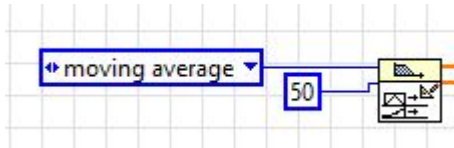
- Best theoretical description
- Straightforward implementations

$$SMA = \frac{A_1 + A_2 + \dots + A_n}{n}$$

where:

A = Average in period n

n = Number of time periods



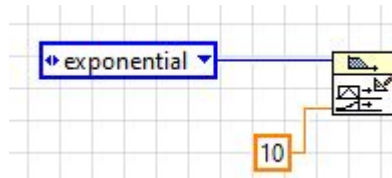
Exponential Filter

- Extremely fast

$$s_0 = x_0$$

$$s_t = \alpha x_t + (1 - \alpha)s_{t-1}, \quad t > 0$$

where α is the *smoothing factor*, and $0 < \alpha < 1$.



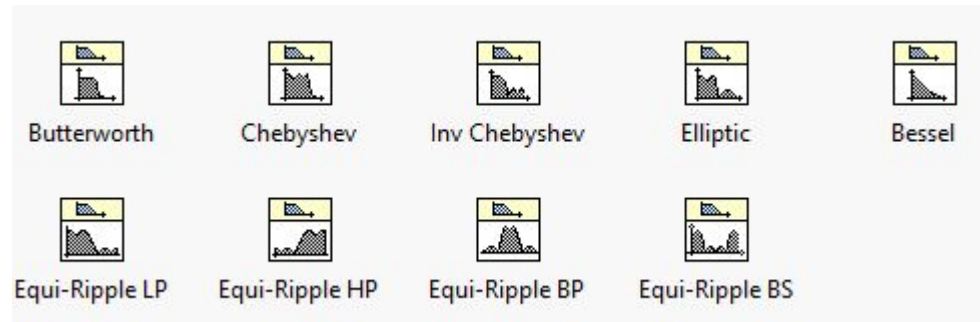
Which One?

The **Butterworth filter** is a type of signal processing filter designed to have a frequency response that is as flat as possible in the passband.

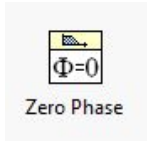
Chebyshev filters are analog or digital filters that have a steeper roll-off than Butterworth filters, and have either passband ripple (type I) or stopband ripple (type II).

An **elliptic filter** (also known as a Cauer filter) is a signal processing filter with equalized ripple (equiripple) behavior in both the passband and the stopband.

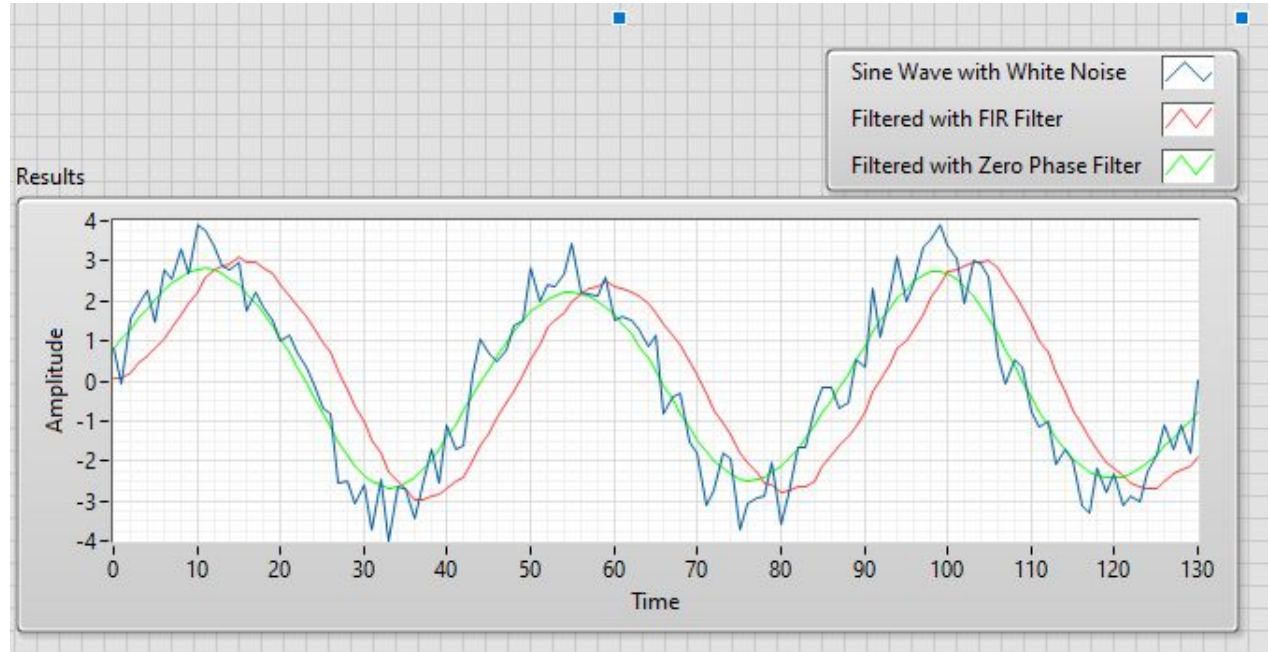
Bessel filter is a type of analog linear filter with a maximally flat group/phase delay (maximally linear phase response), which preserves the wave shape of filtered signals in the passband.



Special Filter - Zero Phase



- only for offline



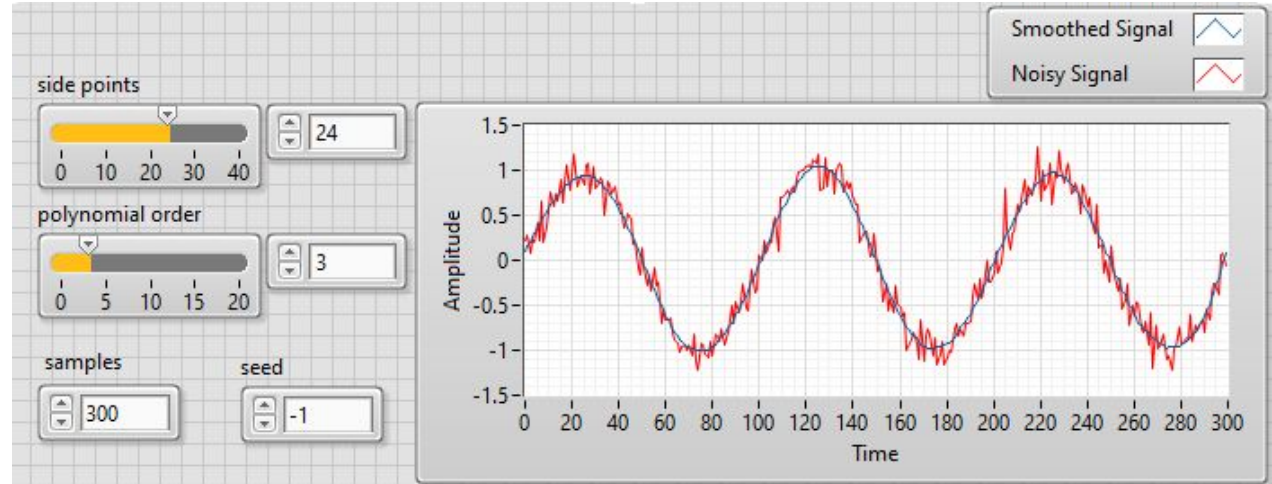
Special Filter - Savitzky Golay



- convolution
- polynomial fitting (LS)

Easy to calculate

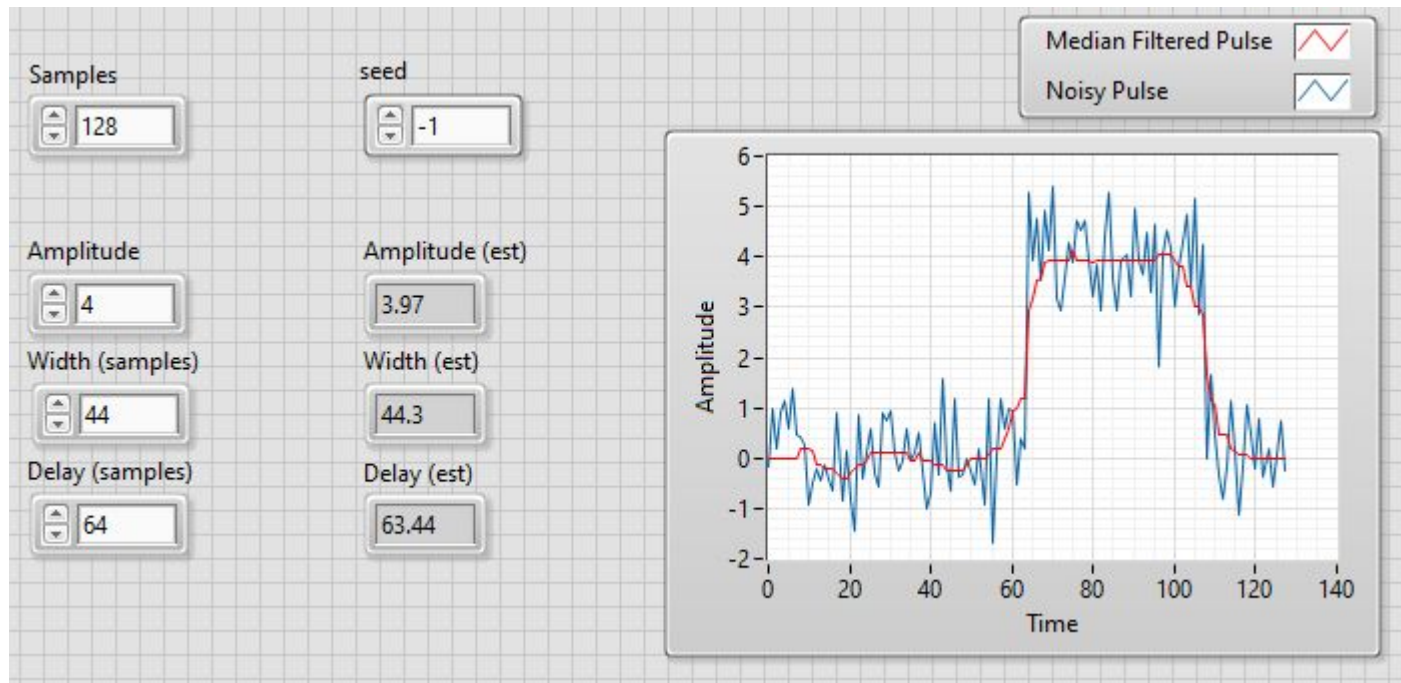
- Derivative
- Integral



Median Filter



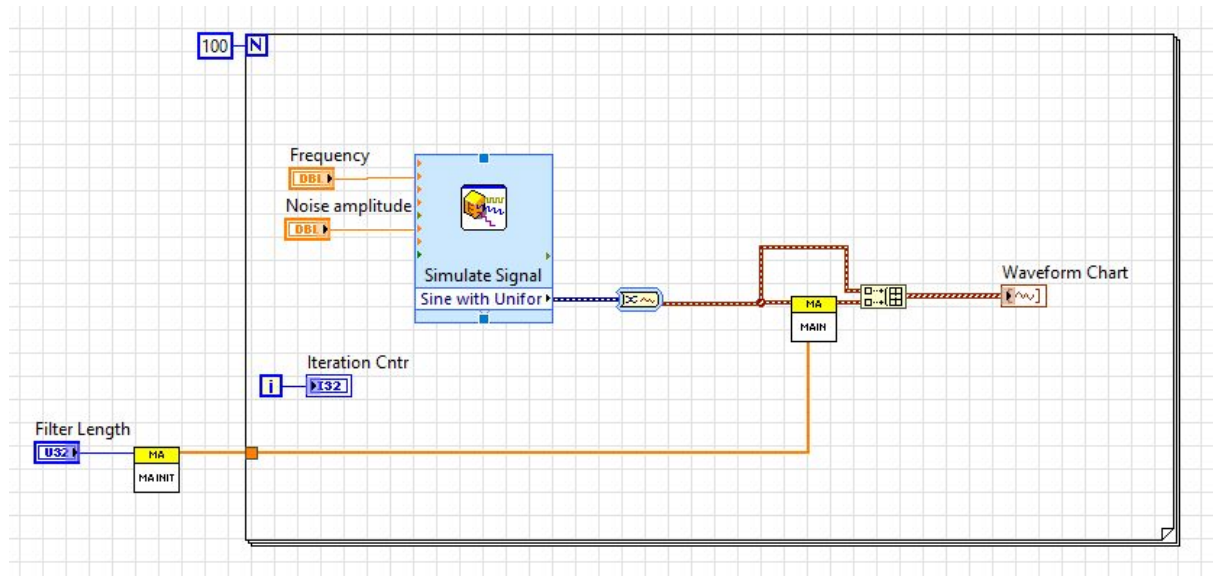
- Non-linear



Testing

Key Performance Indicator

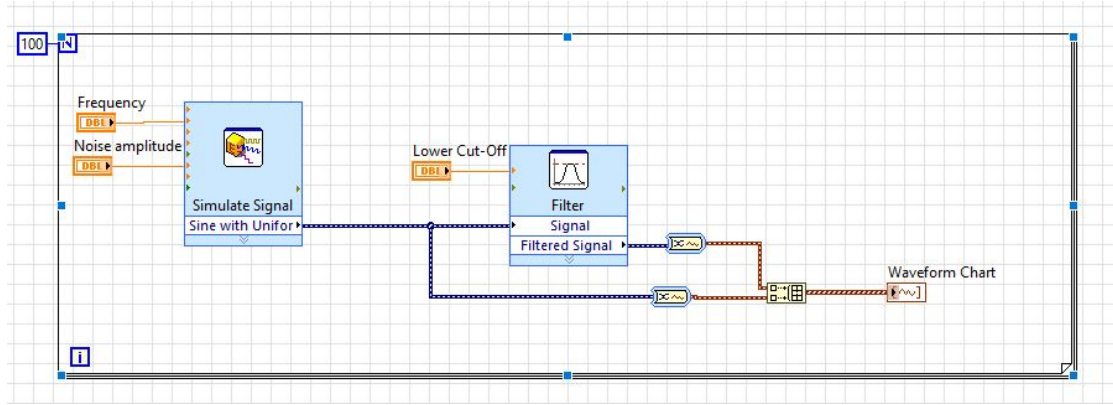
- Speed
- Memory Usage



Moving Average - Manual Implementation

Profile Data

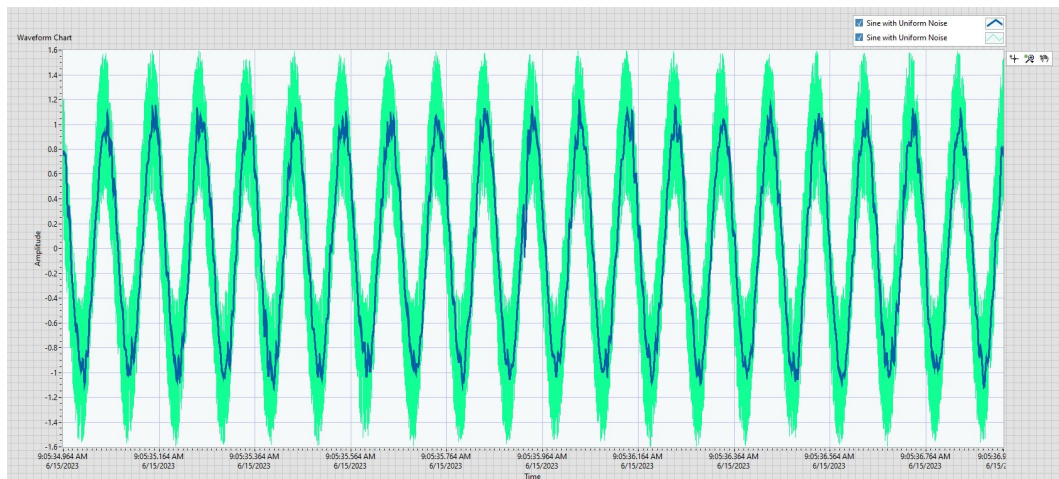
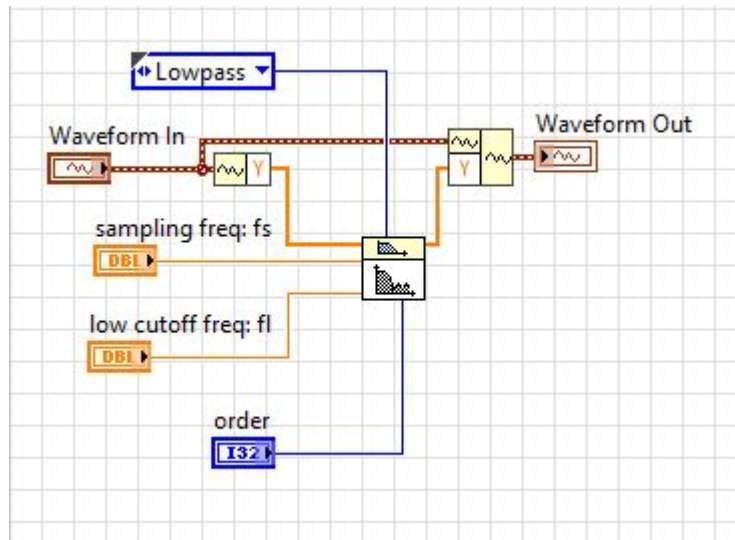
	VI Time	Sub VIs Time	Total Time	# Runs	Average	Shortest	Longest	Diagram	Display	Draw	Tracking	Locals	Avg Bytes	
subSigGeneratorBlock.vi	5726	209962	215689	50	115	68	908	5726	0	0	0	1	1618.30k	1
ex_GenAddAttribs.vi	2888	848	3735	50	58	42	192	2845	0	0	0	42	4.55k	4
ex_SetAllExpressAttribs.vi	504	0	504	50	10	8	24	504	0	0	0	0	4.45k	4
ex_SetExpAttribsAndT0.vi	328	504	832	50	7	5	28	328	0	0	0	0	4.07k	4
Nearest Frequency for Block.vi	93	47	140	50	2	1	3	93	0	0	0	0	6.40k	6
sub2ShouldUseDefSigName.vi	57	0	57	1	57	57	57	57	0	0	0	0	3.49k	3
subShouldUseDefSigName.vi	49	57	106	1	49	49	49	49	0	0	0	0	2.57k	2
Nearest Freq in Int Cycles.vi	47	0	47	50	1	1	2	47	0	0	0	0	4.40k	4
ex_CorrectErrorChain.vi	42	0	42	50	1	1	2	42	0	0	0	0	5.55k	5
subGetSignalName.vi	27	0	27	1	27	27	27	27	0	0	0	0	4.77k	4
Waveform Array To Dynamic.vi	15	0	15	50	0	0	1	15	0	0	0	0	2.05k	2
Dynamic To Waveform Array.vi	14	0	14	50	0	0	1	14	0	0	0	0	2.05k	2
Clear Errors.vi	0	0	0	0	0	0	0	0	0	0	0	0	0.00k	0
DU64_U32AddWithOverflow.vi	0	0	0	0	0	0	0	0	0	0	0	0	0.00k	0
DU64_U32SubtractWithBorrow.vi	0	0	0	0	0	0	0	0	0	0	0	0	0.00k	0
Morpho Test.vi	0	0	0	0	0	0	0	0	0	0	0	0	0.00k	0
Timestamp Add.vi	0	0	0	0	0	0	0	0	0	0	0	0	0.00k	0
Timestamp Subtract.vi	0	0	0	0	0	0	0	0	0	0	0	0	0.00k	0
subInternalTiming.vi	0	0	0	0	0	0	0	0	0	0	0	0	0.00k	0
Moving Average.lvlib:MA Core.vi	4047247	0	4047247	500000	2	2	18812	16219413	2582841	2167021	0	0	8.00k	8
Moving Average.lvlib:Main.vi	1290279	4047247	41762750	50	25806	21190	47516	1052893	32594	203222	1571	0	4792.50k	4
Moving Average.lvlib:Test Main.vi	1200548	41981304	43181852	1	1200548	1200548	1200548	24972	671518	225059	279000	0	1.00k	5
Moving Average.lvlib:MA Init.vi	7	0	7	1	7	7	7	7	0	0	0	0	2.84k	2
[Test Main.vi]	2714	14	2728	50	54	43	186	2714	0	0	0	0	2.99k	2
[Test Main.vi]	130	215689	215819	50	3	2	4	130	0	0	0	0	3.02k	3
NI_AALBase.lvlib:Uniform White Noise.vi	168406	0	168406	50	3368	1415	18393	168406	0	0	0	0	2.25k	2
NI_AALBase.lvlib:Sine Wave.vi	34397	0	34397	50	688	344	16432	34397	0	0	0	0	2.37k	2



- Performance OK
- Cannot tune filter in run-time

The screenshot shows the 'Configure Filter [Filter]' dialog box. The 'Filtering Type' is set to 'Lowpass'. Under 'Filter Specifications', the 'Cutoff Frequency (Hz)' is set to 1000 and the 'High cutoff frequency (Hz)' is set to 400. The 'Infinite impulse response (IIR) filter' option is selected, and the 'Topology' is set to 'Butterworth' and the 'Order' is set to 25. The 'View Mode' is set to 'Signals' and 'Show as spectrum' is unchecked. The 'Scale Mode' is set to 'Magnitude in dB' and 'Frequency in log' is unchecked. The 'Input Signal' section shows a plot of the input signal with 'Amplitude' on the y-axis (ranging from -100 to 100) and 'Time' on the x-axis (ranging from 0 to 1). The plot shows a noisy signal with a red line indicating the 'Sample Rate'. Below the plot, a message states: 'Result Preview and Transfer Function cannot be displayed. The current Filter Specifications do not meet the Nyquist criterion for the given Input Signal.' The dialog box has 'OK', 'Cancel', and 'Help' buttons at the bottom.

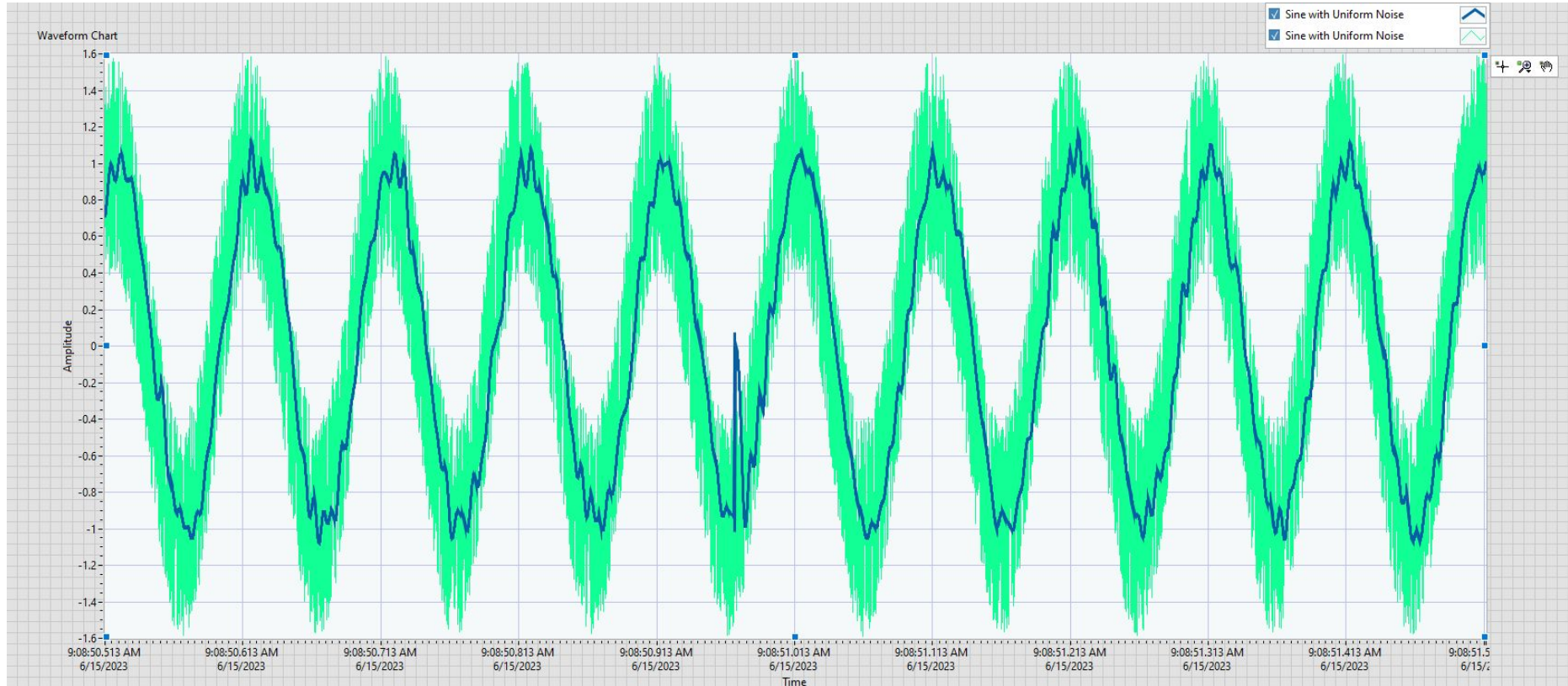
Basic Implementation



subSigGeneratorBlock.vi	/93	11835	12629	50	16	8	40	/93	0	0	0	0	
Basic Implementation.lvlib:Main Inv Cheb.vi	21781	49242	71023	50	436	15	13163	963	605	20212	0	0	Basic Impleme
Basic Implementation.lvlib:Test Main Inv Cheb.vi	74959	84029	158988	1	74959	74959	74959	2148	33754	38846	212	0	Basic Impleme
[Test Main Inv Cheb.vi]	37	12629	12666	50	1	0	4	37	0	0	0	0	Basic Impleme

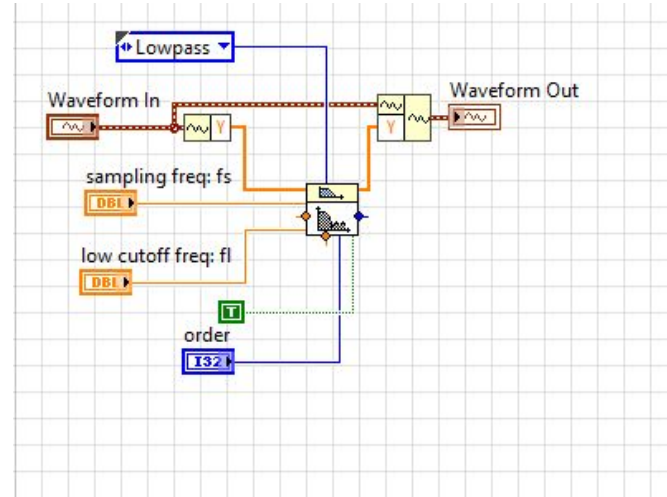
- Fast: 436 us / 1000 samples
- Small Memory Footprint: 326 kByte

Basic Implementation - Problem

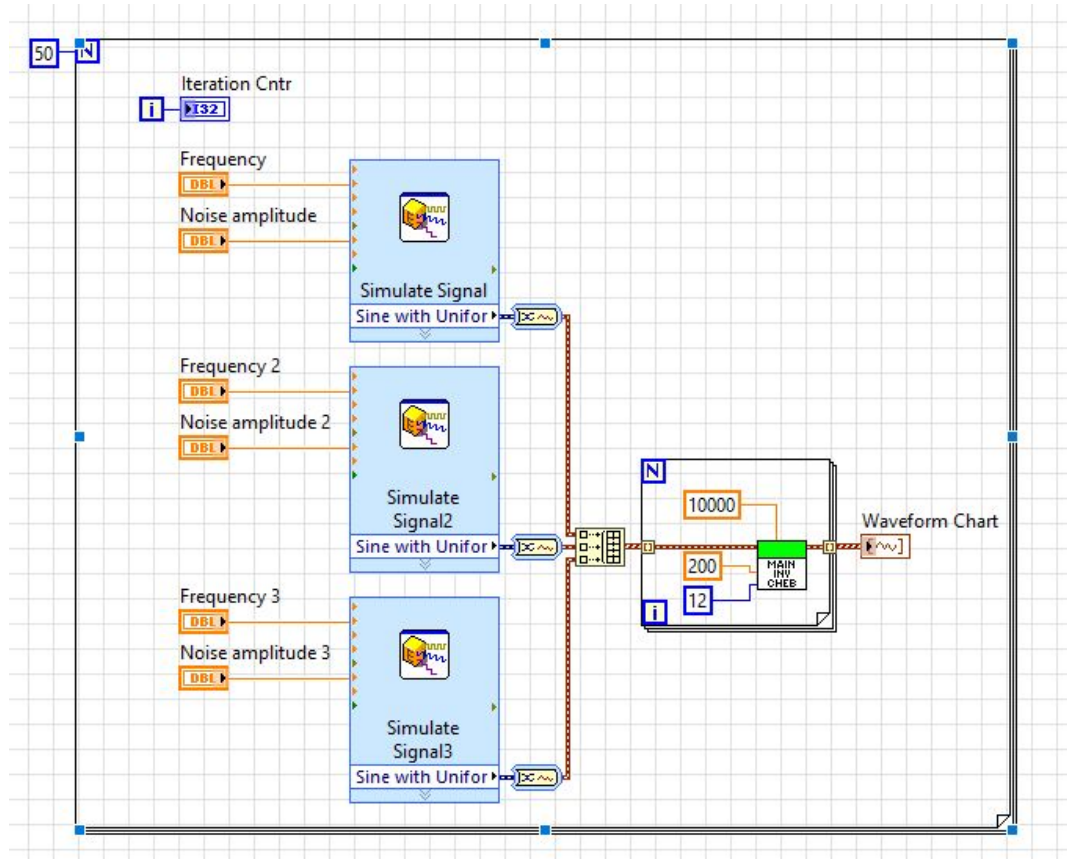


Basic Implementation - Solution

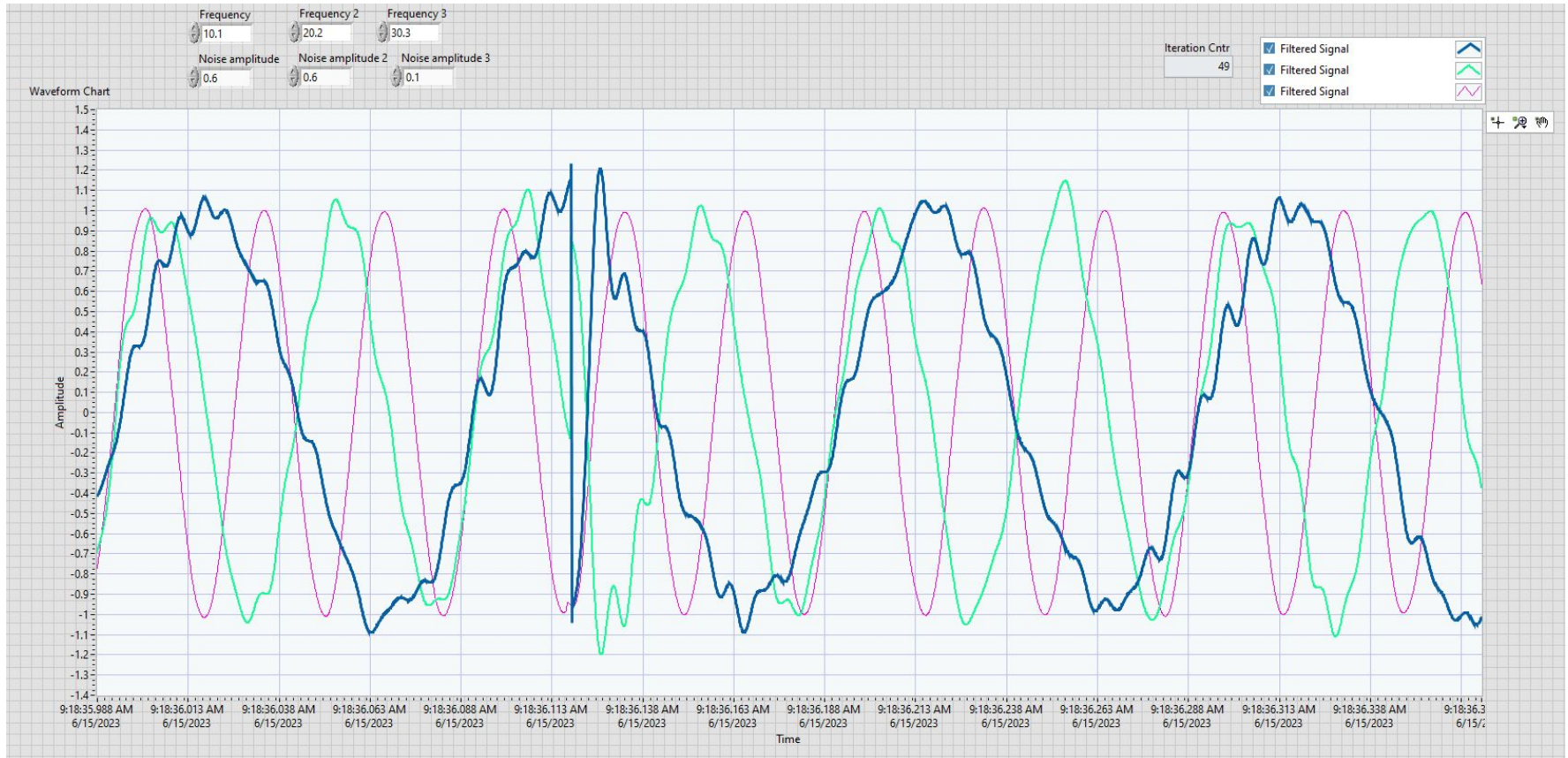
init/cont (init: F)



Multiple Signals



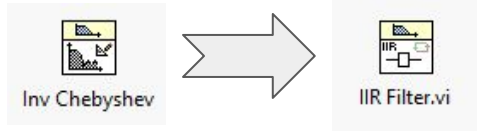
Multiple Signals - Problem



Multiple Signals - Solution

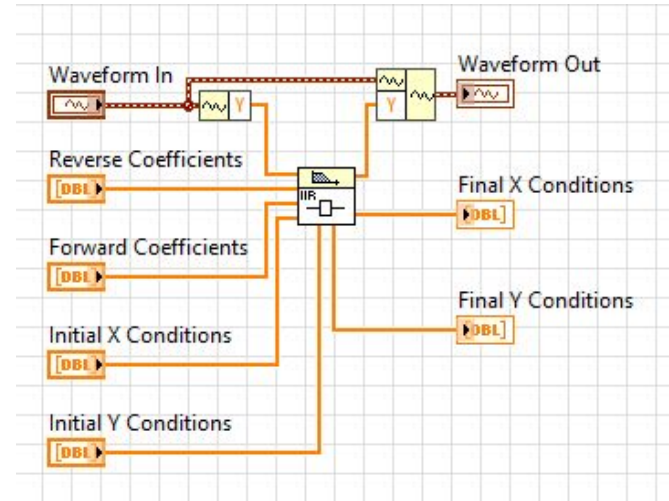
Design once

Filter N times

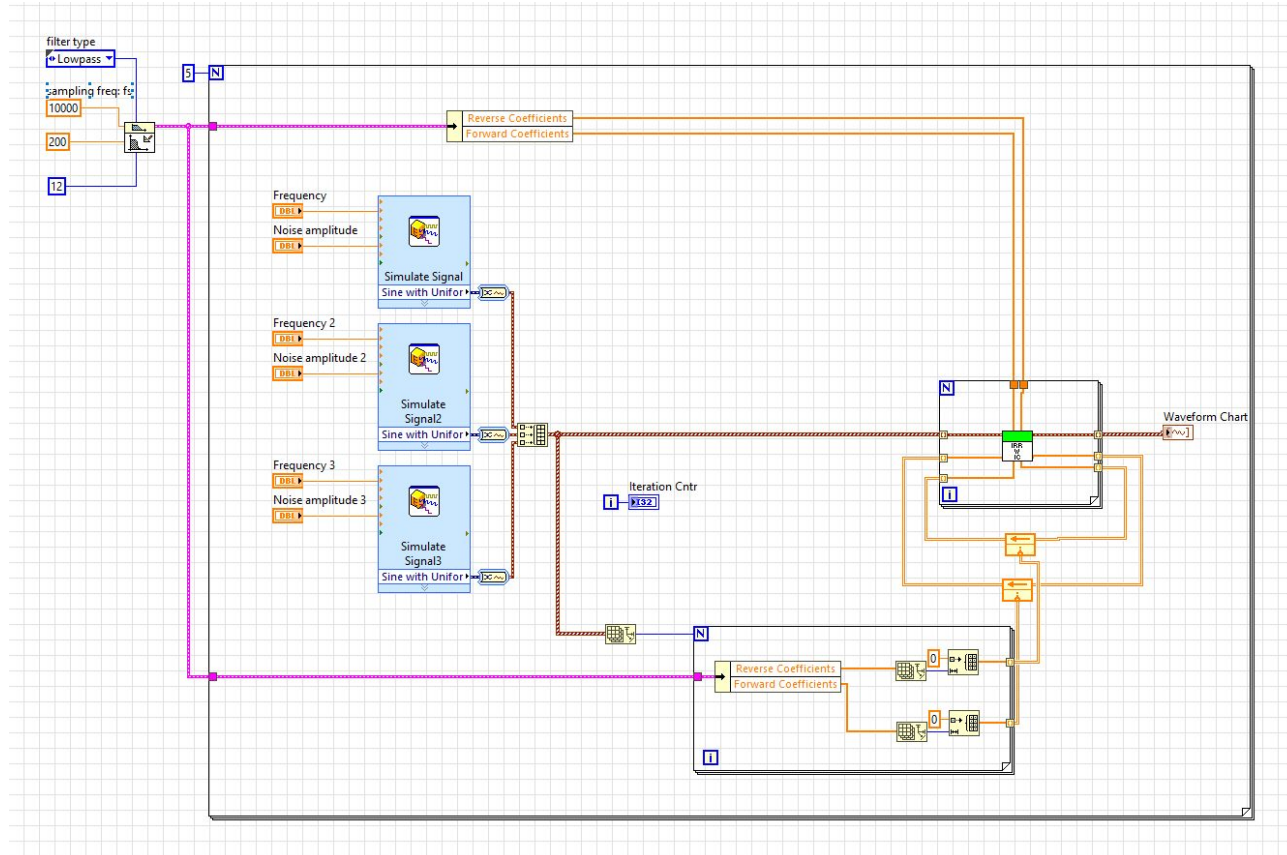


NI nomenclature:

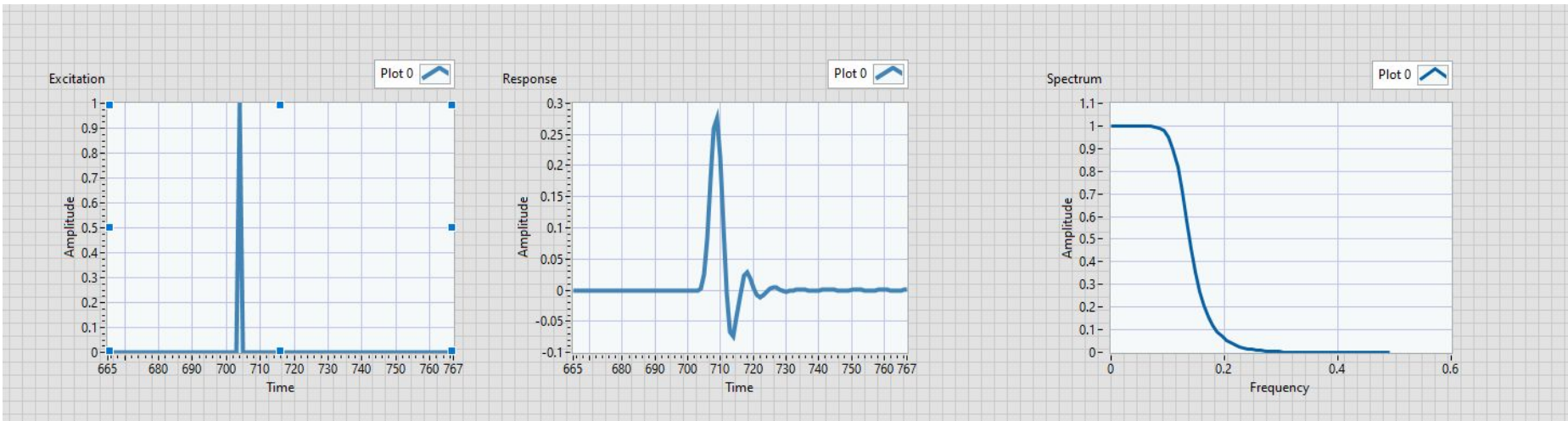
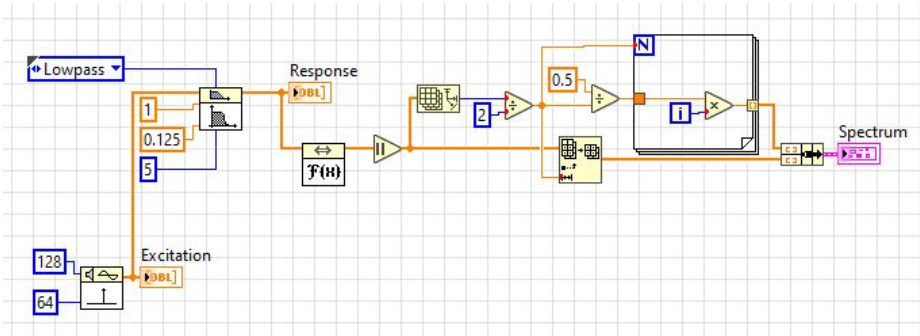
- I.C. : Initial Condition



Multiple Signals - Solution

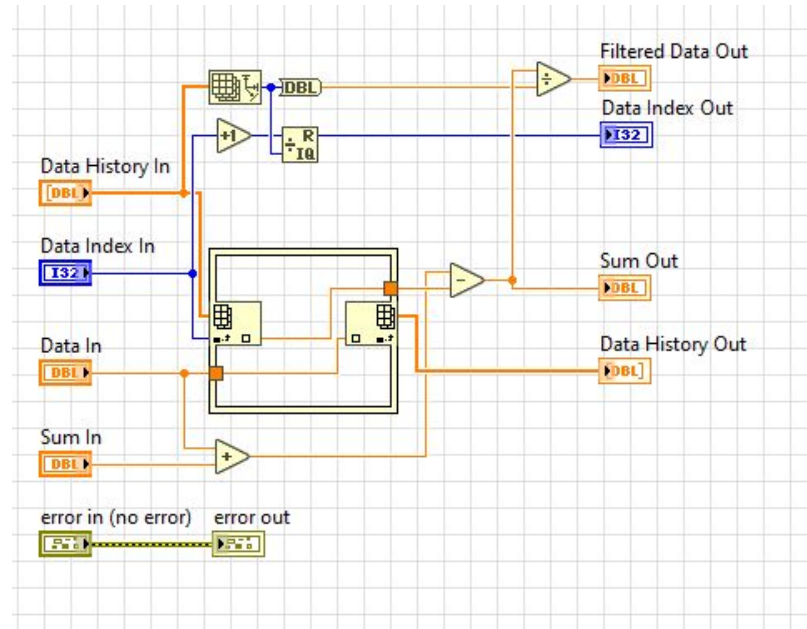


Extra 1 - Simple Spectrum Calculation



Extra 2 - Moving Average - Manual Implementation

- For long measurements better speed



Conclusions

- Express VI: good solution for quick measurement
- 1 channel: broader function selection, rich API
- N channel: not all the functions are available, little bit more programming

Next steps:

- Digital Filter Design Toolkit
- Advanced Signal Processing Toolkit