

Alimomeni

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CA2/ESL

Part1:

Advantages:

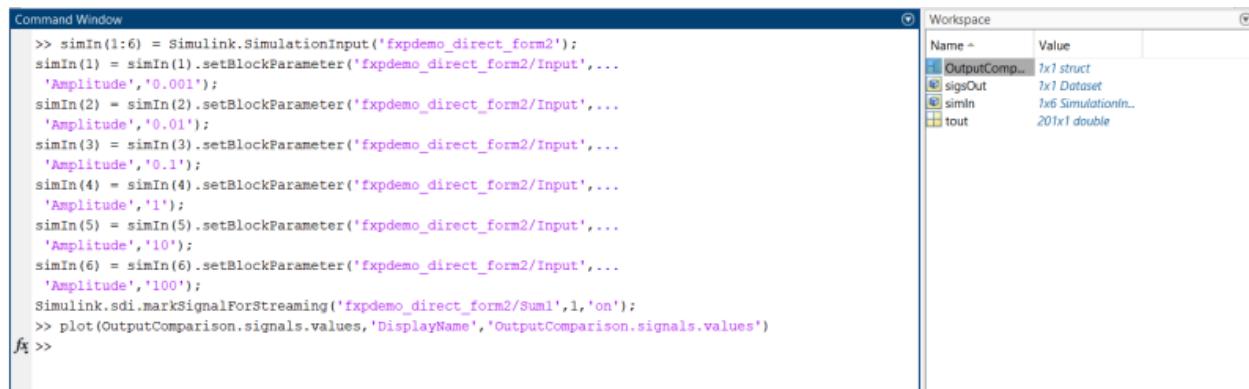
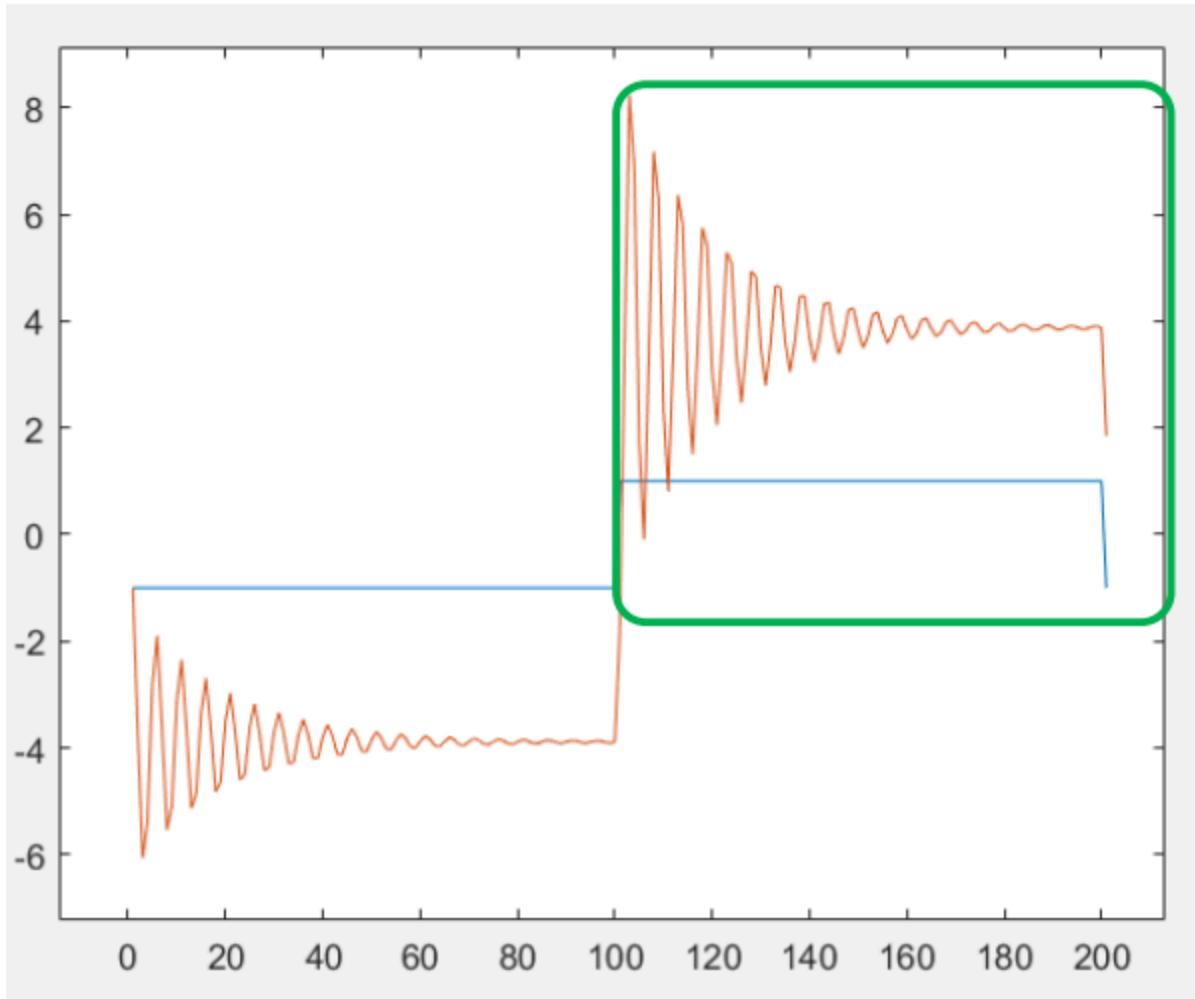
1. **Simpler and faster hardware implementation:** In fixed-point systems, due to simpler computations and fewer required hardware elements, processing speed increases and manufacturing costs decrease.
2. **More precision and control:** Fixed-point systems offer more precision and control over numerical operations in specific, limited ranges, which is useful in real-time and embedded systems.

Disadvantages:

1. **Higher risk of overflow and underflow:** In fixed-point systems, managing overflow and underflow requires careful design, and these issues are more likely compared to floating-point systems.
2. **Limited range:** With the same number of bits, fixed-point systems have a more limited dynamic range compared to floating-point systems, making them less suitable for applications that require handling very large or very small numbers.

Digital Filter Analysis:

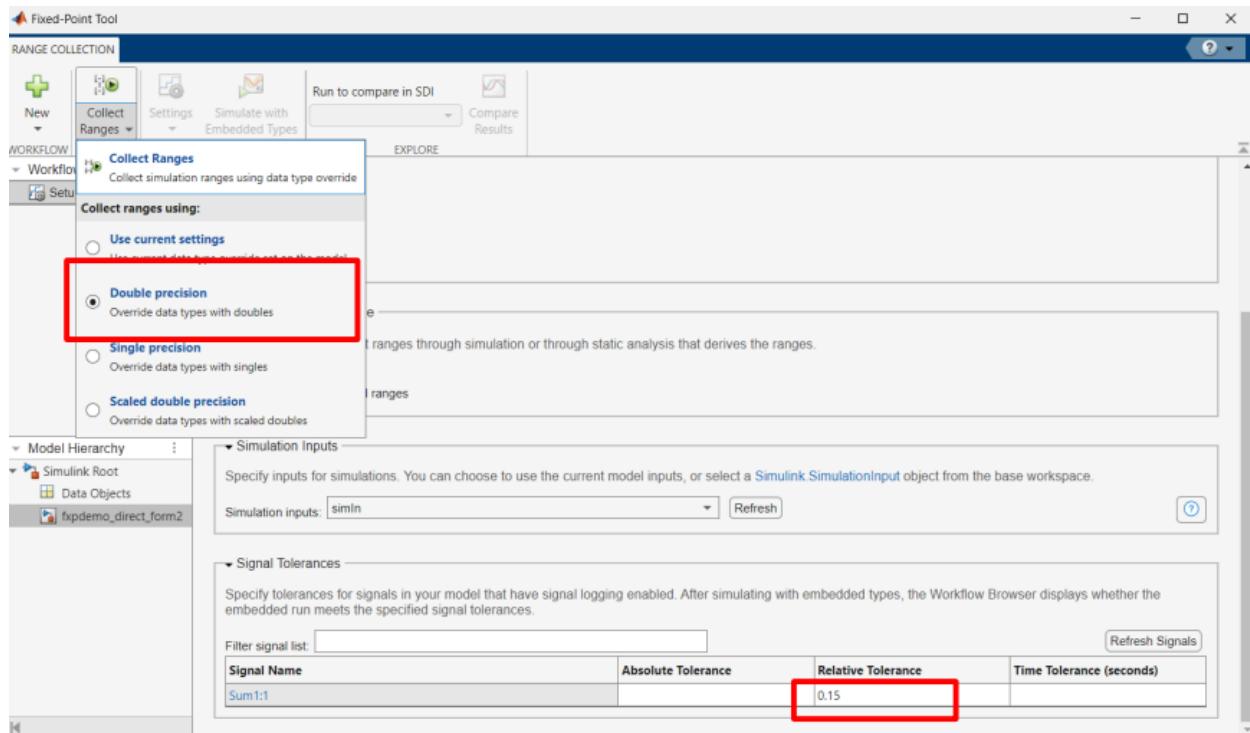
Simulation Results and Comparison Output Chart:



The output chart based on the input, after applying the corresponding MATLAB code, is shown above.

The output approximately corresponds to the green section in the image, which matches and aligns with the main graph.

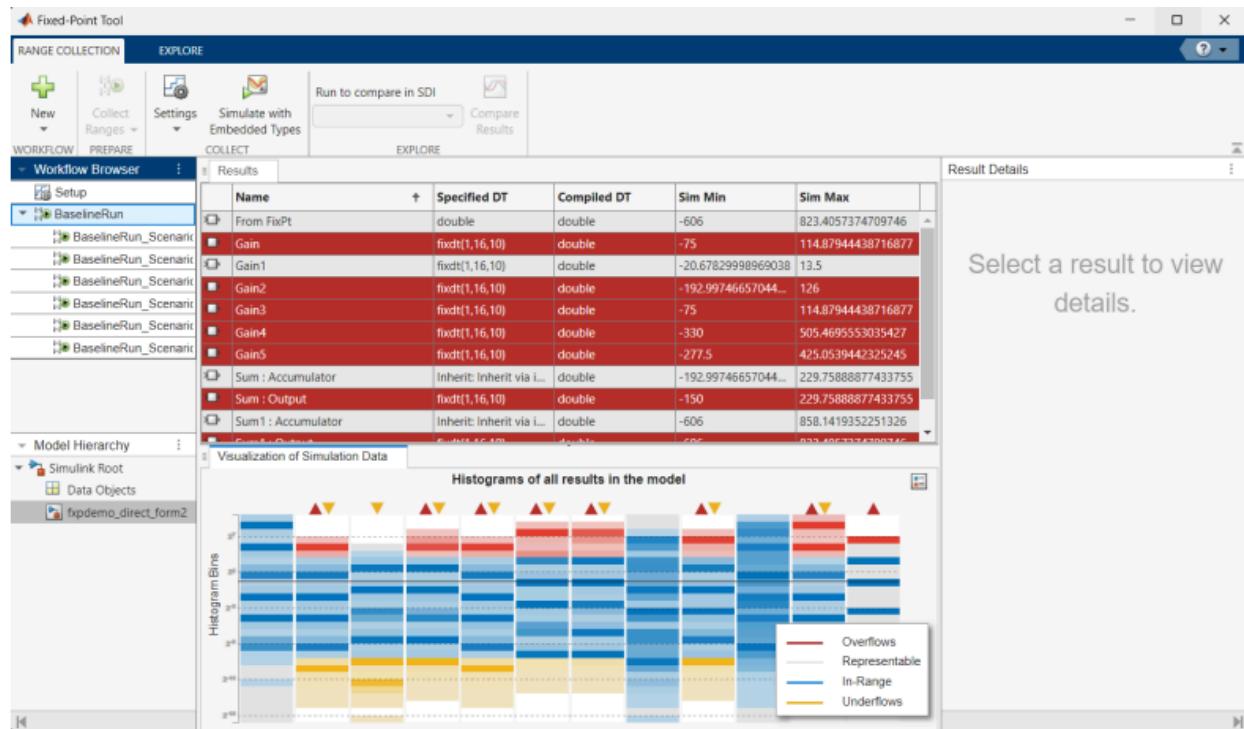
Converting the Digital Filter to Fixed Point and Comparing it with Floating Point Representation:



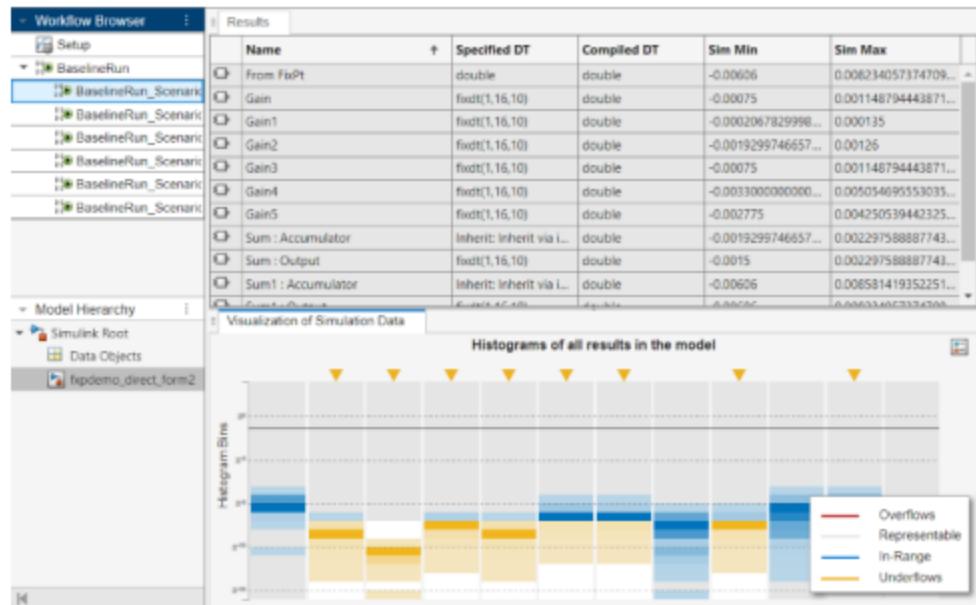
3.Underflow and Overflow Status for Each Set of 6 Inputs and Their Relation to the Input Signal:

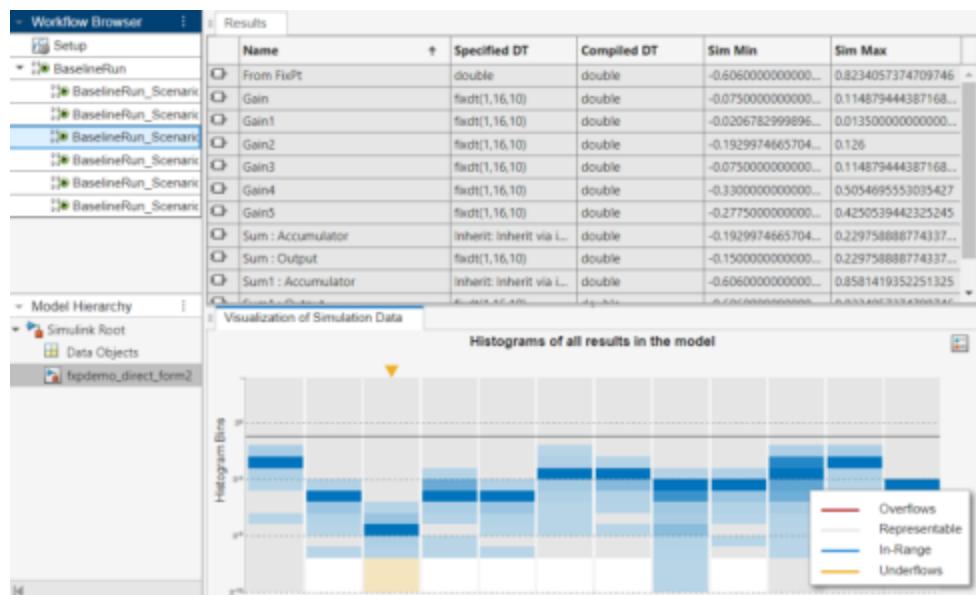
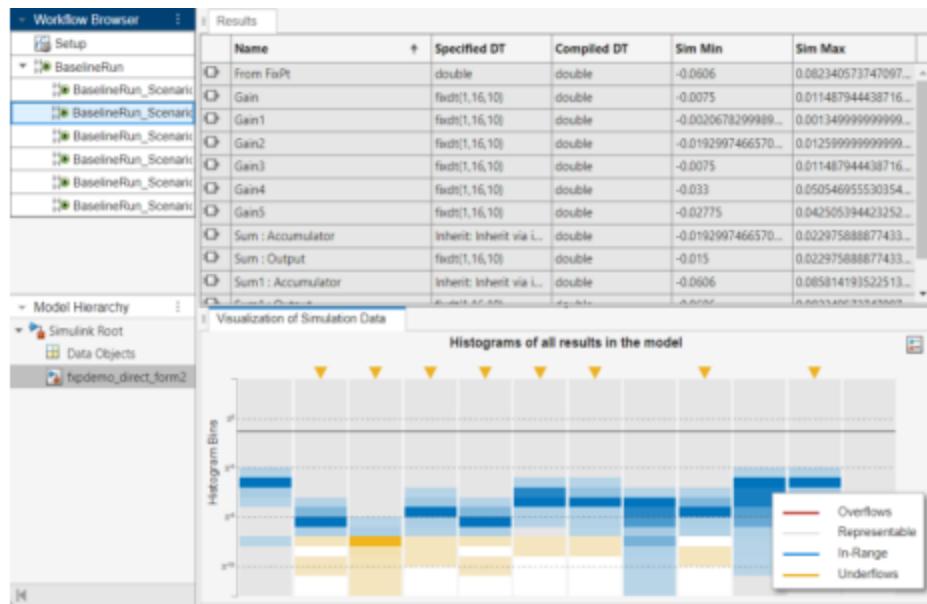
It can be observed that a significant amount of underflow occurs initially, which is expected since the input values are small at the beginning, causing the system to experience underflow.

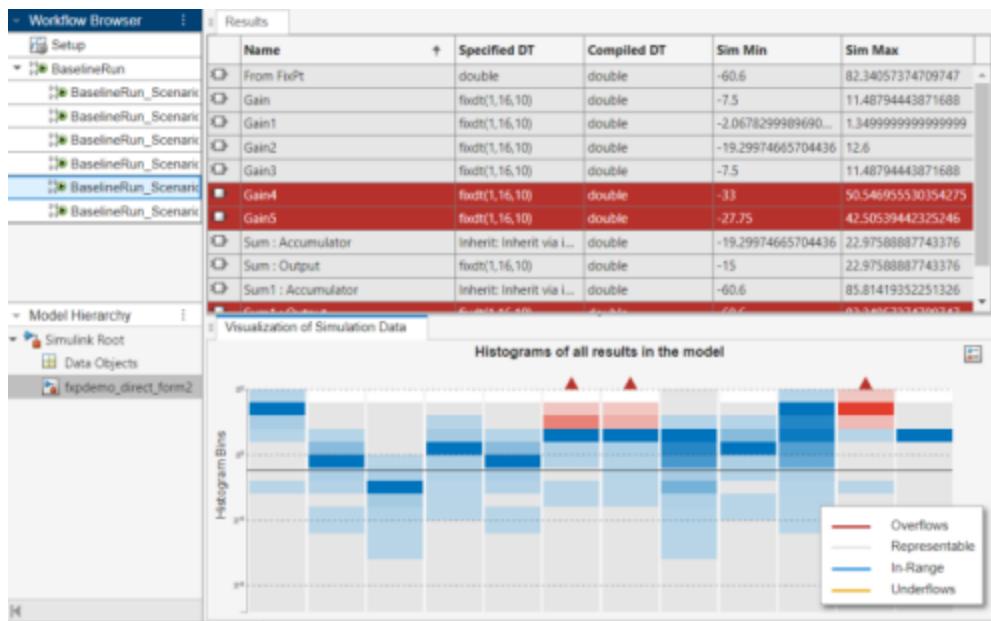
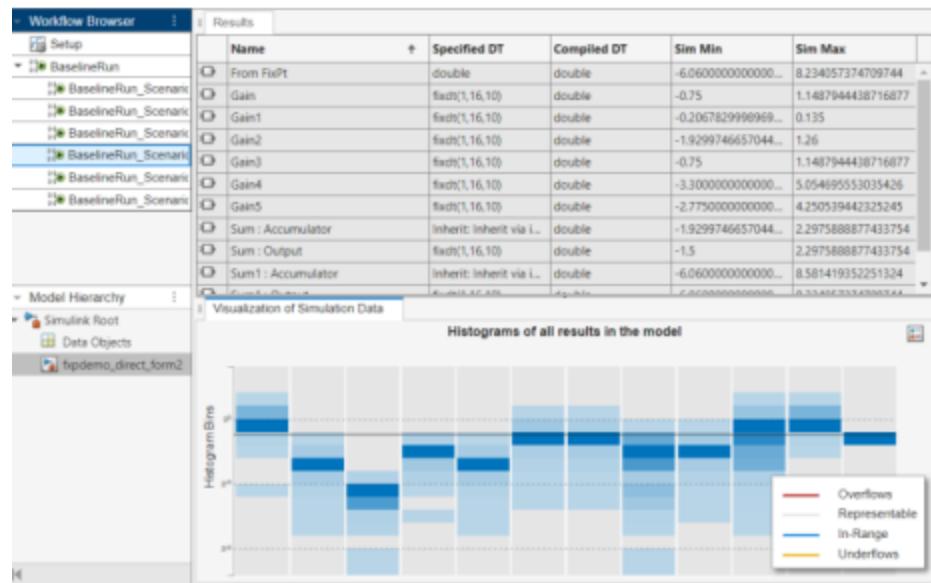
As the input amplitude increases, the amount of underflow is expected to decrease, which is evident in the chart as well. At the stage where the input values reach a moderate level, underflow no longer occurs, and its value becomes zero. Additionally, if the input increases excessively, the system is expected to experience overflow, which is also observed in the graphs.

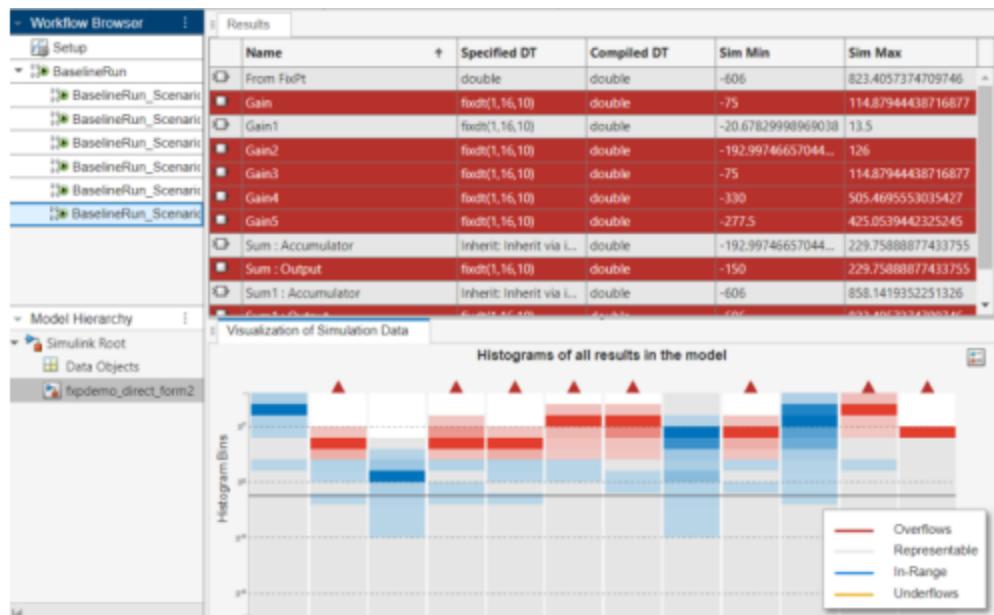


The charts for scenarios 1 to 6 individually are as follows:



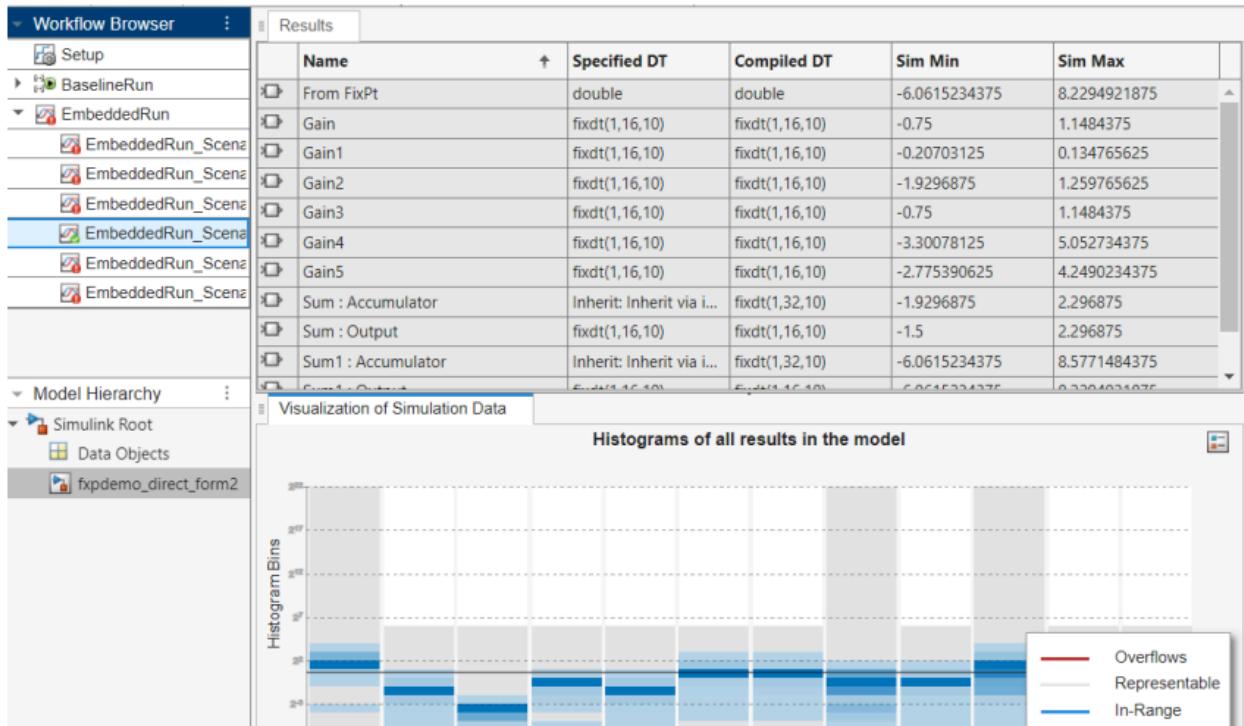


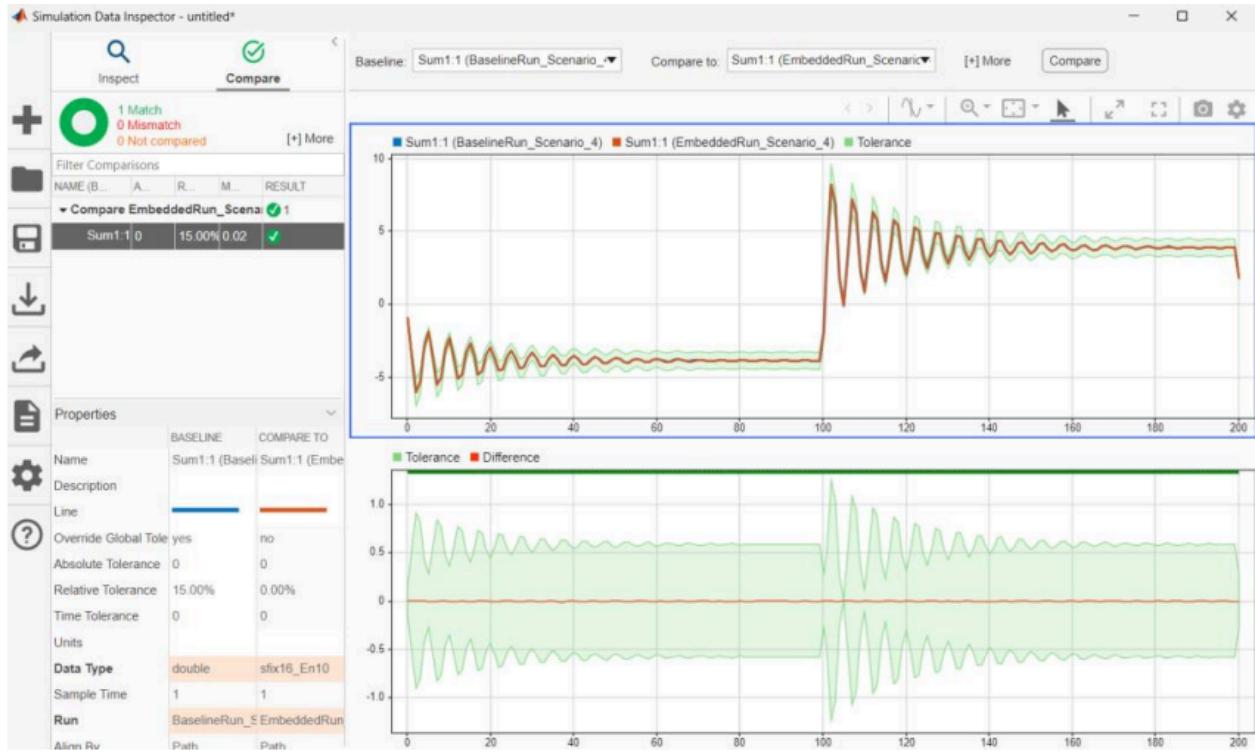




4. Displayed Screen Image:

As can be seen, only scenario number 4 has produced the desired response, which is as follows:



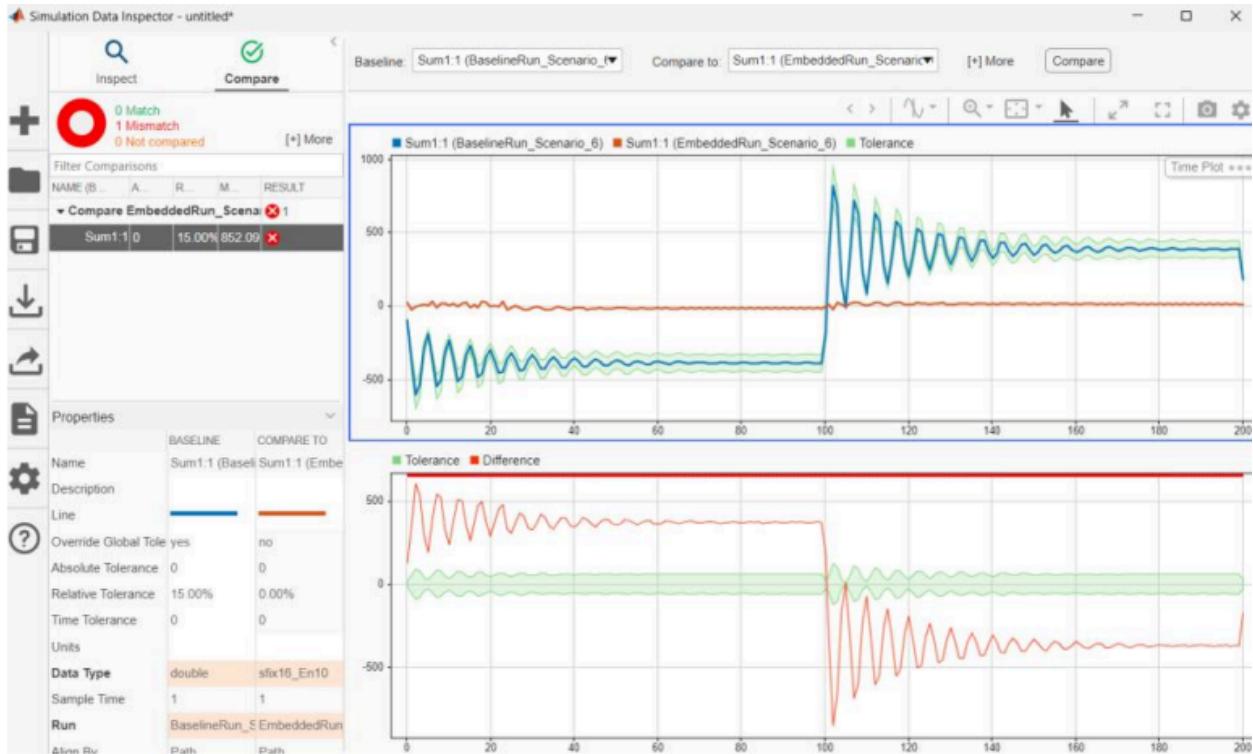


5. Step 4 for Scenario 1 and Scenario 6:



Scenario 1:

In this case, the output has experienced underflow, which is why, as shown in the image, the output is not placed in the desired and displayable section

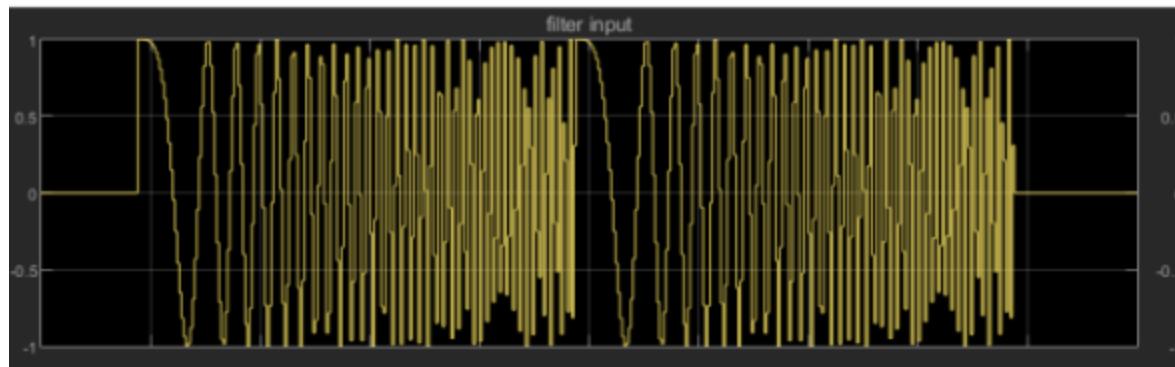


:Scenario 6

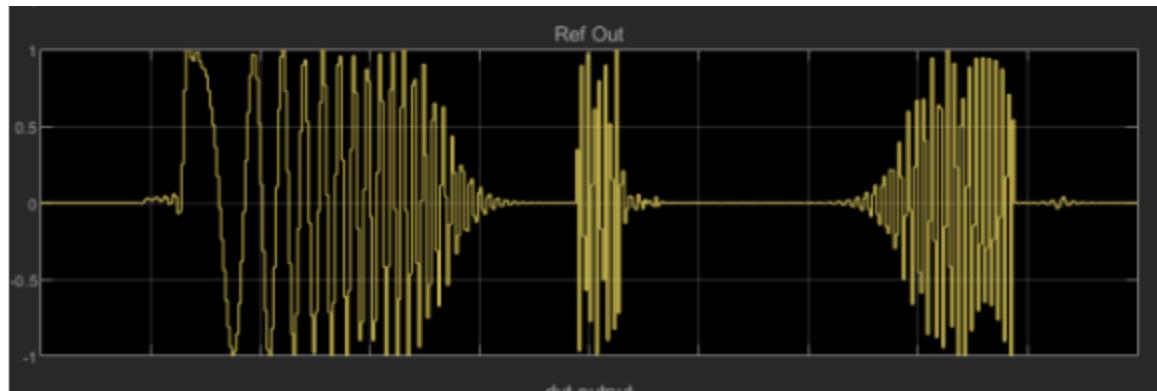
In this case, the system has experienced overflow, which is why, as shown in the image, the output is not placed in the desired and displayable section

Part2:

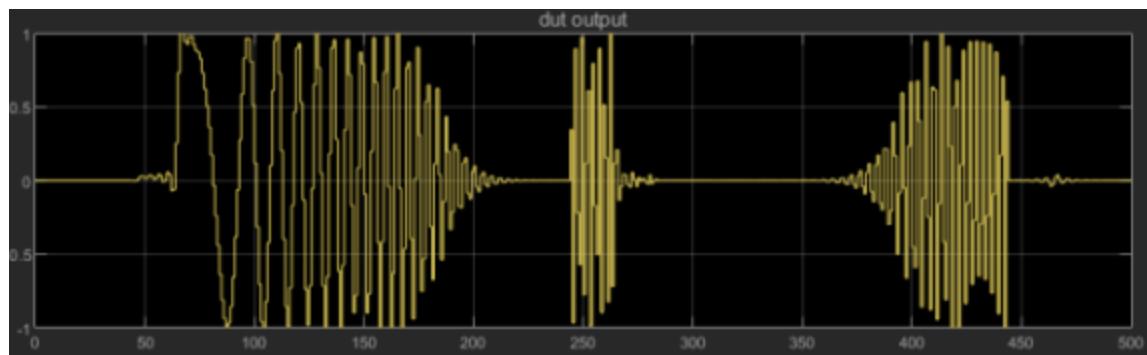
1.



This shows the input signal



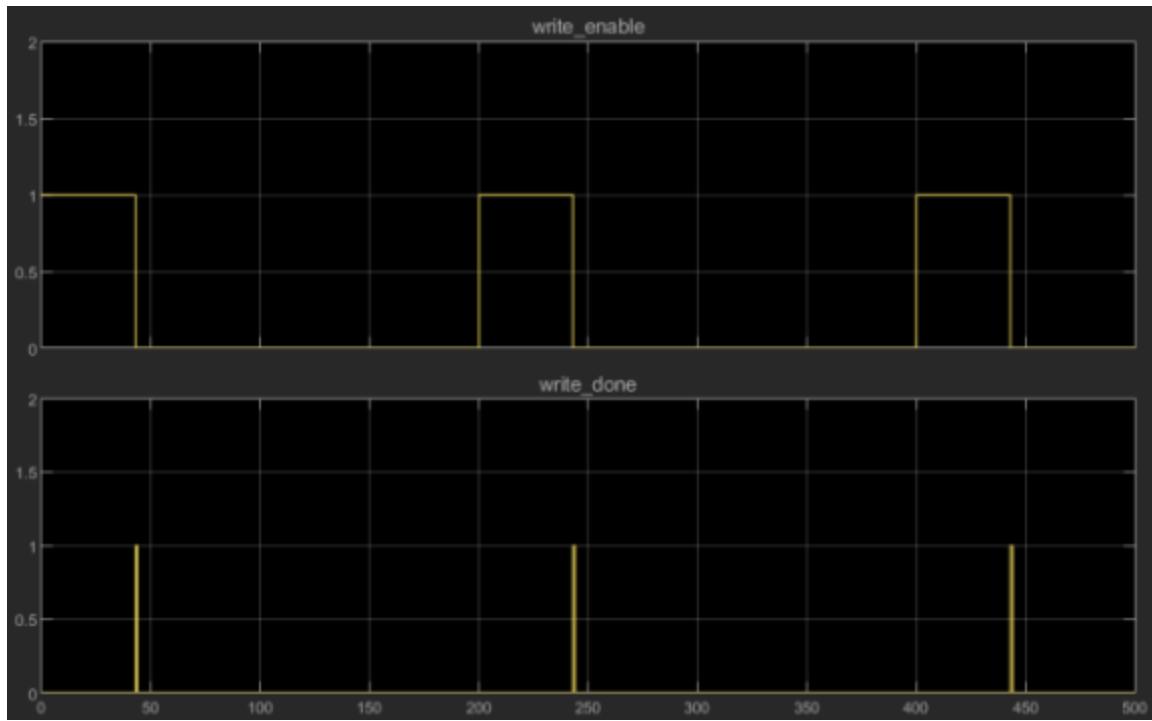
This shows the ideal filter output



This is the output signal of the circuit



Error is the difference between Ref out and dut out



These two are control signals of the circuit

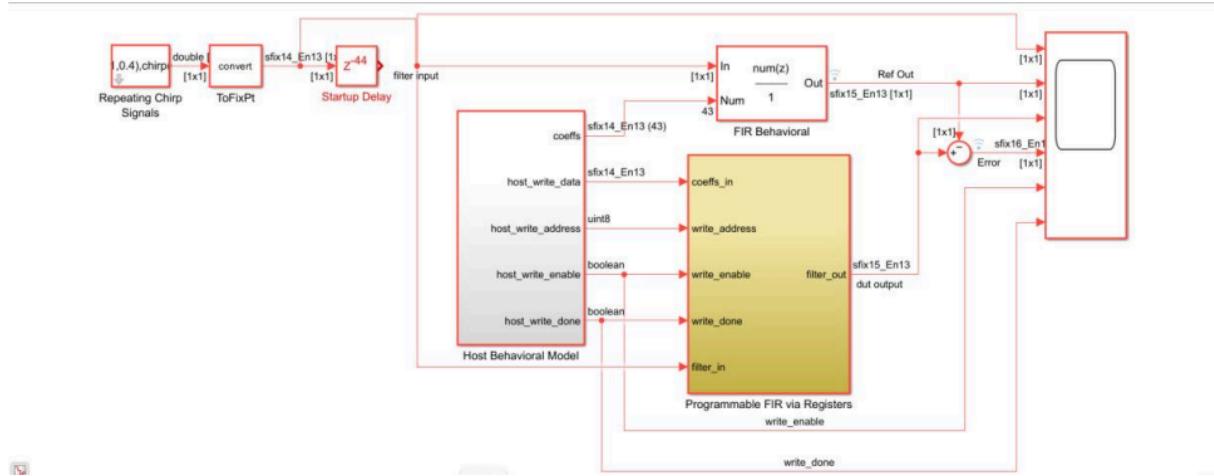
2.

Name	Value
Apass	1
Astop	60
dspprogfirhdl...	1x1 Dataset
f	1x1 lowpass
Fpass	0.4500
Fstop	0.5500
Hhp	1x1 dffir
Hlp	1x1 dffir
NTaps	43
Numerator1	1x43 double
Numerator2	1x43 double
T	200

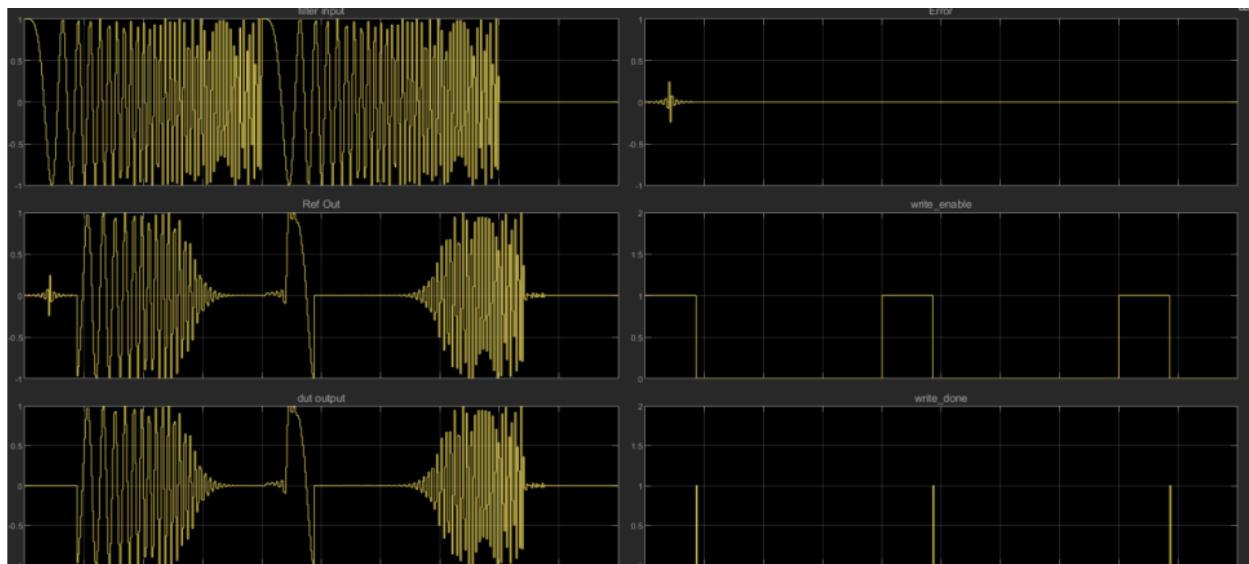
We can assume that the filter has 43 coeffs based on Numerator 1 and Numerator 2.

3:

The circuit if we bypass delay:



the output waveform:



The change in this case, compared to the scenario where there is a delay unit, is the non-zero error at the beginning of the simulation. This is due to the unstable and incorrect output of the block. The reason for this is that both blocks require full loading of all coefficients. The delay unit used introduces a 44-cycle delay in the start of the operation to ensure the coefficients have enough time to load. In the implemented block in the system, using registers and considering that we have two control signals, the output is not generated until the signal becomes equal to 1.

Therefore, we don't have an incorrect output, and the error that occurs is because the output is incorrect.

4:

```
1 Fpass = 0.45; % Passband Frequency
2 Fstop = 0.55; % Stopband Frequency
3 Apass = 1;      % Passband Attenuation (dB)
4 Astop = 60;    % Stopband Attenuation (dB)
5
6 f = fdesign.lowpass('Fp,Fst,Ap,Ast',Fpass,Fstop,Apas
7 Hlp = design(f, 'equiripple','FilterStructure', 'dffir'); % Lowpass
8
9 Hhp = firlp2hp(Hlp); % Highpass
10
11 Numerator1 = Hlp.Numerator; % coefficients of lowpass filter
12 Numerator2 = Hhp.Numerator; % coefficients of highpass filter
13
14 NTaps = length(Numerator1); % number of coefficients
15
16 T = 200; % number of test samples for each set of coefficients
```

In digital signal processing, when designing filters, it is common practice to normalize the frequencies in relation to the Nyquist frequency, which is half the sampling frequency (**Fs/2**). This ensures that the filter design remains consistent regardless of the specific sampling rate used in the system. In this study, we are working with the human hearing range, with a passband frequency (**Fpass**) of 80 Hz and a stopband frequency (**Fstop**) of 8 kHz.

Step-by-Step Frequency Normalization

1. Define the Sampling Frequency (**Fs**):

The sampling frequency for this system is assumed to be **44,100 Hz**, which is standard in many audio applications (e.g., CD-quality audio). This corresponds to a Nyquist frequency of:

Passband and Stopband Frequencies:

- The **passband frequency (Fpass)** is set at **80 Hz**, which is the minimum frequency to pass through the filter.

- The **stopband frequency (Fstop)** is set at **8,000 Hz (8 kHz)**, which is the maximum frequency to be attenuated by the filter.

Normalization of Frequencies:

To normalize the passband and stopband frequencies, we divide each frequency by the Nyquist frequency

These normalized values represent the passband and stopband frequencies in terms of the Nyquist frequency. In a digital filter design, these normalized frequencies are used as inputs to algorithms that calculate filter coefficients.

Importance of Normalization:

Normalization is crucial in digital filter design as it allows the filter's behavior to remain consistent, regardless of the sampling rate. It ensures that the filter's performance is based on relative frequencies, making it easier to apply the same filter design principles across different systems with varying sampling rates. This process also avoids issues that can arise from frequency aliasing and ensures accurate representation of the desired frequency ranges.

Python code to automate the process:

```

1  # Define the sampling frequency
2  Fs = 44100  # in Hz
3
4  # Define the passband and stopband frequencies
5  Fpass = 80  # in Hz
6  Fstop = 8000  # in Hz
7
8  # Calculate the Nyquist frequency (Fs/2)
9  Nyquist = Fs / 2
10
11 # Normalize the frequencies
12 Fpass_normalized = Fpass / Nyquist
13 Fstop_normalized = Fstop / Nyquist
14
15 # Print the normalized frequencies
16 print("Fpass (normalized):", Fpass_normalized)
17 print("Fstop (normalized):", Fstop_normalized)
18

```

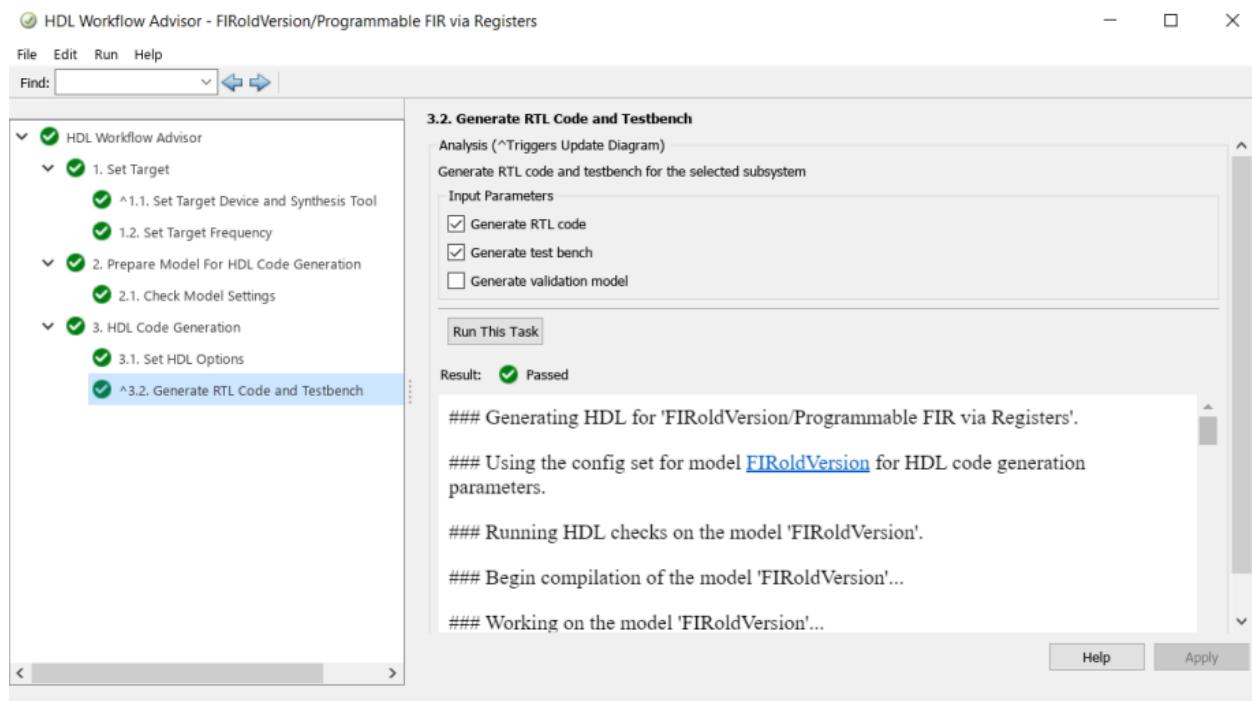
```
Fpass (normalized): 0.003627122174461308 Fstop (normalized):
0.3634251097954708
```

Updated MATLAB code:

```
1 Fpass = 0.0036; % Passband Frequency %80/(44100/2)
2 Fstop = 0.36; % Stopband Frequency %8000/(44100/2)
3 Apass = 1; % Passband Attenuation (dB)
4 Astop = 60; % Stopband Attenuation (dB)
5
6 f = fdesign.lowpass('Fp,Fst,Ap,Ast',Fpass,Fstop,Apas
7 Hlp = design(f, 'equiripple','FilterStructure', 'dffir'); % Lowpass
8
9 Hhp = firlp2hp(Hlp); % Highpass
0
1 Numerator1 = Hlp.Numerator; % coefficients of lowpass filter
2 Numerator2 = Hhp.Numerator; % coefficients of highpass filter
3
4 NTaps = length(Numerator1); % number of coefficients
5
6 T = 200; % number of test samples for each set of coefficients
```

5:

Compiled successfully:



resources:

Generic Resource Report for FIRoldVersion	
Summary	
Multipliers	43
Adders/Subtractors	42
Registers	128
Total 1-Bit Registers	1792
RAMs	0
Multiplexers	127
I/O Bits	57
Static Shift operators	0
Dynamic Shift operators	0

Resources in detail:

Report for Subsystem: [Programmable FIR via Registers](#)

Multipliers (43)

14x14-bit Multiply : 43

Adders/Subtractors (42)

31x31-bit Adder : 1
32x32-bit Adder : 41

Registers (85)

[+] 14-bit Register : 85

Multiplexers (84)

29-bit 3-to-1 Multiplexer : 43
31-bit 3-to-1 Multiplexer : 41

Number of I/O Bits (57)

[+] Number of Input Bits: 41
[+] Number of Output Bits: 16

Report for Subsystem: [coeffs_registers](#)

Registers (43)

14-bit Register : 43

Multiplexers (43)

14-bit 2-to-1 Multiplexer : 43

Critical path:

Critical Path Details			
Id	Propagation (ns)	Delay (ns)	Block Path
1	64.3590	64.3590	Discrete FIR Filter

Input ports:

Port Name	Datatype	Bits
clk	boolean	1
reset	boolean	1
clk_enable	boolean	1
coeffs_in	sfix14_En13	14
write_address	uint8	8
write_enable	boolean	1
write_done	boolean	1
filter_in	sfix14_En13	14

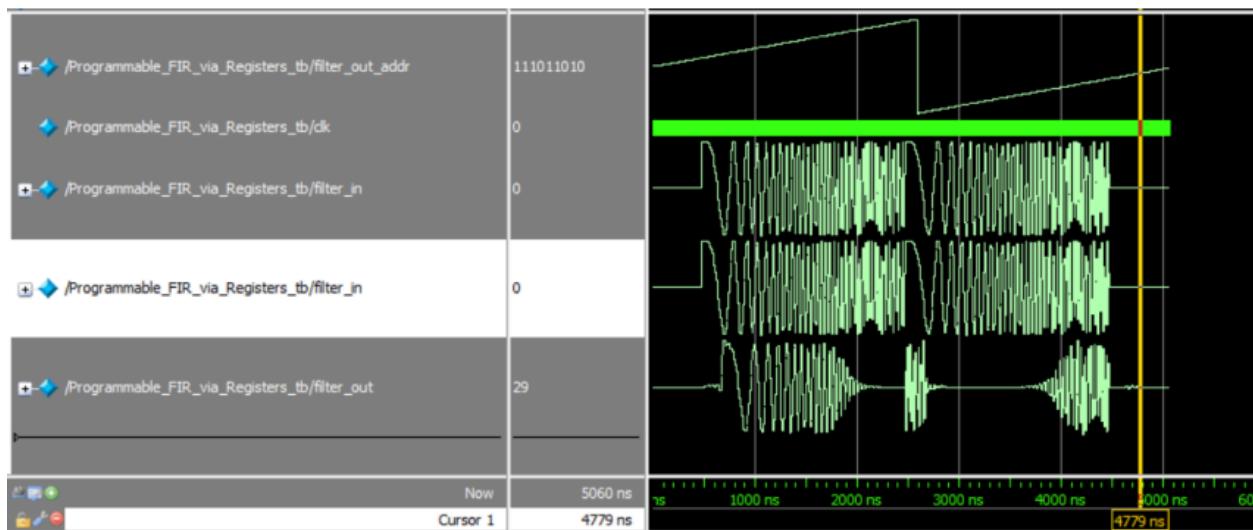
Output ports:

Port Name	Datatype	Bits
ce_out	boolean	1
filter_out	sfix15_En13	15

Modelsim verification:

```
# Compile of coeffs_registers.v was successful.
# Compile of Discrete_FIR_Filter.v was successful.
# Compile of Programmable_FIR_via_Registers.v was successful.
# Compile of Programmable_FIR_via_Registers_tb.v was successful.
# 4 compiles, 0 failed with no errors.
```

Output waveform:



As we expected the output is filtered and it matches the simulink output.