Johns Hopkins Engineering for Professionals 525.742 SoC FPGA Design Lab

Lecture 4 - Filtering

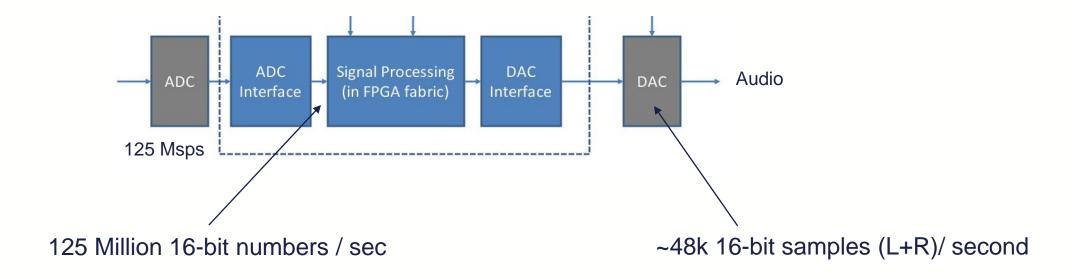


Agenda

- Lab 4 Topics
 - Channel Selection
 - Finite Impulse Response Filters
 - Design (Matlab)
 - FPGA Implementation
 - Computational requirements
 - Fixed point vs floating point issues
 - Modelling of FPGA filter in Matlab
- Demonstrate Lab 4



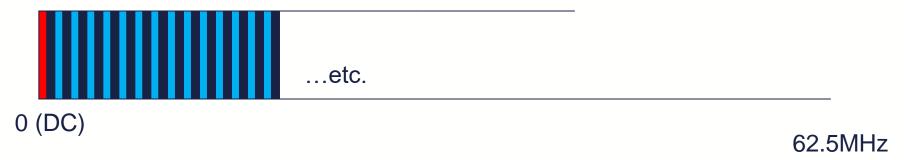
In Progress: Our Radio Receiver Core





Lab 4: Filter

For lab 4, we will presume that our signal of interest is located at the frequencies 0-18kHz, we will design a suitable filter, simulate it and demonstrate it using the framework we started in Lab 4

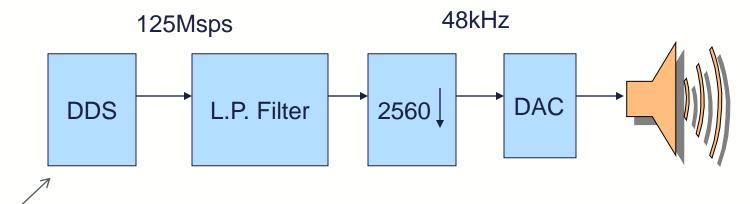


- Filter Task
 - Keep 0-18kHz relatively unchanged
 - Remove (suitably attenuate) all other frequencies from the signal in preparation for reducing the sample rate
 - New sample rate will be approx 48.8kHz



Lab 4: Filter Demonstration

 Implement the channel selection filter which will be used in the SDR.

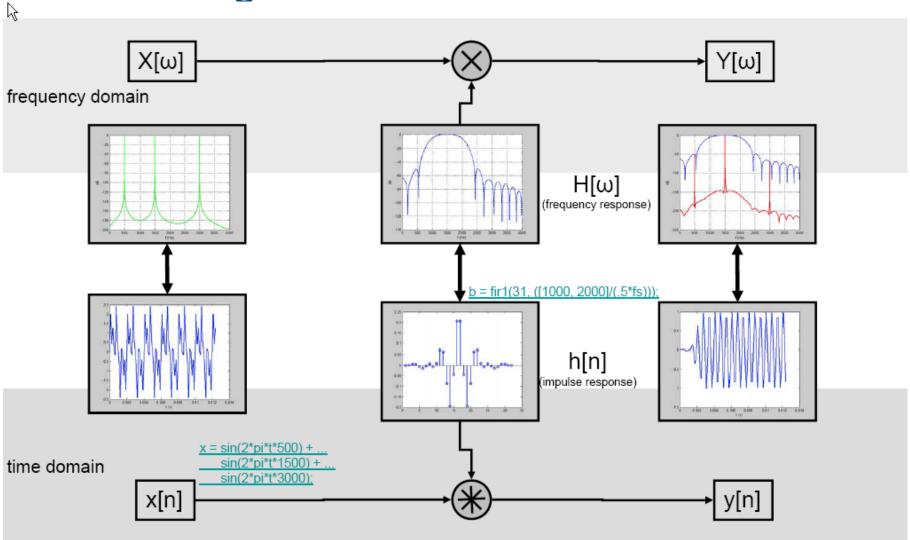


the DDS is a convenient simulator for signals that will eventually be received from an A/D converter

Lab 4 is basically **exactly** like Lab 3, except without the aliasing!

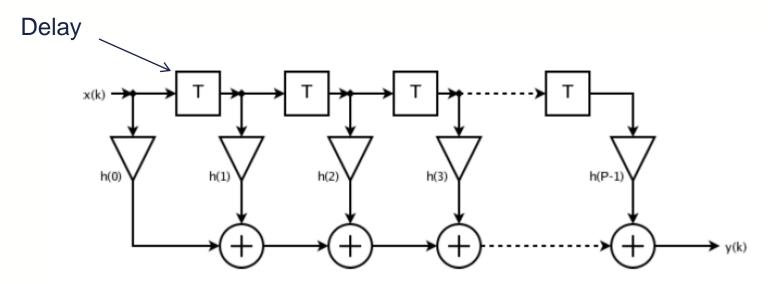


FIR Filtering





FIR Implementation



■ Convolution in time domain is computationally taxing, yet fairly straightforward.

$$\begin{split} (f*g)[n] &\stackrel{\mathrm{def}}{=} \sum_{m=-\infty}^{\infty} f[m] \, g[n-m] \\ &= \sum_{m=-\infty}^{\infty} f[n-m] \, g[m]. \quad \text{(commutativity)} \end{split}$$

Example Convolution

```
Filter length (N) = 3
h = [.2.6.2]
Input data = [0.20.1.20.0.20...]
```

```
>> h = [.2 .6 .2];
>> x = [ 0 20 -1 -20 0 20];
>> y = filter(h,[1],x) % [1] for FIR filter
```

```
y[0] = x[0] * h[0] = .2*0 = 0

y[1] = x[1] * h[0] + x[0]*h[1] = 20*.2

y[2] = x[2]*h[0]+x[1]*h[1]+x[0]*h[2] =

y[3] = x[3]h[0]+x[2]h[1]+x[1]h[2] =

y[4] = x[4]h[0]+x[3]h[1]+x[2]h[2] =

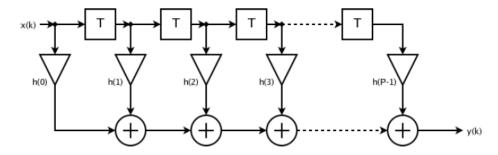
....
```

[0 4.0000 11.8000 -0.6000 -12.2000 0]



Example Convolution

```
Filter length (N) = 3
h = [ .2 .6 .2]
Input data = [ 20 20 20 20 20 20 ... ]
```



```
>> h = [.2 .6 .2];
>> x = [ 20 20 20 20 20 20];
>> y = filter(h,[1],x) % [1] for FIR filter
```

```
y[0] = x[0] * h[0] = 4

y[1] = x[1] * h[0] + x[0]*h[1] = 16

y[2] = x[2]h[0]+x[1]h[1]+x[0]h[2] = 20

y[3] = x[3]h[0]+x[2]h[1]+x[1]h[2] = 20

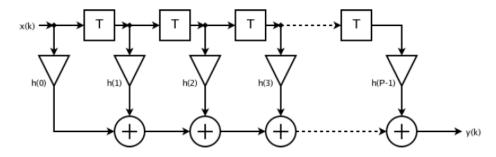
y[4] = x[4]h[0]+x[3]h[1]+x[2]h[2] = 20

....
```



Example Convolution

```
Filter length (N) = 3
h = [ .2 .6 .2]
Input data = [ 0 0 1 0 0 ]
```



```
>> h = [.2 .6 .2];
>> x = [ 0 0 1 0 0];
>> y = filter(h,[1],x) % [1] for FIR filter
```

```
y[0] = x[0] * h[0] = 0

y[1] = x[1] * h[0] + x[0]*h[1] = 0

y[2] = x[2]h[0]+x[1]h[1]+x[0]h[2] = .2

y[3] = x[3]h[0]+x[2]h[1]+x[1]h[2] = .6

y[4] = x[4]h[0]+x[3]h[1]+x[2]h[2] = .2

....
```

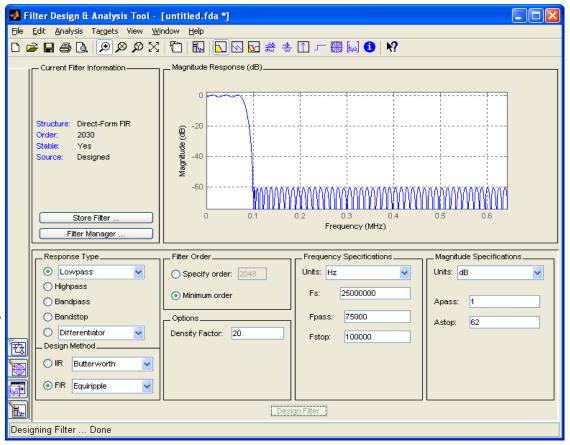
[00.2.6.2]



Example Filter Design Process

Please note that these are not the requirements for Lab 4!, this is an exploration of the design process and case study for design criteria

- Filter Specs:
 - Fs = 25 Msps
 - 75kHz passband
 - 100k stopband
 - Attenuate stopband signals by > 60dB
- FIR Filter with <2048
 <p>coefficients will meet this requirement



Matlab: "fdatool" (goes from specs to h[n])
Instructor provided code will put these in a special Xilinx format

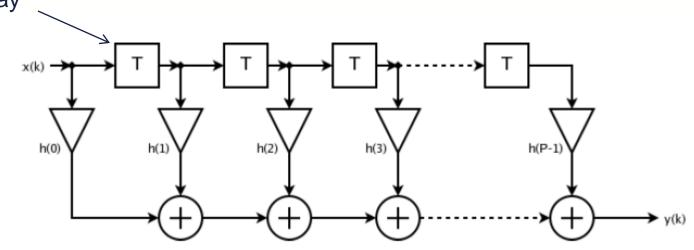


Questions

- Computational requirements :
 - Can our FPGA do the required computation?
 - How much space will it take?
 Multipliers, slices, RAM?
 - How fast must the clock be?
- Numerical accuracy
 - What will be the accuracy differences in our FPGA implementation due to fixed point issues?



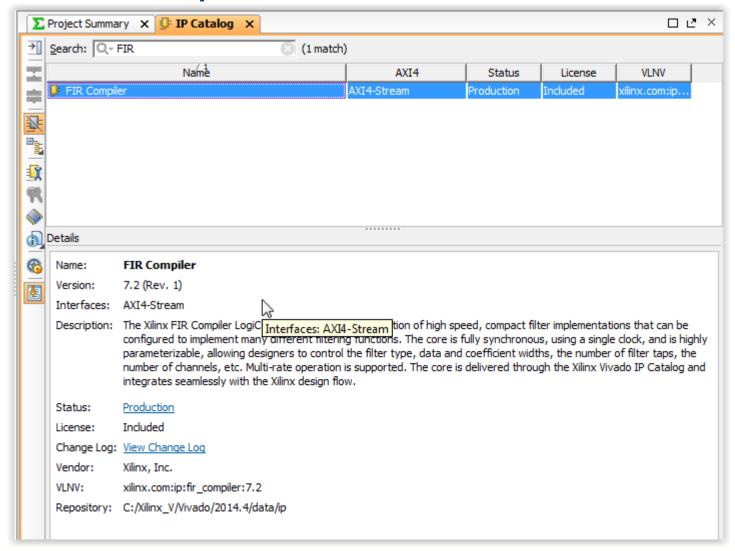
FIR Implementation



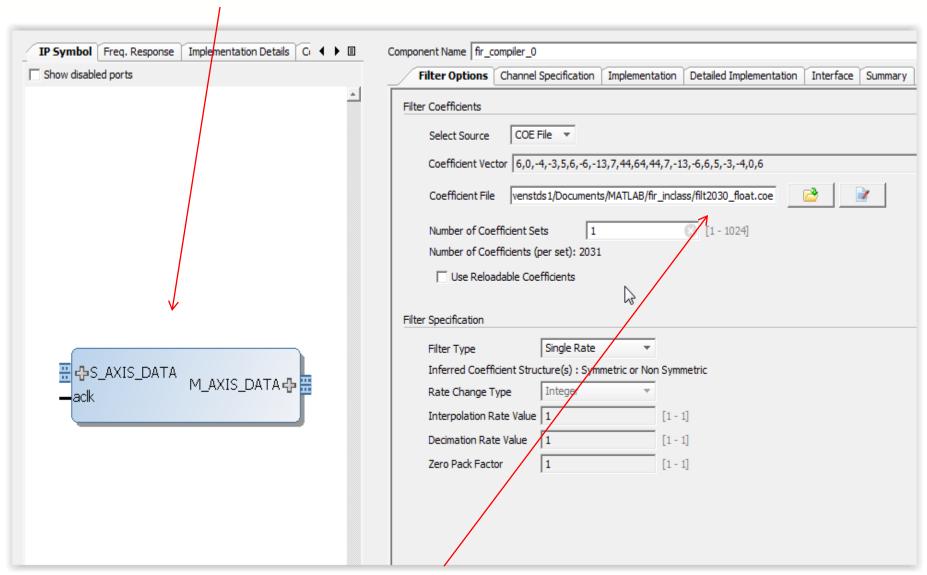
- How many Multiply-Accumulate operations (MAC) are required per sample to implement the filter we designed?
- What is the overall rate of MACs per second?



FIR Compiler

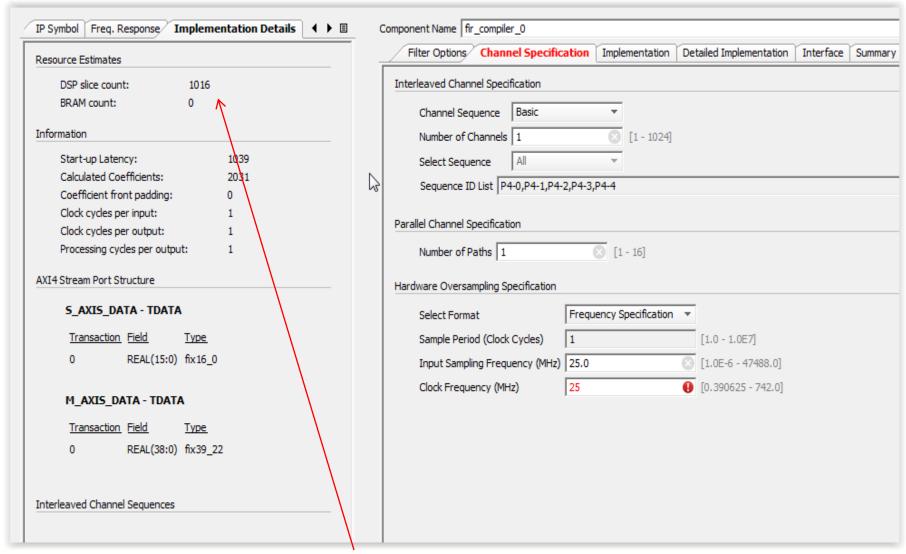


Familiar AXI-Stream interface for in/out



Tool takes as input the "COE" file (which is just a list of coefficients) to get h[n]





Implementation details show what we computed previously with a savings Due to symmetrical structure



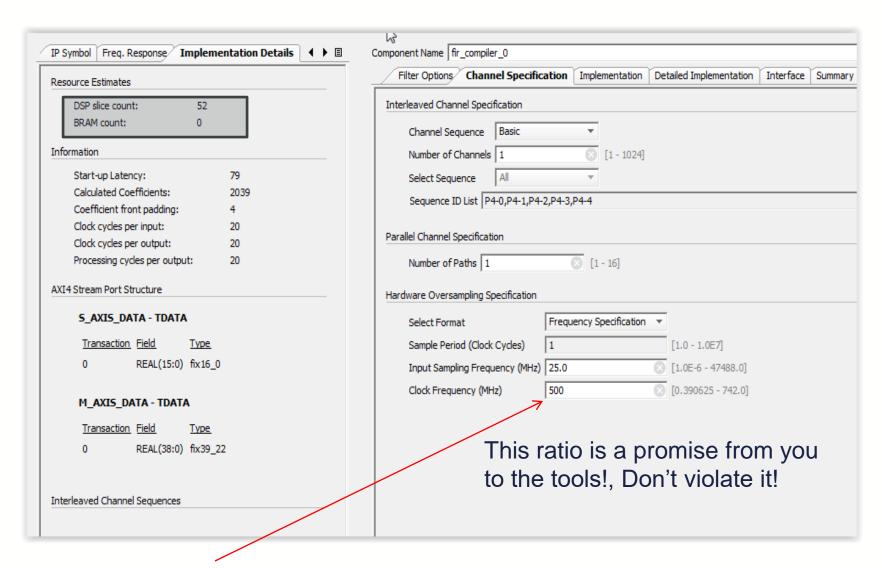
DSP in Xilinx FPGAs

2016 V6 DSP Slices @ 600MHz fmax

Features	Artix-7	Kintex-7	Virtex-7	Spartan-6	Virtex-6
Logic Cells	215,000	480,000	2,000,000	150,000	760,000
BlockRAM	13Mb	34Mb	68Mb	4.8Mb	38Mb
DSP Slices	740	1,920	3,600	180	2,016
DSP Performance (symmetric FIR)	930GMACs	2,845GMACs	5,335GMACs	140GMACs	2,419GMACs

180 S6 DSP Slices @ 390MHz fmax

Zynq 7Z020 : 220 DSP Slices, 276GMACs



If we let filter run at higher clock rate than input sample rate, structure Is automatically adapted such that a convolution takes multiple clocks using Shared multipliers



Zynq DSP Slice (>600MHz)

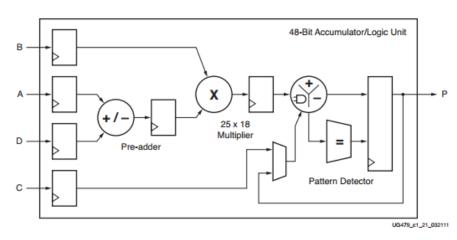
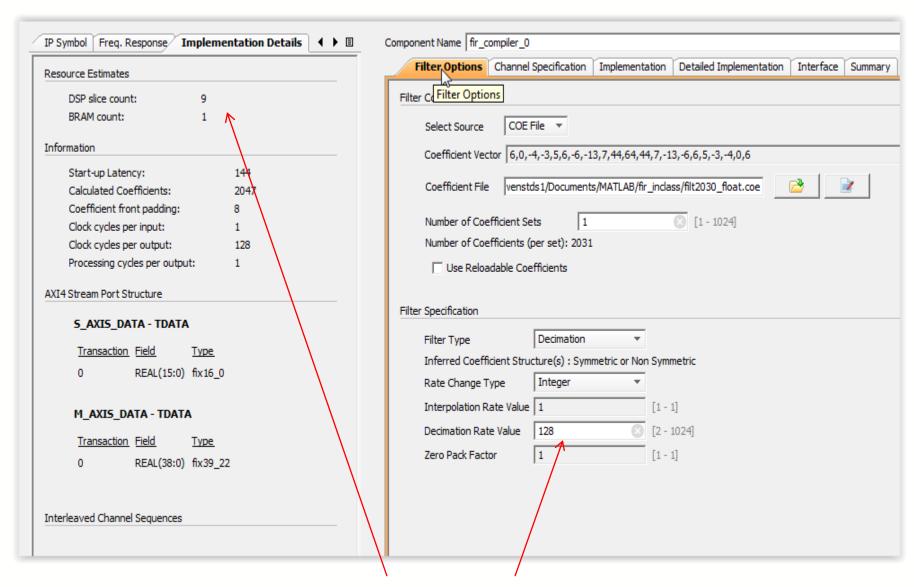


Figure 1-1: Basic DSP48E1 Slice Functionality

Some highlights of the DSP functionality include:

- 25 × 18 two's-complement multiplier:
 - Dynamic bypass
- 48-bit accumulator:
 - · Can be used as a synchronous up/down counter
- Power saving pre-adder:
 - · Optimizes symmetrical filter applications and reduces DSP slice requirements

- 220 * 600M > 130GMACs
 - Our FPGA is capable of doing what we want
 - There will be other features of the FPGA that need multipliers...
- These figures should help put FPGA capabilities for DSP in perspective



Allowing the filter to use all the time between *output* samples for the convolution makes the task easily achievable (this is back to the 25MHz clock!)

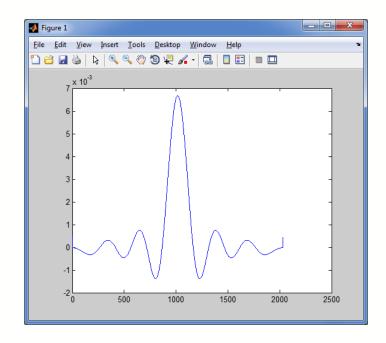


Questions

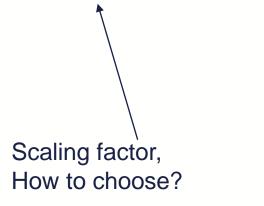
- Computational requirements :
 - Can our FPGA do the required computation?
 - How much space will it take?
 - Multipliers, slices, RAM?
 - How fast must the clock be?
- Numerical accuracy
 - What will be the accuracy differences in our FPGA implementation due to fixed point issues?



Output from FDAtool is floating point coefficients h[NUMTAPS] (aka "impulse response"). In absense of floating point multipliers, this is not directly implementable on our FPGA. We need to transform this impulse response to integers while preserving the function.



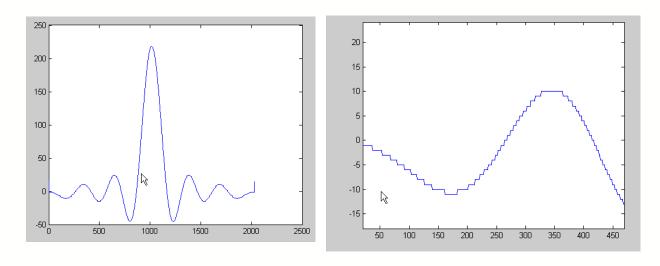
Numint = int32(Num*ScaleFactor+0.5);



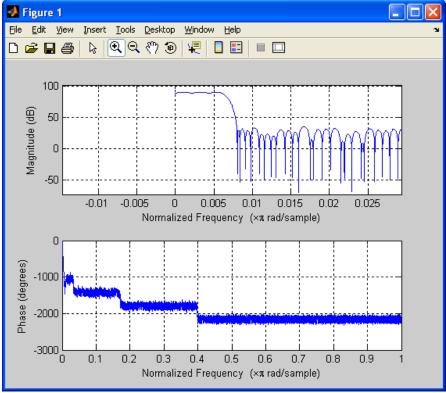
"export" the coefficients to the workspace as "Num"



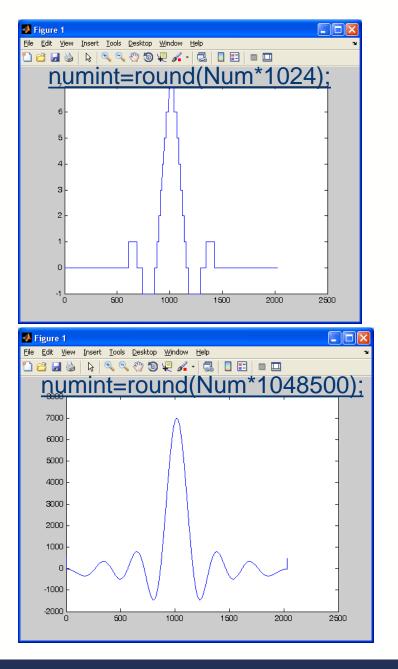
Quantized Coefficients

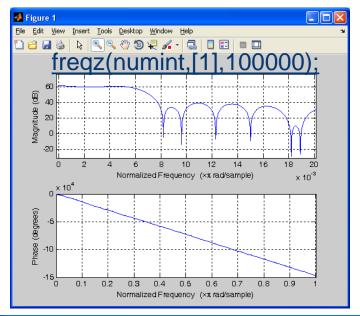


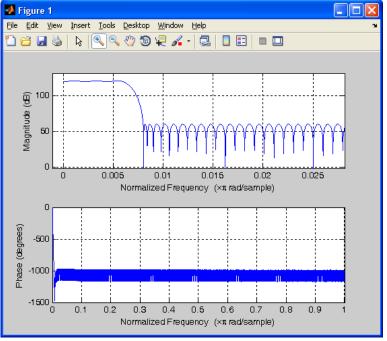
Overall, making the coefficients integers, (after multiplying by 32700) doesn't affect our response too badly. With this scaling factor, our coefficient width is really only 9 bits. Some optimization between coefficient width and filter order could be undertaken if we chose.













Managing Coefficient Scaling

- As we have discussed, the scaling that we apply to the coefficients will affect:
 - Precision of coefficients
 - Frequency response characteristics
 - Filter gain
- There are two methods of handling the transition from Matlab -> Vivado
 - Manually: we explicitly scale coefficients in matlab
 - IP customizer driven



```
Numint = int32(Num*32700+0.5);
fid=fopen('filt2030.coe','w'); \frintf(fid,'radix=10;\ncoefdata=\n');
fprintf(fid,'%d,\n',NumInt);
fclose(fid);
```

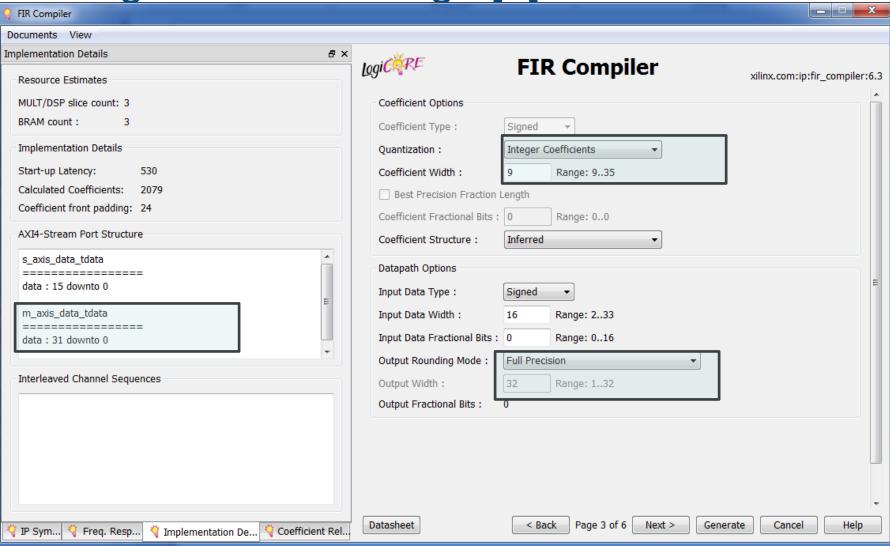
Scaling factor, Choose this for a reasonable

Performance vs. utilization tradeoff

```
radix=10;
coefdata=
15,
-1,
-1,
-1,
-1,
...etc
```

Result is: impulse response of filter in a "coe" file, which we will use later when designing the filter. Note that filter gain has changed though over the unity gain filter we designed in Matlab. *Now, signals in the passband will come out x32700 over the input level.*







- Result is very simple the pure result of convolving our new scaled h[n] with the data.
 - Output will simply be *scale_factor vs. the unity gain filter.
 If we don't want to keep the huge number of bits, how do we reduce this number?
 - Some divisions can be done with shifting (aka picking different bits)



Managing Coefficient Scaling

- As we have discussed, the scaling that we apply to the coefficients will affect:
 - Precision of coefficients
 - Frequency response characteristics
 - Filter gain
- There are two methods of handling the transition from Matlab -> Coregen
 - Manually: we explicitly scale coefficients in matlab
 - Vivado / GUI driven



Writing h[n] to a file

```
>> fid=fopen('filt2030_float.coe','w');
>> fprintf(fid,'radix=10;\ncoefdata=\n');
>> fprintf(fid,'%5.15f,\n',Num);
>> fclose(fid);

-0.000023959423709,
-0.000023496101200,
-0.000023104860244,
-0.000022826569752,
-0.000022504297004,
-0.000022452966158,
...etc.
```

Result is: impulse response of filter in a "coe" file, which we will use later when designing the filter. Note that scaling / quantization of coefficients has not happened yet – this is the raw matlab filter

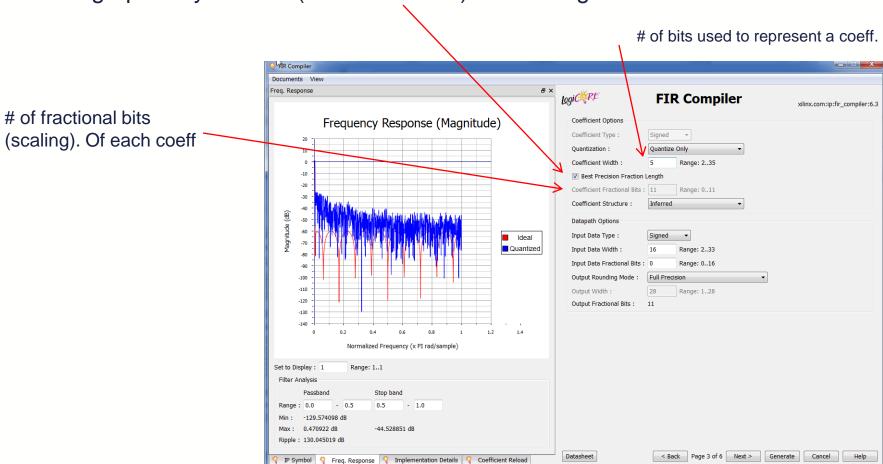


Fixed Point Number Formats

- "Q" notation is used as a standard means of communicating the implicit scaling of a number.
 - QX.Y: This X+Y bit number should be interpreted as having X integer bits and Y fractional bits.
 - In other words, this number is actually your true number multiplied by 2^Y Ex. Q4.4 place values : 8 | 4 | 2 | 1 | .5 | .25 | .125 | .0625
 - Sometimes the X is left out, and the format is just named QY, communicating the number of fractional bits, but not the overall size
- Ex: x"1001" in Q16.0 = 4097. Q14.2 = 1024.2
- Provides a convenient means to track scaling throughout the signal processing chain



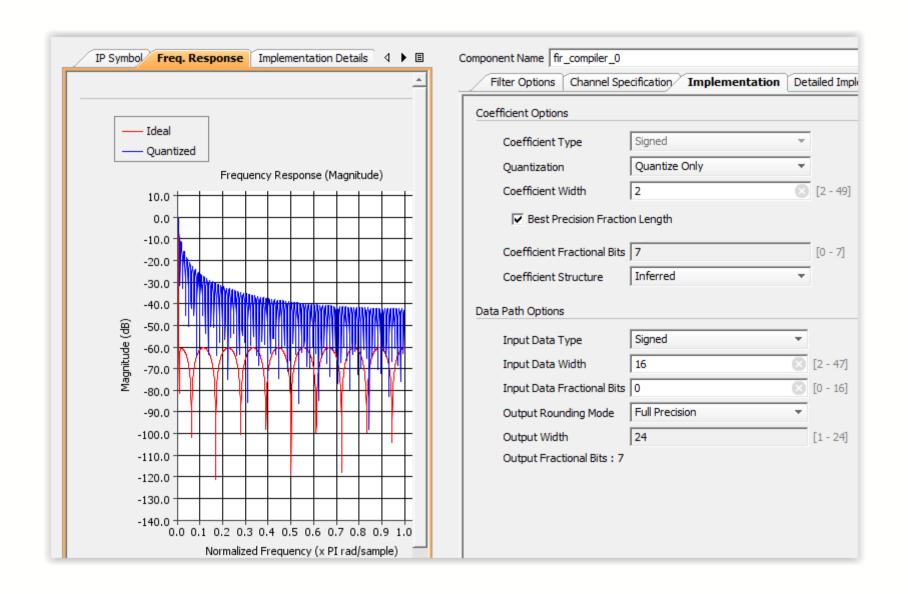
FIR Compiler provides a means for us to input floating point coefficients, and graphically choose (or let it choose) the scaling

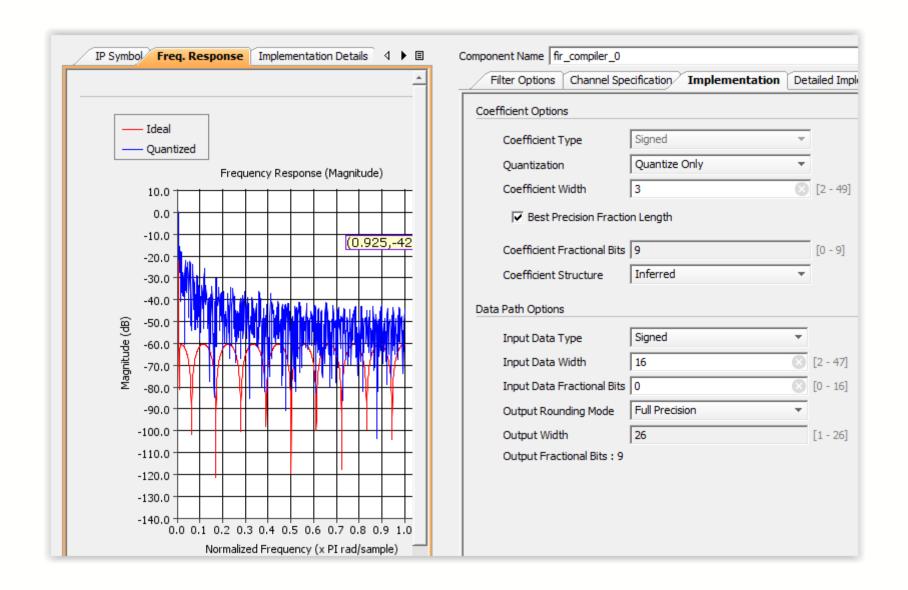




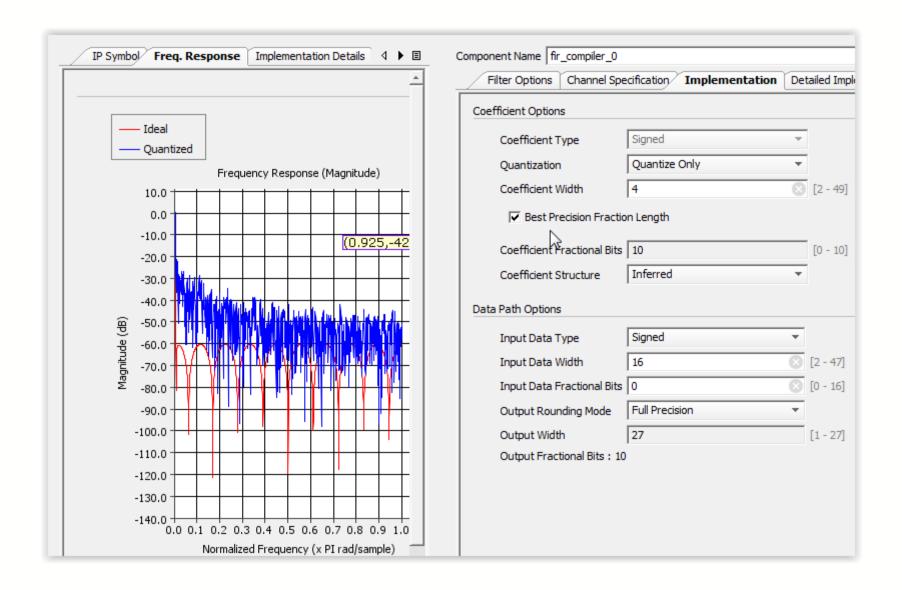
- In this method, scaling is done by IP customizer, which tells you how much it scaled the coefficients by
 - # of fractional bits is shown on the bottom, which tells you how to interpret result. Filter itself has same gain as Matlab (unity) _if_ you interpret the numbers that way
 - Effects of coefficient quantization can be observed easily during design



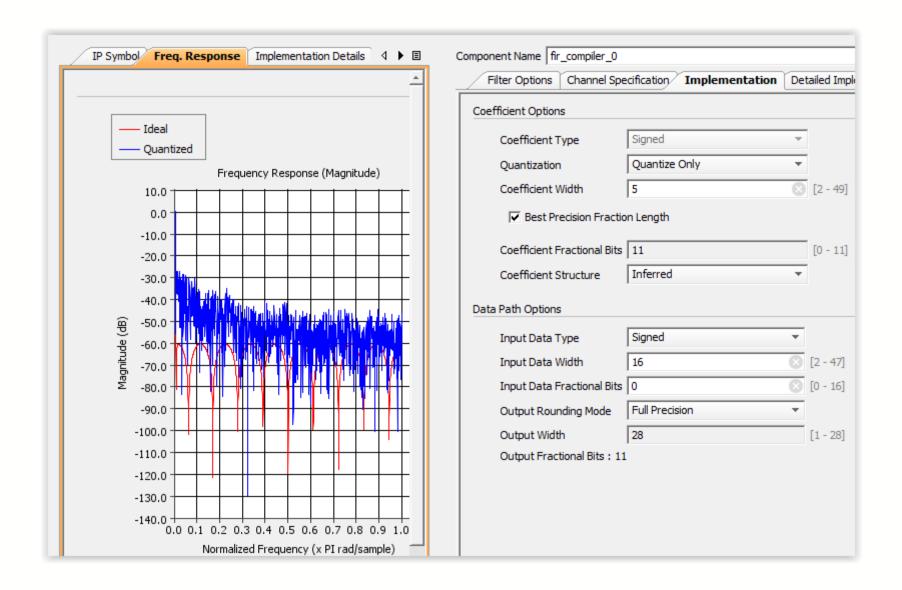




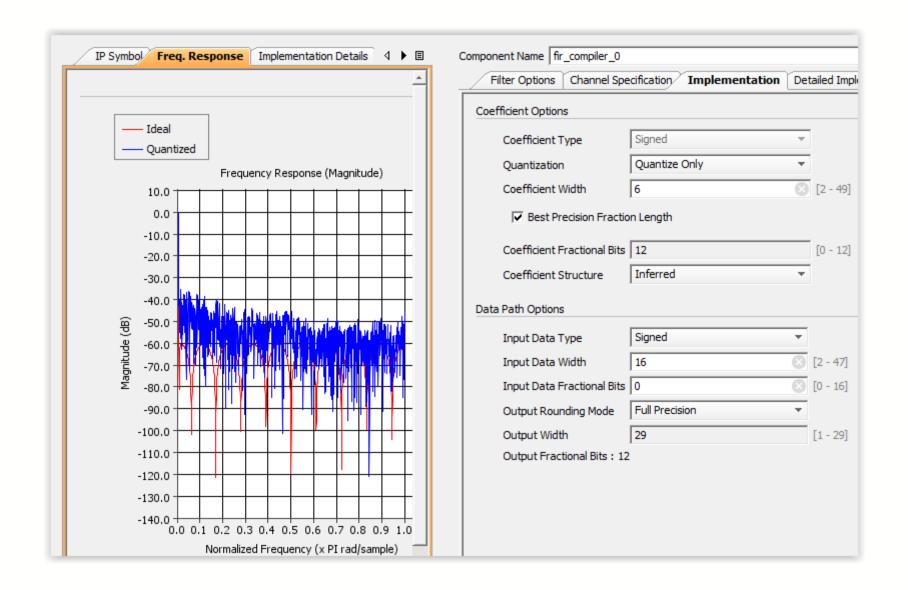




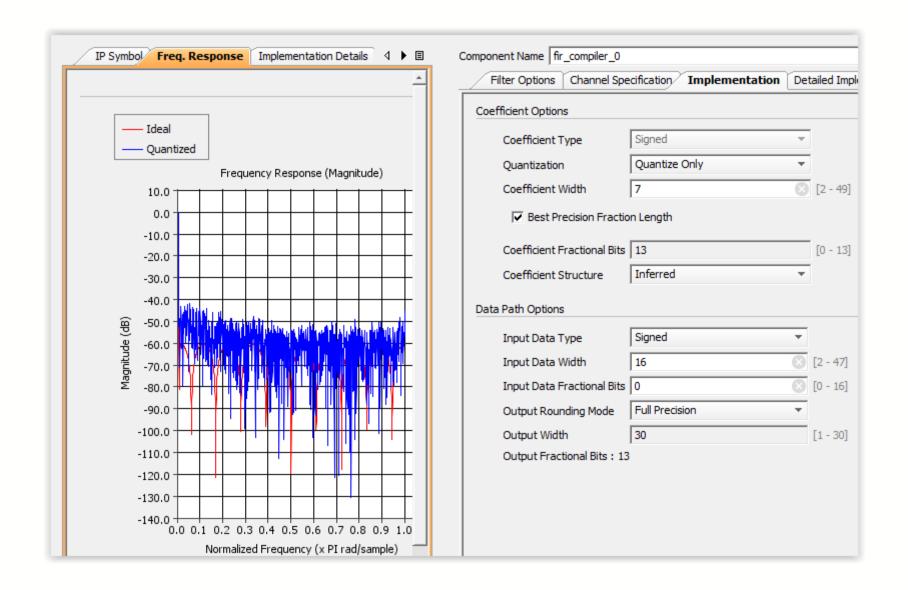




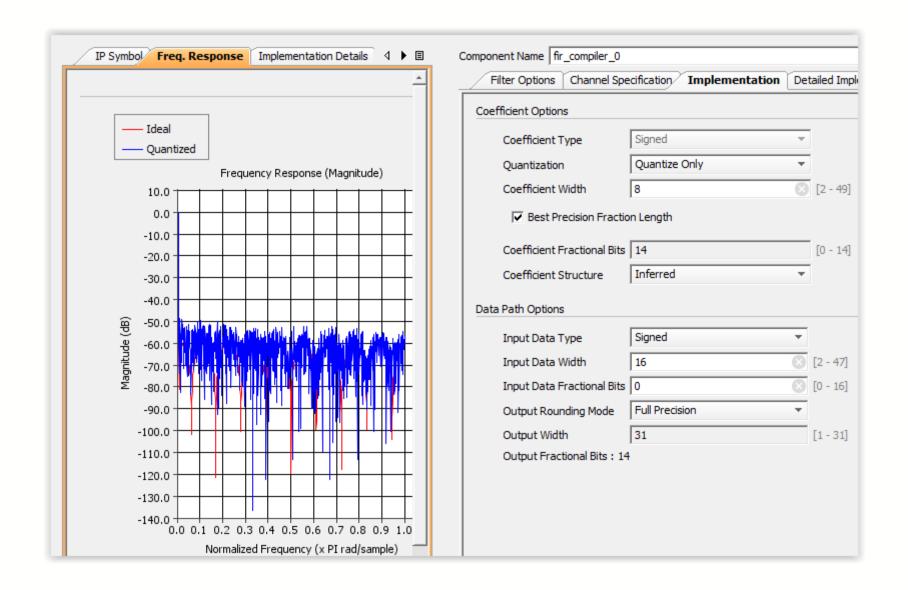




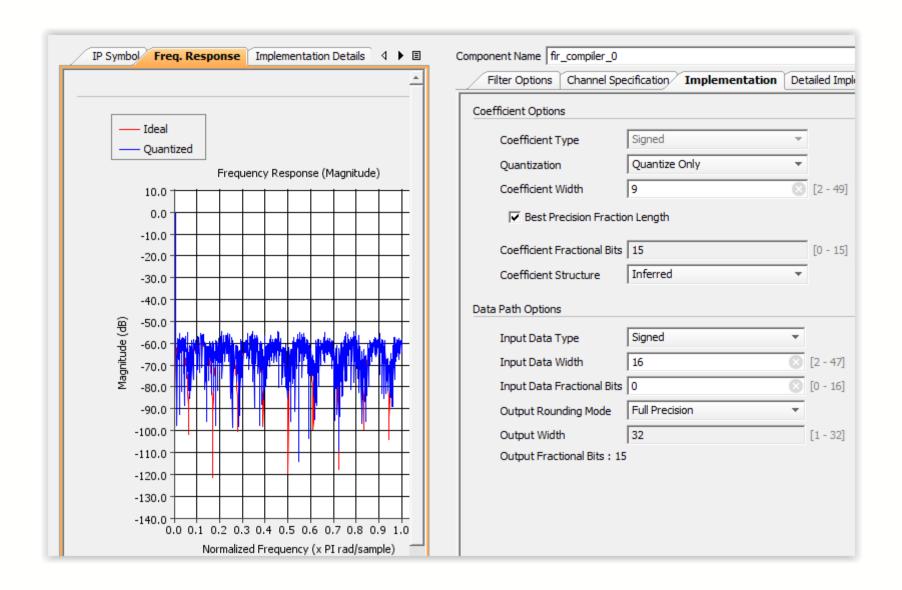




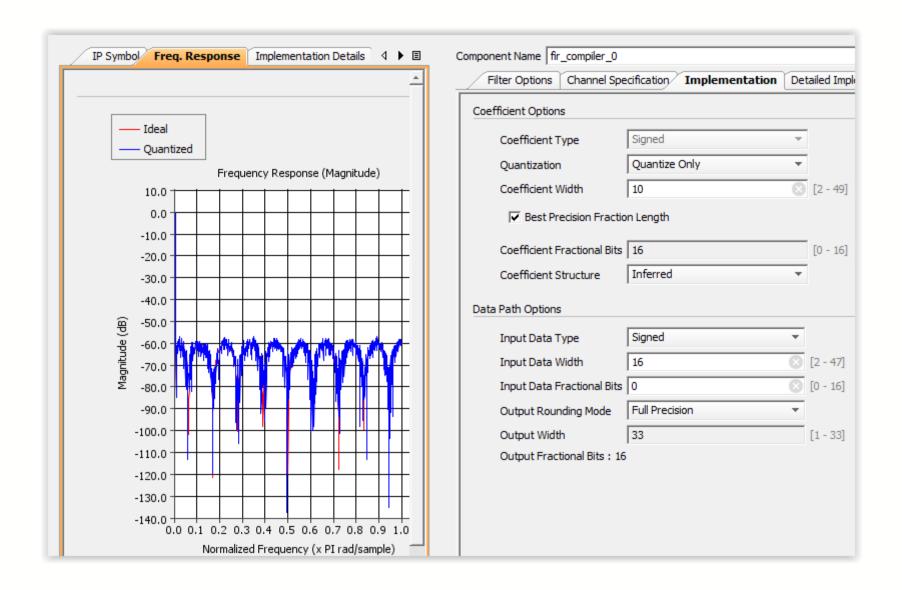




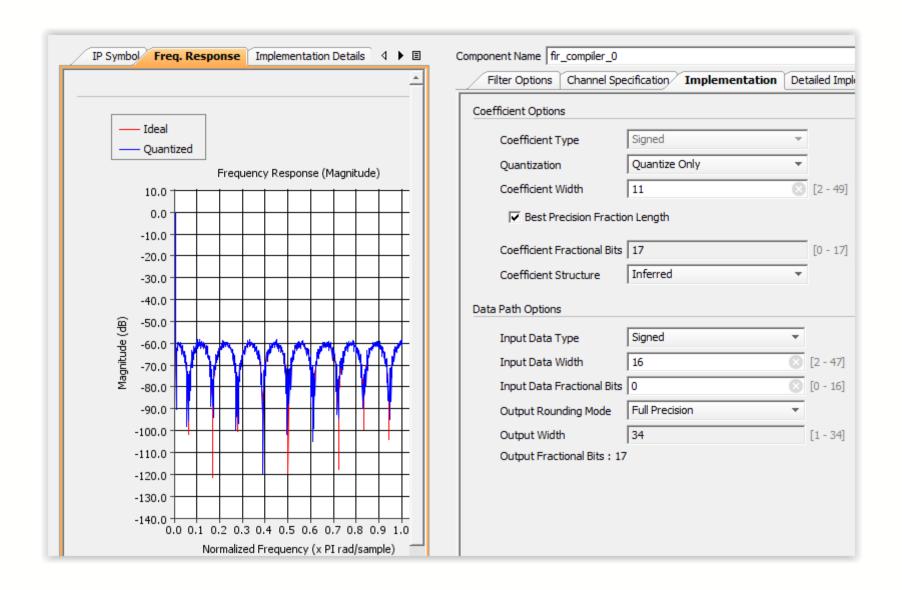




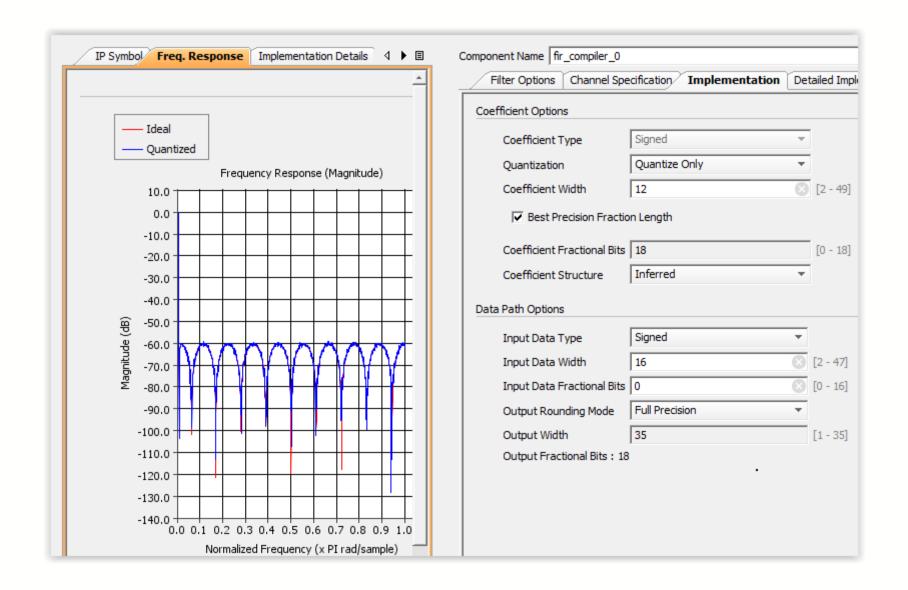




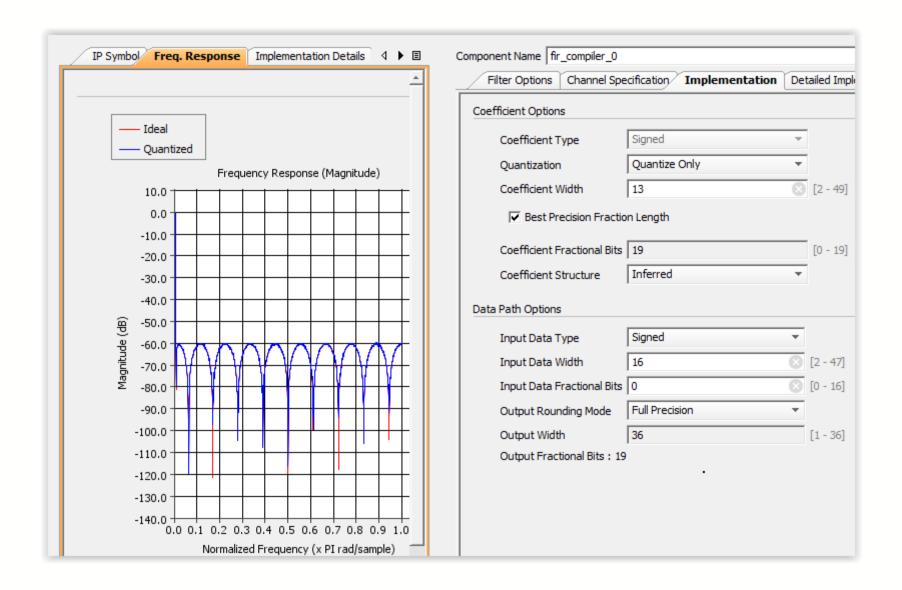














Interpreting Output

- In full precision mode using scaling factor or "number of fractional bits" to make decision on wisest bits to preserve
 - Eventually we want 16-bit outputs, that is what the DAC takes
- Filter core can also "round" for you, you specify output width and it will oblige
 - Make sure you understand result numerically

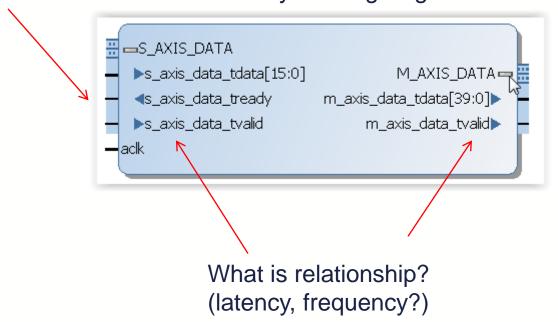
Output will be scaled such that maximum value out of the filter can never overflow

of fractional bits in output is shown. NOTE: it will never go lower than 0.



Decimating FIR Filter Core ("AXI-Stream Interface")

When filter is busy computing this will be low. If you've kept your previous "promise", this will never be low when you are giving it a new sample





Lab Details

- Lab is broken into two parts to emphasize a good design practice
- Lab 4-A
 - Design of the filter itself
 - Using the Xilinx provided bit accurate model for the FIR compiler to choose parameters for the core that produce the desired results
 - Characterizing the performance of the filter with various inputs so that we can know exactly what to expect when placing this in the FPGA
 - Due next week
- Lab 4-B
 - Placing the core into your DDS Lab design (no software changes required)
 - Confirm with a quick ILA experiment that the results match what your model says will happen
 - Due two weeks from today

