ADS1115 Digital Filter Characterization

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The ADS1115 device is a precision, low-power, 16-bit, I2C compatible, analog-to-digital converter. These delta-sigma analog-to-digital converters (ADCs) are different from other types of data converters in that they typically integrate digital filters. The digital filter is crucial to how delta-sigma ADCs are able to obtain fine resolution across a variety of bandwidths. This application report aims to characterize ADS115 digital filter and its frequency response.

I. Introduction

To understand why the digital filter is an important aspect in delta-sigma analog-to-digital conversion, it is critical to have a basic understanding of a delta-sigma modulator. The modulator takes its input from a sample-and-hold circuit which will sample the ADC input at a rate (f_{MOD}) many times faster than the ADC output data rate (f_{DR}).

The digital filters in delta-sigma ADCs serve another function – decimation. These filters decimate data which is output from the modulator at f_{MOD} by a factor known as the oversampling ratio (OSR or N from now on). The relationship between f_{DR} and f_{MOD} is f_{DR} = f_{MOD} / N. The oversampling ratio and filter type combined determine the output bandwidth of the digital filter and overall frequency response of the ADC. Large Ns produce small filter bandwidths, which translates to very good noise performance and simplified anti-aliasing circuitry.

II. Sinc filter

The name “sinc” comes from its frequency response, which takes the form of the sin(x) / x function. The reason the filter has this response is actually tied closely with why it is so often used in delta-sigma ADCs. The digital filter creates a digital output code by summing the modulator output over a certain number of modulator clock periods. The ratio of the modulator rate (f_{MOD}) of the delta-sigma ADC to its output data rate (f_{DR}) is the OSR (N). This is equivalent to taking a moving average of those samples over the sampling period. Taking the moving average in the time domain translates to a first-order sinc response in the frequency domain. The sinc response is equal to zero at integer multiples of the data rate, which appear as notches in the magnitude response plot of the filter.

The amount of averaging increases when cascading multiple sinc filters in series, thus increasing the filter order. In the spectrum, this corresponds to a lower cutoff frequency and a higher stopband attenuation.

Next figure shows the difference in the frequency responses of a first-order sinc filter (Sinc1), three sinc filters in series (a third-order sinc, Sinc3) and five sinc filters in series (Sinc5).

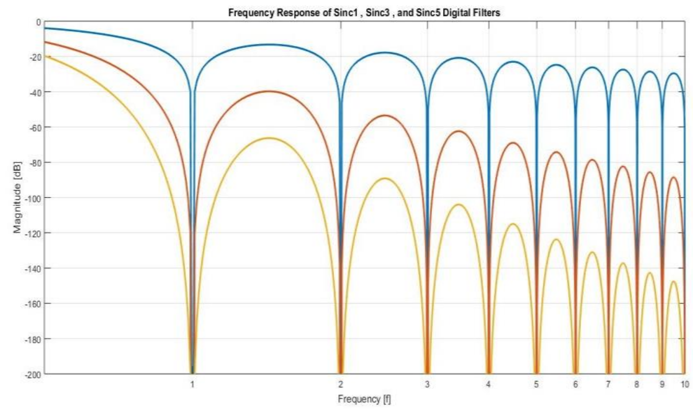

Figure 1 -Frequency Response of Sinc1 , Sinc3 , and Sinc5 Digital Filters-

Looking at these responses, there does not seem to be very much bandwidth in the digital-filter output. This is not a drawback in certain low-bandwidth applications. Some precision sensor applications, like temperature and pressure sensors, do not require much bandwidth for measurement, but need a good low-pass filter to reject out-of-band noise. The sinc filter fits well into these applications.

III. Model for ADS1115

This ADC datasheet does not contain much information about its digital filter

Only a frequency response characteristic curve for a fixed f_{DR} is given, as seen in Figure 2.

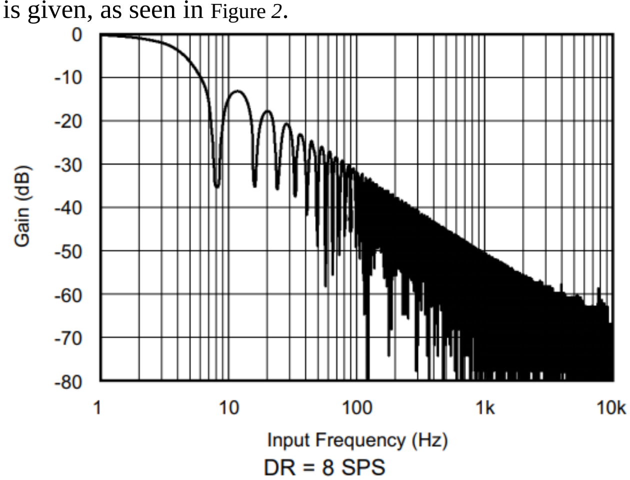

Figure 2 -Digital filter frequency response-

So, a filter characterization is needed. This can be achieved by doing some research on other similar devices from the same manufacturer (Texas Instruments) such as ADS1299 and ADS1252, and also reverse engineering the curve given above.

These two delta-sigma ADCs have more detailed filter characteristics in their datasheets. Transfer functions of the scaled Z-domain and frequency domain are given for these devices.

$$|H(f)| = \left|\frac{\sin\left(\frac{\pi \cdot f \cdot 64}{f_{MOD}}\right)}{64 \cdot \sin\left(\frac{\pi \cdot f}{f_{MOD}}\right)}\right|^5$$

or

$$H(z) = \left(\frac{1 - z^{-64}}{64 \cdot \left(1 - z^{-1}\right)}\right)^5$$

Figure 3 -ADS1252 digital filter transfer function-

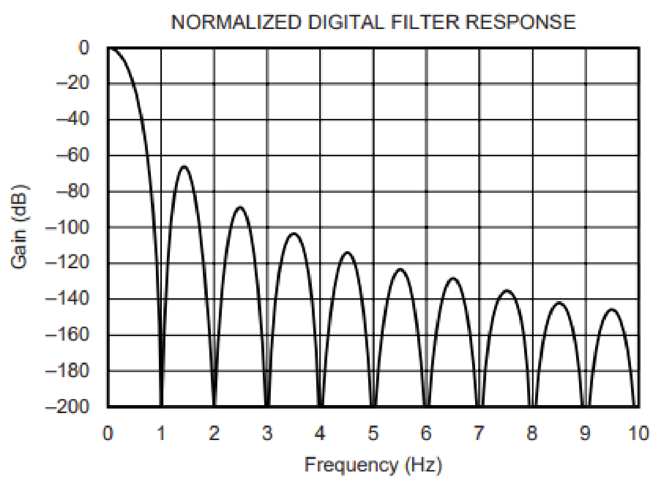

Figura 4 -ADS1252 normalized filter response-

$$|H(z)| = \left|\frac{1 - Z^{-N}}{1 - Z^{-1}}\right|^3$$

$$H(f)| = \left|\frac{\sin\left(\frac{N\pi f}{f_{MOD}}\right)}{N \times \sin\left(\frac{\pi f}{f_{MOD}}\right)}\right|^3$$

Figure 5 - ADS1299 digital filter transfer function- *N: decimation ratio*

Comparing these equations and curves with the ADS1115 filter response, it can be noticed that this ADC digital filter also responds with a sinc characteristic, with similar parameters.

It is important to highlight that while ADS1252 has a fixed oversampling ratio, its modulator frequency may vary tied to an external clock.

On the other hand, ADS1299 has a variable oversampling ratio and a fixed modulator frequency, similar to ADS1115 characteristics.

Then, the transfer function for this ADC is suspected to be like the one in Figure 5. f_{MOD} and $N = f_{MOD}/f_{DR}$ are given in the datasheet.

It remains to be determined the actual order of the filter (n).

Using the *WebPlotDigitizer* tool for extracting data from images and plots, it can be obtained the approximated -3dB gain point of the ADS1115 response curve.

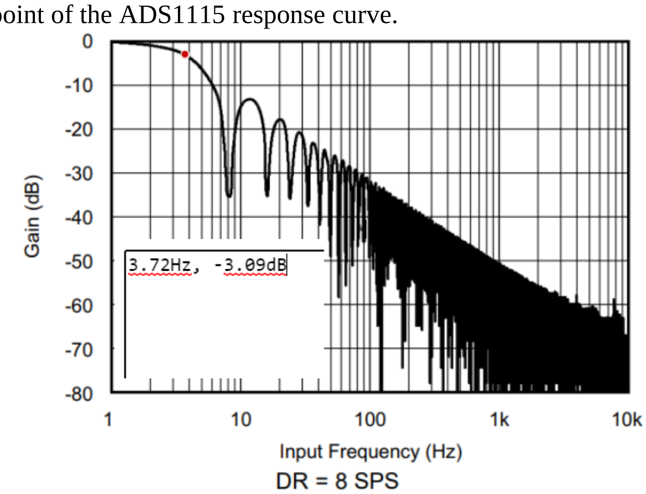

Figure 6 –ADS1115 extracted -3dB gain point-

Modeling the suspected transfer function of the filter (for a $f_{DR} = 8sps$) in MATLAB, this point can be obtained to compare and get the filter order.

The transfer function proposed:

$$|H(f)| = \left| \frac{\sin\left(\frac{N\pi f}{f_{MOD}}\right)}{N * \sin\left(\frac{\pi f}{f_{MOD}}\right)} \right|^n = \left| \frac{\sin\left(\frac{\pi f}{f_{DR}}\right)}{\frac{f_{MOD}}{f_{DR}} * \sin\left(\frac{\pi f}{f_{MOD}}\right)} \right|^n$$

From the datasheet:

$$f_{MOD} = 250kHz$$
$$f_{DR} = 8sps$$

- Modeling for 5th, 3rd and first order:

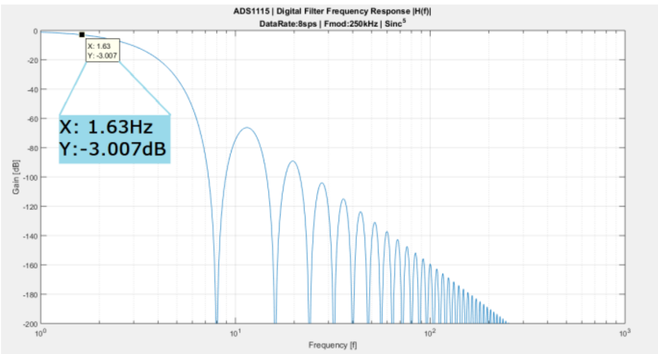

Figure 7 -5th order Sinc response model-

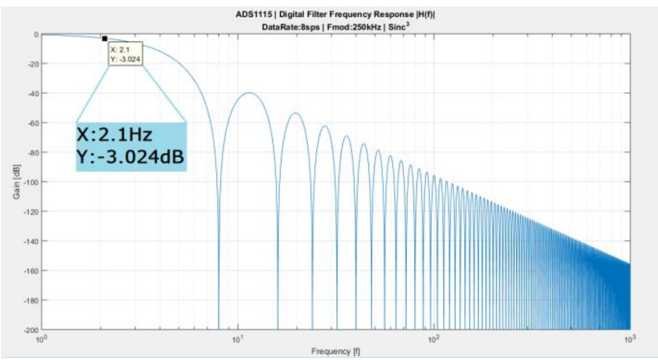

Figure 8 -3rd order Sinc response model-

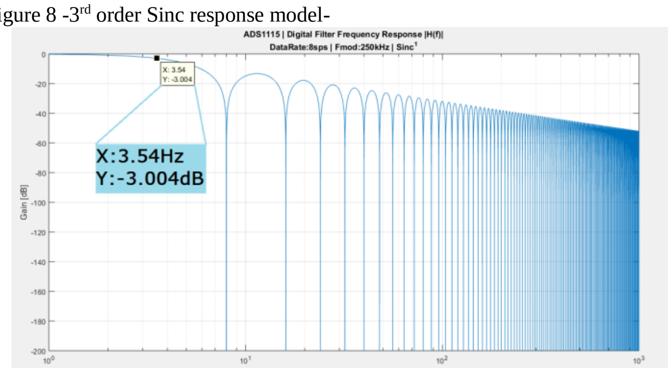

Figure 9 -First order Sinc response-

Comparing all three models with ADS1115 curve it is noticeable that is a first order digital filter (Figure 9).

Using the *WebPlotDigitizer* tool, more points can be extracted and compared with this model now.

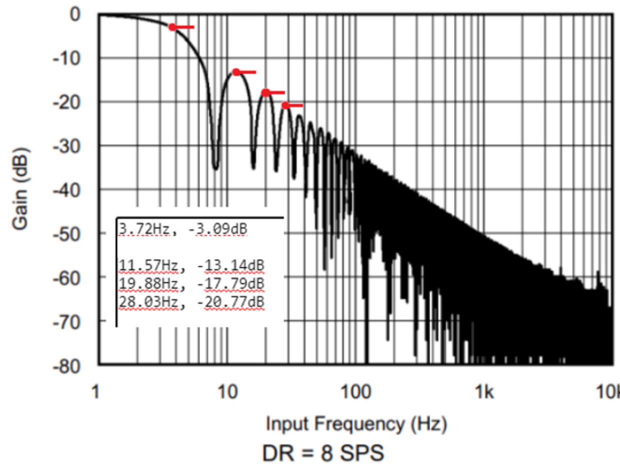

Figure 10 -Extracted points from ADS1115 curve-

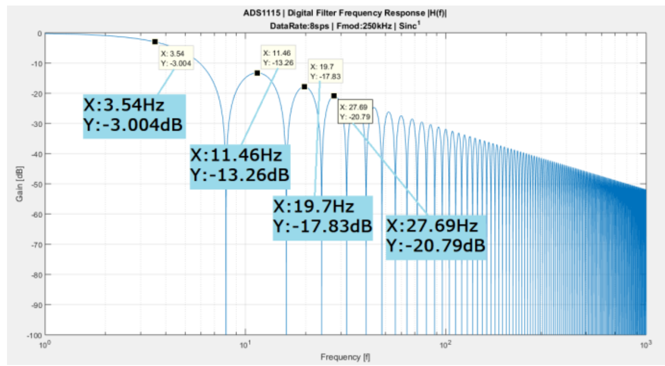

Figure 11 -First order digital filter model points-

Table 1

MODEL VS. ADS1115 CURVE POINTS COMPARISON

Freq. response extracted points	Digital filter model $\|H(f)\|$
-3.09dB 3.72Hz	-3.004dB 3.54Hz
-13.14dB 11.57Hz	-13.26dB 11.46Hz
-17.79dB 19.88Hz	-17.83dB 19.7Hz
-20.77dB 28.03Hz	-20.79dB 27.69Hz

This comparison reveals that the proposed model is quite accurate.

IV. Final Results

Finally, the transfer function for ADS1115 digital filter on the frequency domain is:

$$|H(f)| = \left| \frac{\sin\left(\frac{\pi f}{f_{DR}}\right)}{\frac{f_{MOD}}{f_{DR}} * \sin\left(\frac{\pi f}{f_{MOD}}\right)} \right|$$

- A normalized frequency response for this model is shown below:

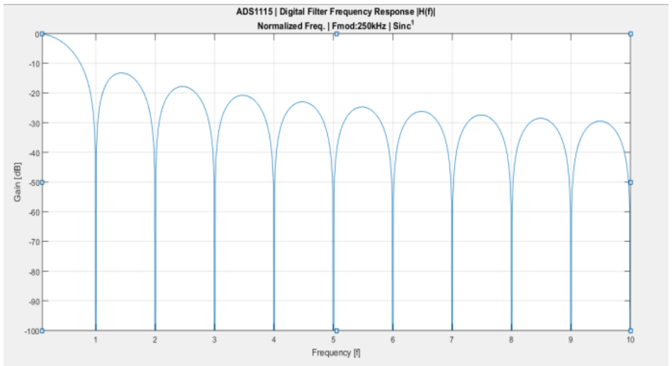

Figure 12 -Normalized (f/f_{DR}) digital filter response [Model]-

- For maximum Data-Rate:

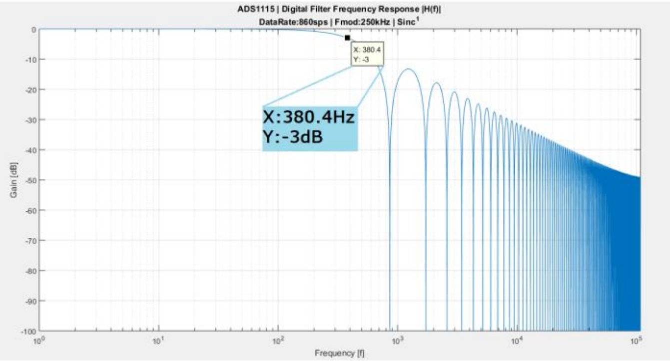

Figure 13 - Frequency response for ADS1115 maximum Data-Rate (860sps) [Model]-

References

[1] B. Pisani, "Digital Filter Types in Delta-Sigma ADCs," Texas Instruments Incorporated, Dallas, TX, USA, App. Rep. SBAA230, May 2017.

[2] Texas Instruments, "ADS111x Ultra-Small, Low-Power, I 2C-Compatible, 860-SPS, 16-Bit ADCs With Internal Reference, Oscillator, and Programmable Comparator", ADS1115 datasheet, May 2009 [Revised Jan. 2018].

[3] Texas Instruments, "24-Bit, 40kHz ANALOG-TO-DIGITAL CONVERTER", ADS1252 datasheet, Sept. 2000 [Revised June 2006].

[4] Texas Instruments, "ADS1299-x Low-Noise, 4-, 6-, 8-Channel, 24-Bit, Analog-to-Digital Converter for EEG and Biopotential Measurements", ADS1299 datasheet, June 2012 [Revised Jan. 2017].

[5] A. Rohatgi, et al., ***WebPlotDigitizer***, ver. 4.4, November 28, 2020.