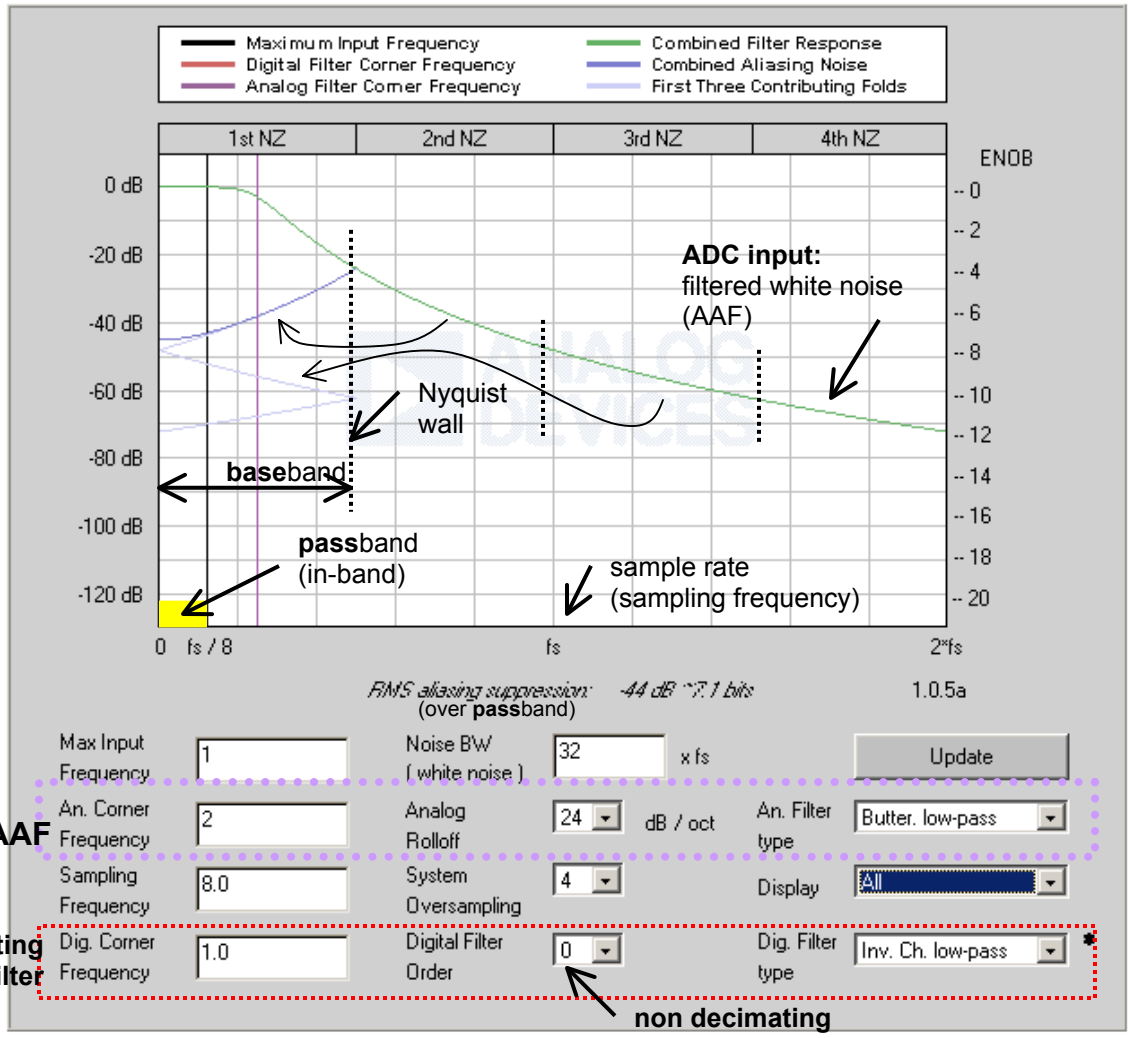
Interactive Design Tools

Analog-to-Digital Converters :

Aliasing Suppression in an Ideal [classic (non-delta-sigma)] A/D Converter with bandlimited white noise input (through filtering [AAF] and "over"sampling [decimating filter])

NZ: Nyquist Zone



Aliasing suppression in *oversampled* systems is achieved through a combination of analog (AAF) and digital (decimating) filtering.

An **ideal ADC** is assumed - distortion free, unlimited bandwidth, etc. - in order to focus solely on aliasing effects. The input signal is also assumed to be noise free, but the most practical use of this applet is to find a **combination of filtering and "over"sampling** that pushes aliasing below the *noise floor* of the input, or the system overall.

> The finite rolloff of practical analog filters means there are always some **under**sampled high frequency components that fold into the **base**band (1st NZ: "Nyquist Zone"), and appear in the sampled signal as noise. Digital filtering can then be used on the sampled signal to eliminate frequencies between **pass**band and Nyquist. For the simple case of an input composed of bandlimited white noise, this applet gives an estimate (*RMS average suppression*) of how much out-of-band signal will be folded into the **pass**band.

Behavior of this hypothetical ADC is assumed frequency-independent, so only **frequency ratios** matter - the units must be uniform, but are otherwise irrelevant. Note: the *default* example places the analog corner at 2X the maximum frequency of interest to minimize potential phase distortion in the passband.

Instructions

The applet above illustrates aliasing and its suppression through filtering and oversampling in a classic (non-sigma-delta) A/D converter. An ideal ADC is assumed - distortion free, unlimited bandwidth, etc. - in order to focus solely on aliasing effects. The input signal is also assumed to be noise free, but the most practical use of this applet is to find a combination of filtering and oversampling that pushes aliasing below the noise floor of the input, or the system overall.

The finite rolloff of practical analog filters means there are always some undersampled high frequency components that fold into the passband, or "Nyquist Zone" (NZ), and appear in the sampled signal as noise.

For the simple case of an **input** composed of **bandlimited white noise**, this applet gives an estimate of how much out-of-band signal will be folded into the baseband (1st NZ). In many real-world situations, the out-of-band signal is of lesser amplitude and so this estimate is too conservative; in other cases it might not be conservative enough - it depends on the out-of-band signal level. *The bandwidth limit for the input white noise signal is a multiple of the sampling frequency. By default, that multiple is 32X; the maximum is 256X.*

Aliasing suppression in oversampled systems is achieved through a combination of analog and digital filtering, but digital filtering cannot replace a high-quality analog filter because it cannot remove aliasing noise after it's been folded into the passband by the sampling process. Instead, **oversampling** must be used to put enough octaves and attenuation between the Nyquist frequency and the highest passband frequency of interest. Digital filtering can then be used on the sampled signal to eliminate frequencies between passband and Nyquist. For simplicity, the additional aliasing noise that results from downsampling [decimating] the sampled signal is **not** shown.

How to use this applet:

Enter the **Maximum Input Frequency**, **Sampling Frequency** and **Oversampling Ratio** in the fields provided. Hit "Enter" or click "Update" to recompute the display. (The maximum input frequency is marked with a black line.)

Select **Digital** and **Analog Corner Frequencies**. Behavior of this hypothetical ADC is assumed frequency-independent, so only frequency ratios matter - the units must be uniform, but are otherwise irrelevant. A red cursor marks the digital filter corner frequency and purple the analog corner. The default example places the analog corner at 2X the maximum frequency of interest to minimize potential phase distortion in the passband. ***** *Please note that for speed of computation, digital filter characteristics are those of a BT-transformed IIR filter.*

Select filter rolloffs and types. The combined filter response of the analog and digital filters is shown in green.

Combined aliasing noise is shown in dark blue with the first three contributing folds in light blue. Combined noise is summed from DC to the frequency specified by **Noise BW**. (Note: 32X the sampling frequency is usually plenty and the applet will slow down if too much noise BW is specified)

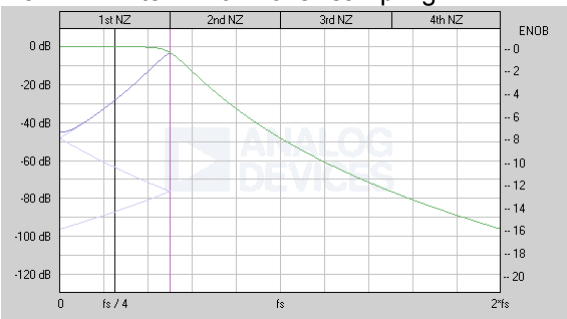
Suppression in **dB** is shown on the left vertical axis with the corresponding Equivalent Number Of Bits (**ENOB**) shown at right. A summary of the RMS average suppression over the passband is immediately below the chart.

Experiment by changing the analog filter parameters and comparing the results to changing the **Oversampling Ratio**. Changing the oversampling ratio multiplies the **Max Input Frequency** to get a **Sampling Frequency** but the reverse direction doesn't work - it's sampling frequency that's important and the oversampling ratio menu is just a convenience.

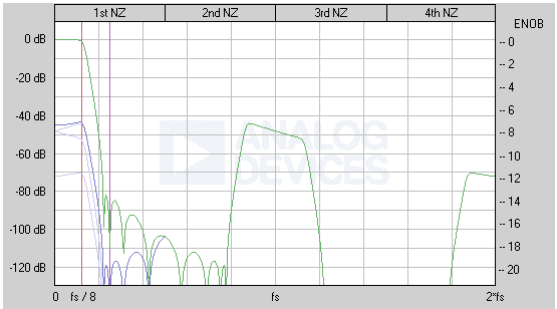
Quickstart:

Compare the effects of maximum analog filter with low oversampling with a more moderate filter, with higher oversampling. Try maximum and no digital filtering.

max ANA filter w/ low “over”sampling:



max DIG filtering:



http://www.analog.com/Analog_Root/static/techsupport/designtools/interactiveTools/adc/adcalias.html