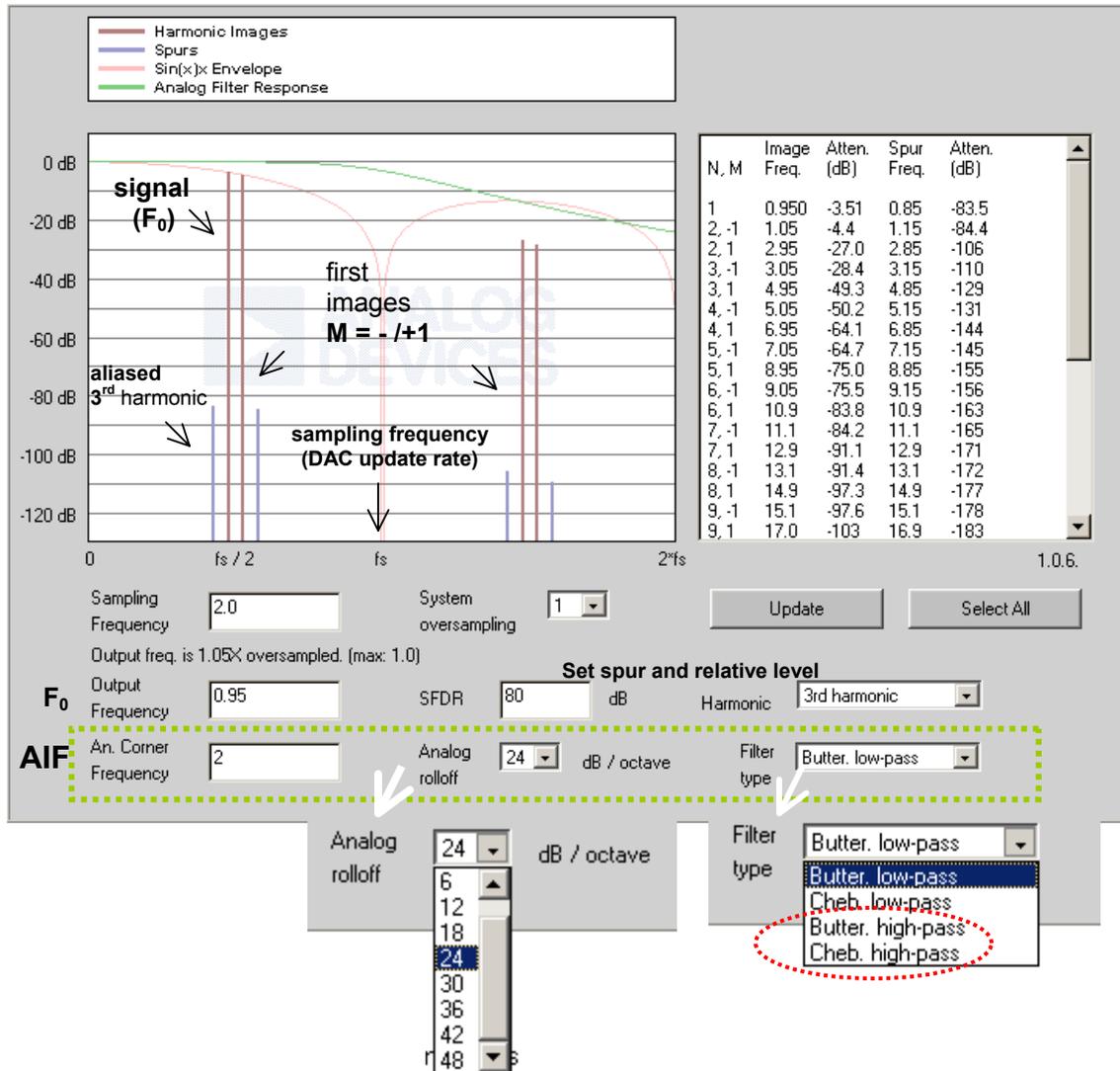


# Interactive Design Tools

## Digital-to-Analog Converters :

### Harmonic Images [and spurs] in an Ideal [NRZ mode] D/A Converter



Images are located at  $N \cdot F_s + M \cdot F_o$ , where  $M = +/-1$ .

// Note: there is a mistake in the Table. Can you spot it? //

To illustrate suppression of these images, the applet can apply a simulated post-DAC **analog filter (AIF)**.

Without external filtering image amplitudes *roll off* as  $\sin(x)/x$  where  $x = \pi \cdot F_o / F_s$  [in NRZ mode]. Spurious 2nd or 3rd harmonics of the fundamental are assumed to result from D/A nonlinearities and so remain folded within the baseband.

Behavior of this hypothetical DAC is assumed frequency-independent, so only **frequency ratios** matter - the units must be uniform, but are otherwise irrelevant.

The **table** data is selectable and can be *copied and pasted* into a spreadsheet. Use "Select All" to conveniently select the entire contents before copying.

## Instructions

The applet above demonstrates harmonic images (red) and spurs (blue) in an idealized DAC output.

Images are located at  $N \cdot F_s + M \cdot F_o$ , where  $M = \pm 1$ . Without external filtering image amplitudes roll off as  $\sin(x)/x$  where  $x = \pi \cdot F_o / F_s$ . Spurious 2nd or 3rd harmonics of the fundamental are assumed to result from D/A nonlinearities and so remain folded within the baseband. These spurs in turn have their own harmonic images that roll off as  $\sin(x)/x$ . The  $\sin(x)/x$  envelope is shown in light red.

To illustrate suppression of these images, the applet can apply a simulated post-DAC **analog** filter.

The output signal is assumed to have been generated at the DAC **update rate** (or perfectly reconstructed from a lower data rate) - digital filtering isn't useful because it can't affect images at multiples of the DAC update rate nor the distortion spurs that occur as part of the analog conversion process. Harmonics must therefore be suppressed by analog filtering.

Digital filtering is useful for sample rate upconversion (**not** shown) where it produces a similar set of digitally filtered images, but no spurs.

## How to use this applet:

Enter the **Sampling Frequency** or **Oversampling Ratio** in the fields provided.

Hit "Enter" or click "Update" to recompute the display.

Select the **Analog Corner Frequency**. Behavior of this hypothetical DAC is assumed frequency-independent, so only frequency ratios matter - the units must be uniform, but are otherwise irrelevant.

Select filter rolloff and type. The analog filter response is shown in green.

Image frequencies and amplitudes are shown in a table at top right. The first column gives the  $N$  ( $\cdot F_s$ ) and whether it's  $\pm F_o$  ( $M = \pm 1$ ). The table data is selectable and can be copied and pasted into a spreadsheet. Use "Select all" to conveniently select the entire contents before copying.

Experiment by changing the analog filter parameters and comparing the results to changing the **Oversampling Ratio**. Changing the oversampling ratio changes the **Sampling Frequency** but the reverse is not true - it's sampling frequency that's important and the oversampling ratio menu is just a convenience.

**SFDR** is used to set the relative level of spurs, which are assumed here to be the result of D/A nonlinearity.

**Harmonic** selects whether these distortion spurs are most prominent at 2X ( $\downarrow$ ) or 3X (default) the **Output Frequency**.

By varying the output frequency in small increments, you can see how the harmonics move and fold.

SFDR is a DAC datasheet parameter and, in general, depends on both the **sample rate** and output frequency, among other variables. However, a single compromise number is often specified.

