

Oversampling Interpolating DACs

by Walt Kester

INTRODUCTION

Oversampling and digital filtering eases the requirements on the antialiasing filter which precedes an ADC. The concept of oversampling and interpolation can be used in a similar manner with a reconstruction DAC. For instance, oversampling is common in digital audio CD players, where the basic update rate of the data from the CD is 44.1 kSPS. Early CD players used traditional binary DACs and inserted "zeros" into the parallel data, thereby increasing the effective update rate to 4-times, 8-times, or 16-times the fundamental throughput rate. The 4×, 8×, or 16× data stream is passed through a digital interpolation filter which generates the extra data points. The high oversampling rate moves the image frequencies higher, thereby allowing a less complex lower cost filter with a wider transition band. In addition, there is an increase in the SNR within the signal bandwidth because of the process gain. The sigma-delta DAC architecture uses a much higher oversampling rate and represents the ultimate extension of this concept and has become popular in modern CD players.

The same concept of oversampling and interpolation is also utilized in high speed DACs used in communications applications, relaxing the requirements on the output filter as well as increasing the SNR due to process gain.

OUTPUT SPECTRUM OF A RECONSTRUCTION DAC

The output of a reconstruction DAC can be represented as a series of rectangular pulses whose width is equal to the reciprocal of the clock rate as shown in Figure 1.

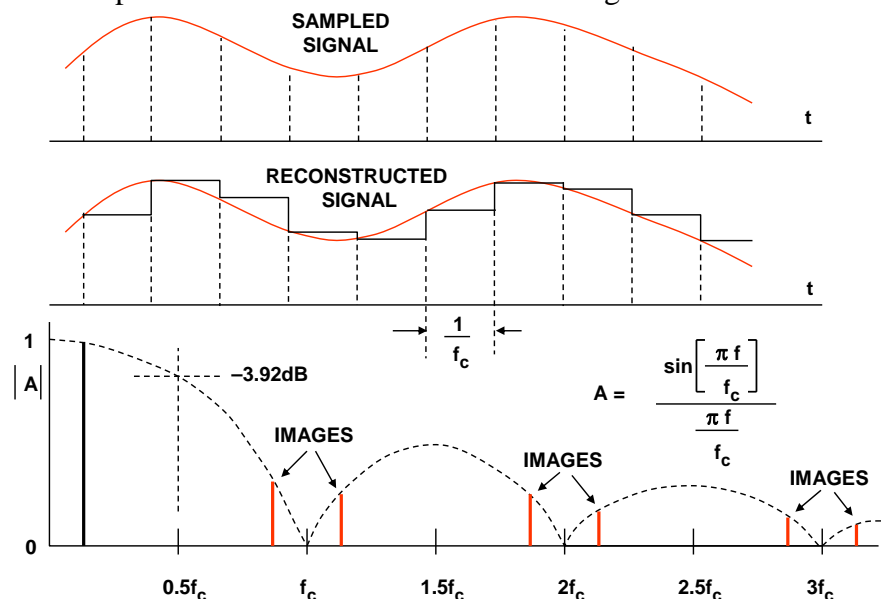


Figure 1: Unfiltered DAC Output Showing Images and $\sin(x)/x$ Roll Off

Note that the reconstructed signal amplitude is down 3.92 dB at the Nyquist frequency, $f_c/2$. An inverse $\sin(x)/x$ filter can be used to compensate for this effect if required. The images of the fundamental signal occur as a result of the sampling function and are also attenuated by the $\sin(x)/x$ function.

OVESAMPLING INTERPOLATING DACS

The basic concept of an oversampling/interpolating DAC is shown in Figure 2. The N-bit words of input data are received at a rate of f_c . The digital interpolation filter is clocked at an oversampling frequency of Kf_c , and inserts the extra data points. The effects on the output frequency spectrum are shown in Figure 2. In the Nyquist case (A), the requirements on the analog anti-imaging filter can be quite severe. By oversampling and interpolating, the requirements on the filter are greatly relaxed as shown in (B). Also, since the quantization noise is spread over a wider region with respect to the original signal bandwidth, an improvement in the signal-to-noise ratio is also achieved. By doubling the original sampling rate ($K = 2$), an improvement of 3 dB is obtained, and by making $K = 4$, an improvement of 6 dB is obtained. Early CD players took advantage of this, and generally carried the arithmetic in the digital filter to more than N-bits. Today, most DACs in CD players are sigma-delta types.

One of the earliest publications on the oversampling/interpolating DAC concept was by Ritchie, Candy, and Ninke in 1974 (Reference 1) and followed by a 1981 patent (filing date) by Mussman and Korte (Reference 2).

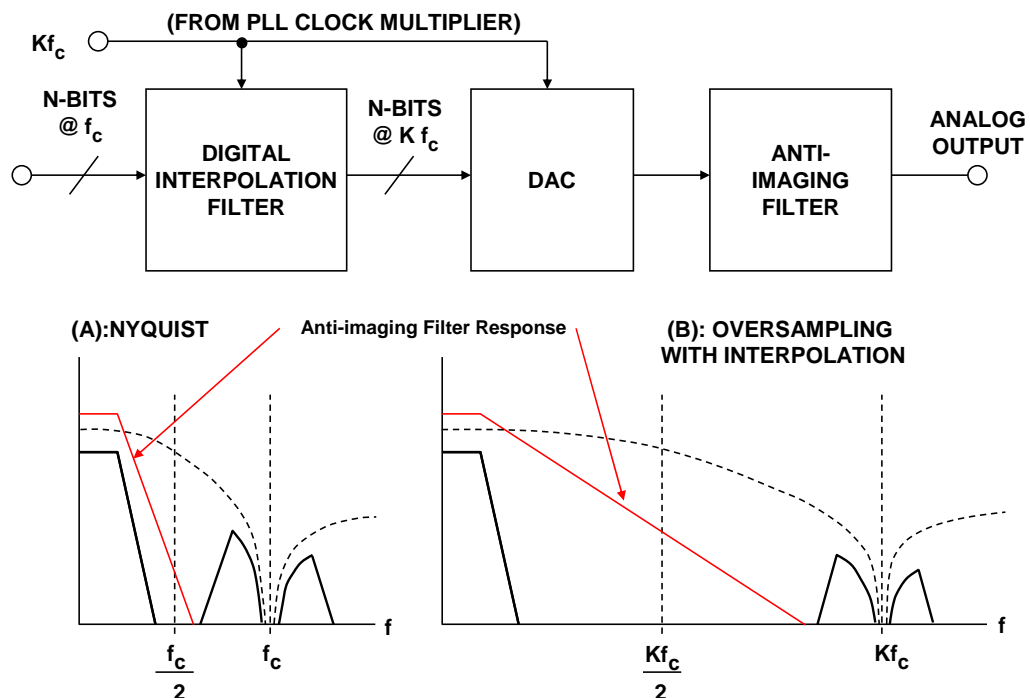
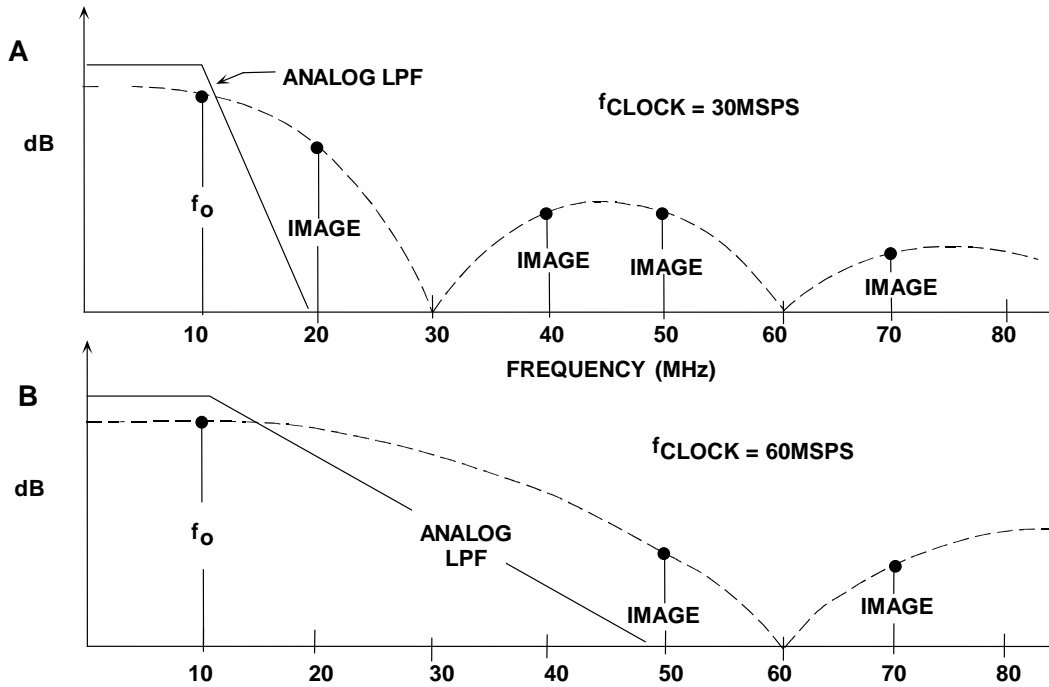


Figure 2: Oversampling Interpolating DAC

The following example illustrates the concept of oversampling using some actual numbers. Assume a traditional DAC is driven at an input word rate of 30 MSPS (see Figure 3A). Assume the DAC output frequency is 10 MHz. The image frequency component at $30 - 10 = 20$ MHz must be attenuated by the analog antialiasing filter, and the transition band of the filter starts at 10 MHz and ends at 20 MHz. Assume that the image frequency must be attenuated by 60 dB. The filter must therefore go from a passband corner frequency of 10 MHz to 60 dB of stopband attenuation over the transition band between 10 and 20 MHz (one octave). A filter gives approximately 6-dB attenuation per octave for each pole. Therefore, a minimum of 10 poles is required to provide the desired attenuation. Filters become even more complex as the transition band becomes narrower.



**Figure 3: Analog Filter Requirements for $f_o = 10$ MHz:
(A) $f_c = 30$ MSPS, and (B) $f_c = 60$ MSPS**

Assume that we increase the DAC update rate to 60 MSPS and insert a "zero" between each original data sample. The parallel data stream is now 60 MSPS, but we must now determine the value of the zero-value data points. This is done by passing the 60-MSPS data stream with the added zeros through a digital interpolation filter which computes the additional data points. The response of the digital filter relative to the $2\times$ oversampling frequency is shown in Figure 3B. The analog antialiasing filter transition zone is now 10 to 50 MHz (the first image occurs at $2f_c - f_o = 60 - 10 = 50$ MHz). This transition zone is a little greater than 2 octaves, implying that a 5- or 6-pole filter is sufficient.

The [AD9773/AD9775/AD9777](#) (12-/14-/16-bit) series of Transmit DACs (TxDAC[®]) are selectable $2\times$, $4\times$, or $8\times$ oversampling interpolating dual DACs, and a simplified block diagram is shown in Figure 4. These devices are designed to handle 12-/14-/16-bit input word rates up to 160 MSPS. The output word rate is 400 MSPS maximum. For an output frequency of 50 MHz,

an input update rate of 160 MHz, and an oversampling ratio of 2×, the image frequency occurs at 320 MHz – 50 MHz = 270 MHz. The transition band for the analog filter is therefore 50 MHz to 270 MHz. Without 2× oversampling, the image frequency occurs at 160 MHz – 50 MHz = 110 MHz, and the filter transition band is 50 MHz to 110 MHz.

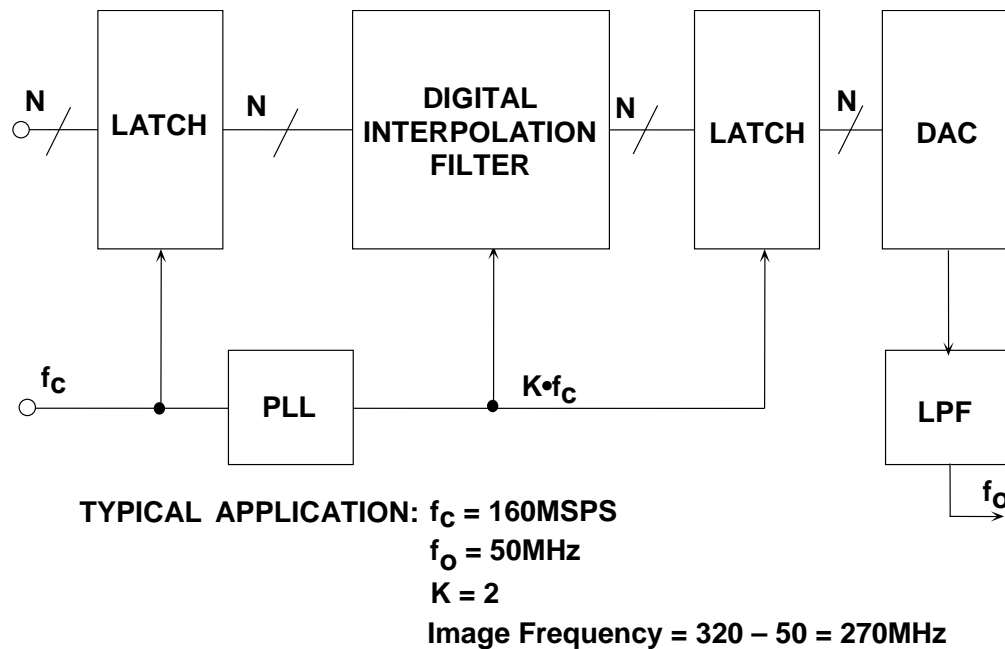


Figure 4: Oversampling Interpolating TxDAC® Simplified Block Diagram

Notice also that an oversampling interpolating DAC allows both a lower frequency input clock and input data rate, which are much less likely to generate noise within the system.

SIGMA-DELTA DACS

Sigma-delta DACs operate very similarly to sigma-delta ADCs, however in a sigma-delta DAC, the noise shaping function is accomplished with a digital modulator rather than an analog one.

A Σ - Δ DAC, unlike the Σ - Δ ADC, is mostly digital (see Figure 5A). It consists of an "interpolation filter" (a digital circuit which accepts data at a low rate, inserts zeros at a high rate, and then applies a digital filter algorithm and outputs data at a high rate), a Σ - Δ modulator (which effectively acts as a low pass filter to the signal but as a high pass filter to the quantization noise, and converts the resulting data to a high speed bit stream), and a 1-bit DAC whose output switches between equal positive and negative reference voltages. The output is filtered in an external analog LPF. Because of the high oversampling frequency, the complexity of the LPF is much less than the case of traditional Nyquist operation.

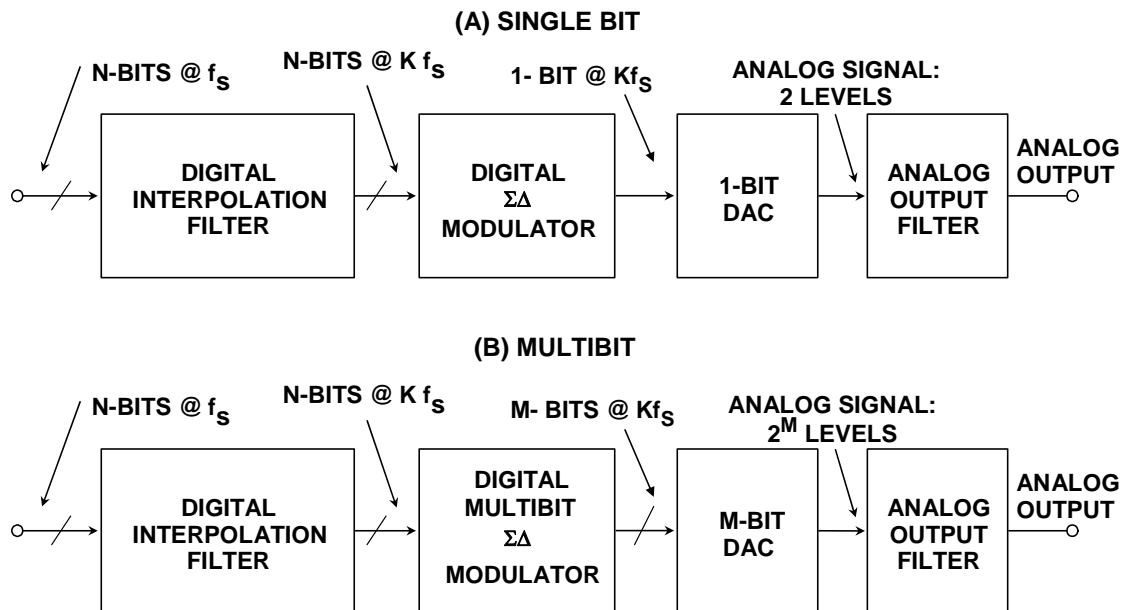


Figure 5: Sigma-Delta DACs

It is possible to use more than one bit in the Σ - Δ DAC, and this leads to the *multibit* architecture shown in Figure 5B. The concept is similar to that of interpolating DACs previously discussed, with the addition of the digital sigma-delta modulator.

In the past, multibit DACs have been difficult to design because of the accuracy requirement on the n-bit internal DAC (this DAC, although only n-bits, must have the linearity of the final number of bits, N). The AD195x-series of audio DACs, however use a proprietary *data scrambling* technique (called *data directed scrambling*) which overcomes this problem and produces excellent performance with respect to all audio specifications.

The [AD1955](#) multibit sigma-delta audio DAC is shown in Figure 6. The AD1955 also uses data directed scrambling, supports a multitude of DVD audio formats and has an extremely flexible serial port. THD + N is typically 110 dB.

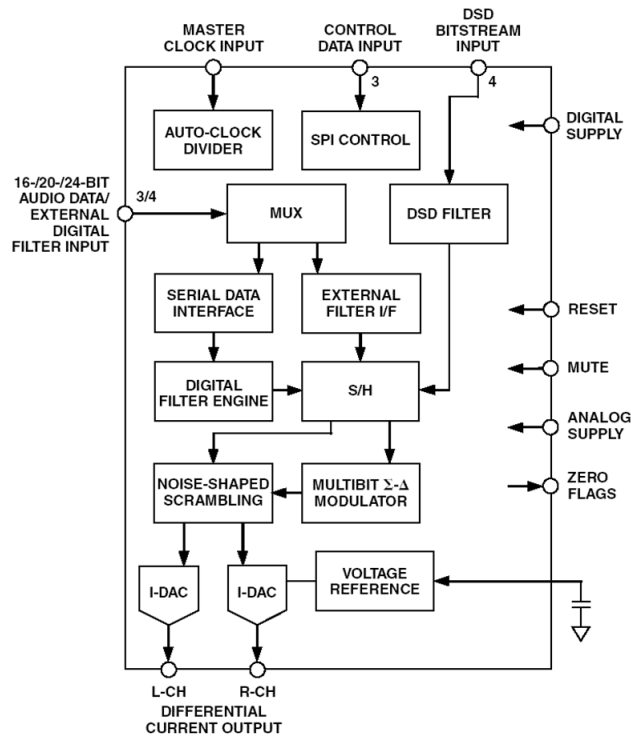


Figure 6: AD1955 Multibit Sigma-Delta Audio DAC

SUMMARY

Oversampling used in conjunction with digital filtering is a powerful tool in modern sampled data systems. We have seen how the same fundamental theory is applicable to both ADCs and reconstruction DACs. A primary advantage is the relaxation of the requirements on the antialiasing/anti-imaging filter. Another advantage is the increase in SNR which occurs because of the process gain.

The Σ - Δ ADC and DAC architecture is the ultimate extension of the oversampling concept and is the architecture of choice for most voiceband and audio signal processing data converter applications.

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