ASIC implementation of Up-sampling Built in 6GS/s-16bit DAC

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Abstract. This paper presents an interpolation filtering model including four Digital interpolation filters. The interpolation filter circuit can effectively increase the data rate without increasing the input rate of the interface, and realize the up-sampling processing of the external input low-frequency digital signal to meet the high-speed DAC core conversion data requirement.

1. Introduction
Multirate systems face the problem of data rate conversion. The basic methods of rate conversion are extraction and interpolation operations. The conversion that reduces the sampling rate is called extraction, and the conversion that increases the sampling rate is called interpolation. Under the premise of satisfying the sampling theorem, there are two ways to convert the data rate: one is to convert the digital signal obtained from a certain sampling rate into an analog signal through a digital-to-analog converter, and then use another sampling rate to sample. Another method is to directly convert the sampling rate using a digital signal processing method. The digital processing method is more direct, convenient and flexible. The high speed, high precision DAC is a typical multirate system. Since the rate of the interface cannot meet the requirements of the high-speed DAC core conversion data, we need to use interpolation filtering technology. The interpolation filter circuit can effectively increase the data rate without increasing the input rate of the interface, and realize the up-sampling processing of the external input low-frequency digital signal to meet the high-speed DAC core conversion data requirement.

A common technique in digital circuitry for multirate systems is the interpolation filtering. Due to a wide range of applications and market needs, many people study the techniques for designing these filters. Halfband filters [1]–[7], particularly FIR halfband filters, are widely used in RF/IF data converters, as anti-imaging upsampling filters, or anti-aliasing downsampling filters.

This work presents an upsampling circuit structure built in high speed, high precision DAC. The system can realize 2X, 3X, 4X, 6X, 8X, 12X interpolation by bypass filter. The hardware overhead is reduced by the structural design of the half-band filter and the optimized design of the coefficient generation circuit.

The remainder of this paper is organized as follows. Section II is a brief description of previous work. In Section III, we propose our interpolation filtering structure. Results are presented in Section IV. We conclude the paper in Section V.
2. Previous work

2.1. Digital filtering
A filter whose function is to filter the input signal. Linear time invariant systems whose time domain input and output relationships are

\[ y(n) = x(n) \otimes h(n) \]  

If there are \( \tilde{X}(e^{j\omega}) \) and \( \tilde{Y}(e^{j\omega}) \) Fourier transforms, respectively, then the relationship between input and output is

\[ \tilde{Y}(e^{j\omega}) = \tilde{X}(e^{j\omega}) \cdot H(e^{j\omega}) \]  

\( H(e^{j\omega}) \) is the frequency response of the system.

When \( x(n) \) passes through the system \( h(n) \), the output \( y(n) \) no longer contains the frequency component of \( |\omega| > \omega_c \). The ingredients are passed "without distortion". Therefore, different shapes of \( H(e^{j\omega}) \) can be designed to obtain different filtering results.

Digital filter means that the input and output of the filter are discrete time signals, and the impulse response is also discrete. Delays, multipliers, and adders are essential components of hardware implementation of digital filters.

2.2. Interpolation filtering
Digital interpolation filtering technology can effectively improve the data rate. The specific structure is firstly interpolated by zero value and then realized by a low-pass filter. This paper uses the interpolation filter method, namely the discrete domain interpolation method. If \( N \)-time interpolation is implemented, \((N-1)\) sample values are inserted between two adjacent points, which may be \((N-1)\) zero-valued points, and the output is:

\[ X_{\text{d}}(z) = \sum_{n=-\infty}^{\infty} x_e(n)z^{-n} = \sum_{n=0,\pm N}^{\infty} \frac{x(n)}{N} \cdot \frac{z^{-n}}{N} = X_e(z) \]  

Bring \( z = e^{j\omega} \) into the spectrum of \( X_e(n) \)

\[ X_{\text{d}}(e^{j\omega}) = \frac{X_e(e^{j\omega})}{N} \]  

It can be concluded from the above formula that after interpolation, the sampling frequency is expanded. In addition to the spectrum of the swimming signal, the baseband signal is also imaged at \( \pm 2\frac{2\pi}{N} \) and \( \pm 4\frac{2\pi}{N} \) ... The pass filter filters out the image signal, which acts as a signal smoothing in the time domain. The main function of the low-pass filter is to filter the interpolated signal. There is no feedback loop in the Finite Impulse Response (FIR) filter structure, and the phase frequency has a good linearity. In order to ensure the performance of the D/A converter, an FIR filter is used.

The filter order \( N \) is determined by bandpass ripple \( \delta_{\text{pass}} \), stopband ripple \( \delta_{\text{stop}} \), transition band bandwidth \( f_{\text{tran}} \), and filter operating frequency \( f_s \):

\[ N = \frac{B_{\text{pass}} + B_{\text{stop}}}{f_{\text{tran}} / f_s} \]  

where \( f_{\text{tran}} = f_{\text{stop}} - f_{\text{pass}} \), \( f_{\text{stop}} \) is the filter stop band frequency, and \( f_{\text{pass}} \) is the pass band frequency. It can be obtained that the order required by the filter is related to the bandwidth of the transition band, and the narrower the transition band, the higher the required filter order. In order to reduce the occupation of circuit area and power consumption by high-order filters, interpolation filters generally use Half-Band Filter (HBF).

2.3. Half-band filters
The half-band filter has a particularly important position in multi-rate signal processing because it is particularly suitable for realizing the power-doubling extraction or interpolation, and is computationally efficient and real-time.
The so-called half-band filter refers to a filter whose frequency response $H$ satisfies the following relationship.

\[
\begin{align*}
\omega_p &= \pi - \omega_s \\
\delta_p &= \delta_s = \delta
\end{align*}
\]

Or the band width $\pi - \omega_s$ of the half band filter is equal to the pass band width $\omega_p$, and the pass band ripple and the stop band ripple are also equal. $\omega_s$ and $\omega_p$ are the passband upper limit and the stopband lower limit frequency of the filter, respectively.

The half band filter has the following properties:

1. The value of the impulse response not included in the even serial number of the half-band filter is 0, that is,

\[ h(n) = 0, \text{ } n = 0, \pm 2, \pm 4, \ldots \]

2. The impulse response of the half-band filter is doubled compared to the interpolation rate of the signal, so there is an aliasing effect in the transition band, which can make the anti-aliasing in the passband region.

3. $H(e^{j\omega}) = 1 - H(e^{j(\pi - \omega)})$, and the passband corrugated stopband corrugations are equal, that is, $\delta_p = \delta_s = \delta$, so that nearly half of $h$ is 0, which reduces the computational complexity by half in practical applications.

In addition to the intermediate coefficient, all the even coefficients of HBF are 0, so the HBF calculation is reduced by nearly half compared with other linear phase FIR filters of the same length. Meanwhile, the HBF intermediate coefficient is 0.5, and the two sides of the coefficient can share a multiplication unit with respect to the center symmetry. The hardware cost of the multiplier is halved, so the HBF has a high calculation rate and high utilization of hardware resources, and is particularly suitable for real-time processing of signals.

### 3. Proposed Interpolation Filtering Model

The clock frequency of the DAC is 6 GHz, the input signal bandwidth is up to $6/2 \times 80\% = 2400$ MHz, the transition band bandwidth is $f_{trang} < 1200$ MHz, the ripple is $\delta < 0.001$ dB, the stopband attenuation is greater than 85 dB; the input signal bandwidth is slightly lower than Nyquist bandwidth helps to properly relax the f-requirement, thereby reducing the required half-band filter order.

The half-band filter is designed by using the Filter Design and Analysis Toolbox provided by Matlab, and the order of the half-band filter and the specific coefficient are determined by the Minimum Order method provided by Matlab.

In order to save hardware overhead, we set the minimum coefficient and finally determine the minimum coefficient order to be 51. We use the multiplier multiplexing method to construct the structure of the 51-order half-band filter as shown in Figure 1. This structure can halve the overhead of the multiplier.

Because the even branch coefficient has only the intermediate coefficient $h(26)=0.5$, and the interpolation filter gain is 2 in order to compensate the amplitude loss caused by the 2x interpolation zero value interpolation. The even branch only needs to output the input sequence as it is. In order to further save the number of registers, the delay unit that realizes the even branch by using the folded structure can be shared with the odd branch, and thus the final two-phase structure of the filter is obtained as shown in Figure 1.

The two-phase decomposition principle can divide the filter into two equal-divided branches, and the input data is input to the parity branch at a rate $f_0$. Each input of a sample $X(n)$, the odd-even branches are respectively calculated to obtain corresponding results, and then the switch is switched at a rate of $2f_0$. Switching back and forth between the two branches of the parity gives the output sequence $Y(n)$. 
The coefficients in the table are all reduced to the simplest integer, and the common factor is $2^{15}$. The coefficients of the design are given in Figure 2. The even coefficients in the table are all 0, and the odd coefficients are the same in pairs.

The advantage of using the look-up table to generate coefficients is fast, but the need to introduce storage and addressing units increases hardware consumption. In the design of this paper, the method generated in the field is used to directly generate coefficients for use. In order to reduce the hardware overhead, high-speed data interpolation filtering can be realized at the same time, the multiply-accumulate structure is optimized, and the sub-operation result sharing technique is adopted. Finally, 23 adders are used, and the depth is 5, and the multiplication of 13 coefficients is completed. The coefficient generation circuit is the core module of the half-band filter, as shown in Figure 3.
Figure 3. The coefficient generation circuit of half-band filter.

An upsampling system is built with three half-band filters and one FIR filter. The system can realize 2X, 3X, 4X, 6X, 8X, 12X interpolation by bypass filter. The first 2X filter in the figure is a 51-order half-band filter, the first 3X filter is a 67-order FIR filter, and the second 2X filter is a 19-stage half-band filter. The third 2X filter is an 11th-order half-band filter. The structure of the system is shown in Figure 4.

Figure 4. Proposed interpolation filtering model.

4. Result
The frequency domain characteristics of the half-band filter are shown in Figure 5. The clock frequency is 6 GHz. Under the normalized frequency, the passband frequency $f_{\text{pass}}=0.4$, the stopband frequency $f_{\text{stop}}=0.6$, the transition band bandwidth $f_{\text{trans}}<0.2$, and the ripple $\delta < \pm 0.001$dB, the stopband attenuation is greater than 85dB;
In the DSP system, the sampling mode of the ADC sinusoidal analog input signal frequency and sampling clock frequency that meets the requirements of equation (6) is coherent sampling:

$$\frac{\tilde{f}_s}{\tilde{f}_i} = \frac{C}{N}$$

(8)

Where $\tilde{f}_s$ is the sampling clock frequency, $\tilde{f}_i$ is the input analog signal frequency of the ADC, $N$ is the sampling data length, and $C$ is the sampling data length $N$ containing the number of complete input signals, requiring $N$ and $C$ to be prime numbers. The FFT method requires $N$ to be an integer power of 2, therefore, $C$ is a condition that any odd number can satisfy the mutual prime.

On the one hand, interpolation filtering can effectively increase the data rate and meet the system’s demand for data rate. On the other hand, interpolation filtering can guarantee and improve the quantization noise and linearity of the signal. The interpolated time domain waveform is shown in Figure 6.

After the FFT transformation, the waveform to the frequency domain is as shown in Figure 7.
Figure 7. The time domain waveform of 2x 4x 8x.

5. Conclusion
An upsampling circuit structure built in high speed, high precision DAC is proposed in this paper. The proposed upsampling circuit can effectively increase the data rate without increasing the input rate of the interface, and realize the up-sampling processing of the external input low-frequency digital signal to meet the high-speed DAC core conversion data requirement. The hardware overhead is reduced by the structural design of the half-band filter and the optimized design of the coefficient generation circuit. The system can realize 2X, 3X, 4X, 6X, 8X, 12X interpolation by bypass filter.

Acknowledgments
This work was supported by the Self-Supported Program of Beijing Microelectronics Technology Institute.

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