An Efficient Waveform Reconstruction Method for Digital Bandwidth Interleaving Sampling System

Yubo Ma[1,2], Jie Wu[1,2]

[1]Department of Modern Physics, University of Science and Technology of China, Hefei 230026, China

[2]State Key Laboratory of Particle Detection and Electronics

Introduction

High end electronic measurement equipment is an in-dispensable part of the development of communication and electronic technology, and its core part is a high-speed ADC. However, the bandwidth and speed of analog-to-digital converters(ADCs) are usually limited by the microelectronics process. Multi-channel parallel sampling is an attractive way to solve this problem, which can increase the sampling rate and bandwidth of the ADC. Time-interleaved ADC (TIADC) is the most common method to increase the sampling rate. TIADC adjusts the sampling clock phase of each channel so that high-speed signals are alternately sampled by multichannel parallel low-speed ADCs, so that the sampling rate of the total system is multiplied. However, the TIADC sampling structure faces the problems of gain, DC offset, sample-time mismatches between multiple channels. The emergence of digital bandwidth interleaving (DBI) technology makes this problem have a good solution. Recently there are some studies on DBI sampling systems. Some are studies on analog LO and digital LO synchronization methods, replacing classical method of inserting pilot tone to achieve phase synchronized mixing. Some are discussed for amplitude and phase correction methods. However, these works are not optimized for the waveform recovery algorithm. The DBI systems implemented in are all based on a large amount of computing resources or use software to complete the recovery of part of the waveform. This paper introduces an efficient waveform reconstruction method to solve this problem.

Proposed Waveform Reconstruction Method

The sampling and reconstruction process of the two-channel DBI system is shown in fig. 1. The input $x(t)$ which is bandlimited to π/T is divided into high frequency and low frequency parts through a dixplexer filter $H_1(\Omega)$, $H_1(\Omega)$. The low-frequency part is directly sampled by the ADC, and the high-frequency part is down-converted by an analog mixer, and the image frequency is eliminated by de-image filter $H_{2a}(\Omega)$ before entering the ADC. As for the digital signal processing part, in order to prevent the image frequency of digital conversion from aliasing to the target frequency, it is necessary to apply an upsampler and anti-aliasing filters $F_1(e^{j\omega})$, $F_2(e^{j\omega})$ before digital mixing. Then the HF band siginal is passed through an upconverter and anti-image filter $F_{2a}(e^{j\omega})$. Finally, the reconstruction filter bank $G_1(e^{j\omega})$, $G_1(e^{j\omega})$is applied to realize the waveform reconstruction.

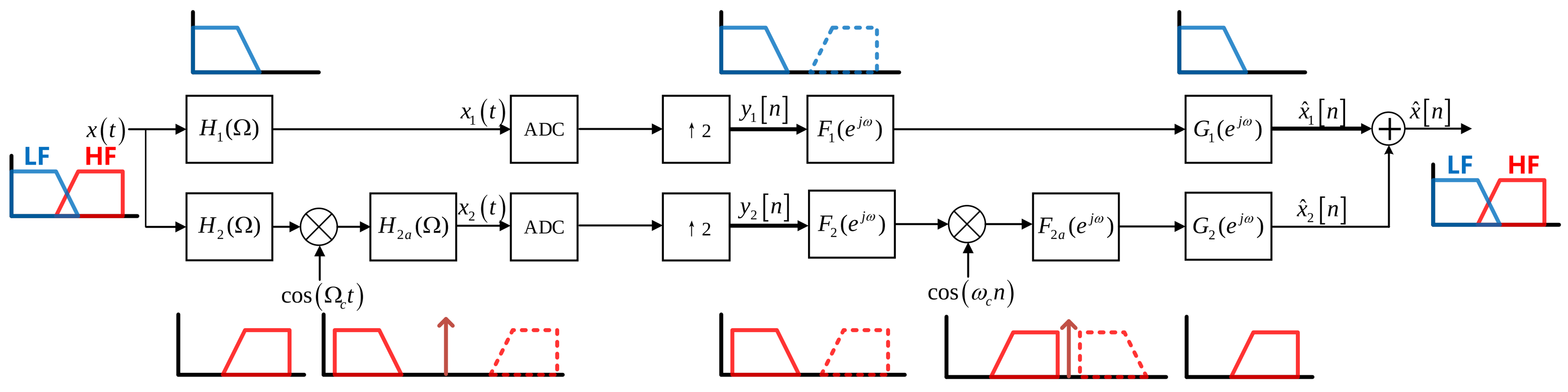

Fig. 1. Two-channel DBI system

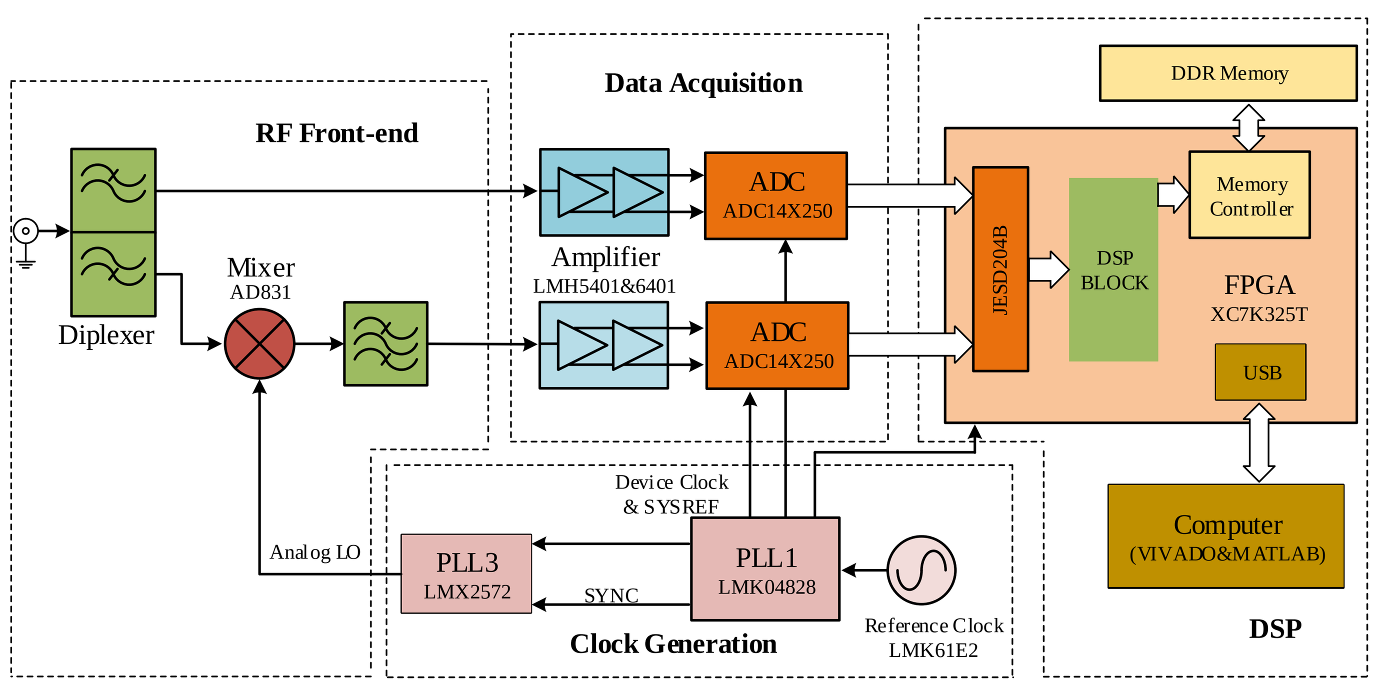

Fig. 2. DBI system evaluation platform

Compared with the traditional time-domain recovery method, the proposed method performs most of the signal processing in the frequency domain. It first converts discrete time-domain signals into discrete frequency-domain signals through Fast Fourier transform(FFT), and then completes digital mixing, filtering, and corrections in the frequency domain. Finally, the reconstructed frequency domain components is converted back into a time domain signal by Inverse Fast Fourier transform (IFFT).

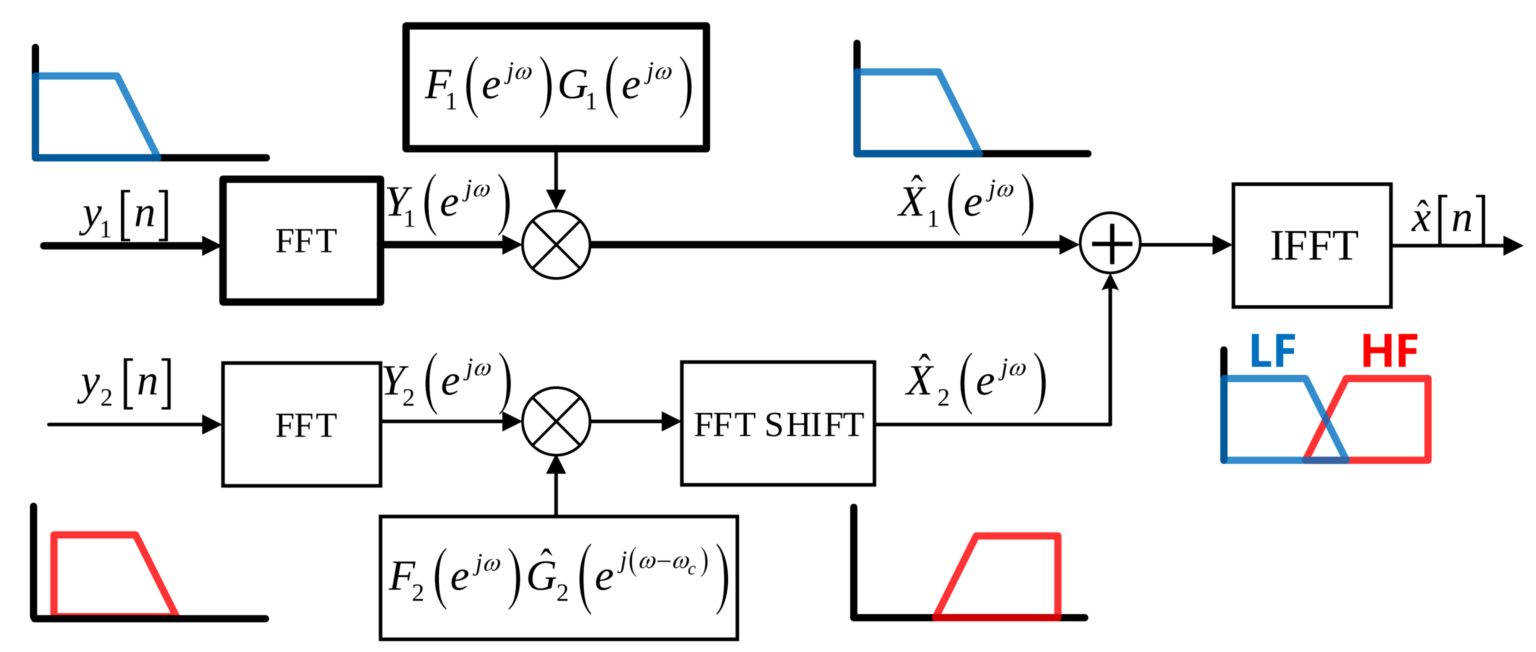

Fig. 3. Proposed waveform recovery method

Test Results and Discussion

To illustrate the efficiency of the proposed method. We compare the resource utilization of the two method in the FPGA. Furthermore, under different word length conditions, the reconstructed signals' SNR, SFDR are measured to compared the performance of the two methods. The number of available DSPs in FPGA XC7K325T is 840, the classical time-domain method consumes 643 DSP resources, and the proposed method is only 127. So the proposed method saves 61.43% of the system's DSP resources. When the filter coefficient word length is 6 bits, Fig. 5 shows the difference in waveform reconstruction quality between two methods. The result shows that the SFDR of the proposed method is more than 20 dB higher than the classical method. The SNR of the proposed method is more than 4.1 dB higher than the classical method. Fig. 5 shows differences in SNR and SFDR of reconstructed signals by two methods under different word lengths. As word length increases, the gap between time domain methods and the proposed method gradually narrows.

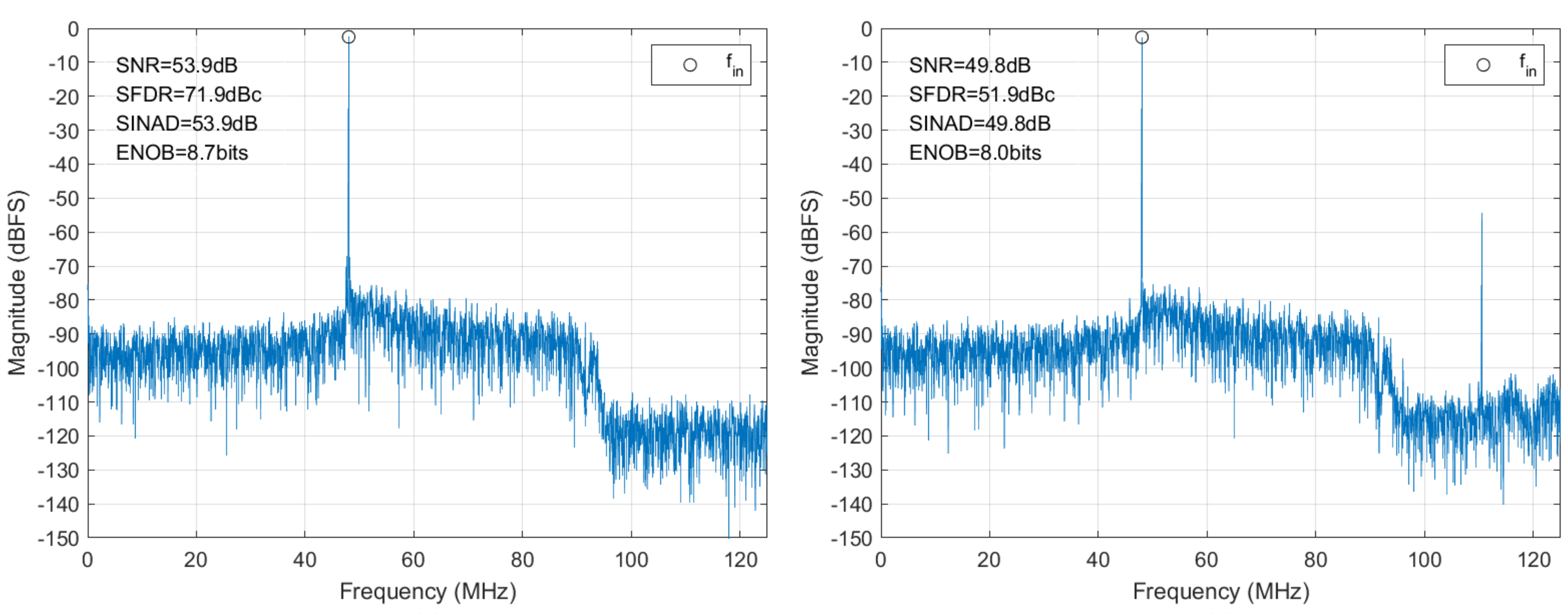

Fig. 4. Single tone test spectrum(6 bits word length): (a) 48MHz input signal for proposed method; (b) 48MHz input signal for classical method

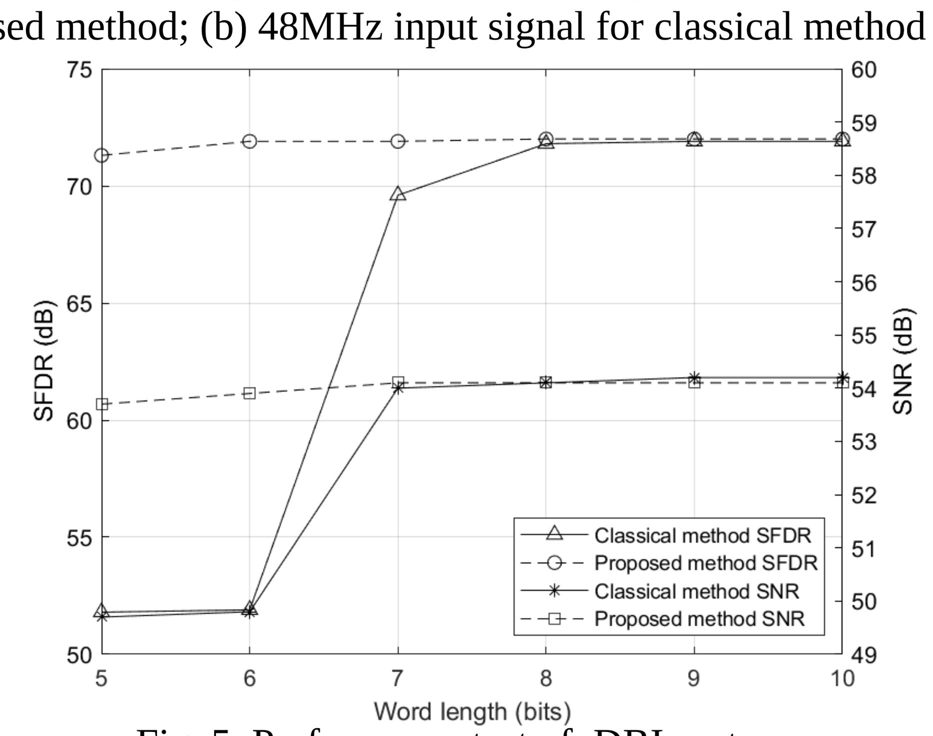

Fig. 5. Performance test of DBI system

 Author e-mail: myb692@mail.ustc.edu.cn