Frequency Domain Approach for CMOS Ultra-Wideband Radios

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Abstract: Ultra wideband (UWB) is a promising new communication scheme for short-range, high data rate applications. Processing UWB signals necessitates impractically high sampling rates for analog-to-digital converters. In this paper, we propose a new approach to address this problem.

I. Introduction

When compared to traditional narrow band communication systems, ultra wideband (UWB) technology has several advantages such as high data rate, low-radiated power, excellent immunity to multipaths, and simple hardware. UWB suits many applications such as wireless home networking, sensor network communications, and through-the-walls sensing.

Realization of UWB systems in CMOS technology is highly desirable for most UWB applications, but it poses a great challenge for VLSI designers. The challenge stems from the fact that UWB is based on extremely narrow pulses, which require high-speed circuits. Two major challenges for CMOS implementation of UWB radios are generation of narrow pulses at the transmitter and sampling the pulses at the receiver. Existing CMOS analog-to-digital converters (ADCs) cannot meet the sampling rate necessary for UWB pulses compliant to the FCC spectrum allocation [1]. For example, eight times oversampling of a pulse with a 250 ps pulse width requires an ADC sampling rate of 32 GHz. In addition to the high sampling rate, timing jitter makes precise control of sampling time intractable. Use of a parallel bank of ADCs has been suggested [2]. However, the approach still requires precise timing to control the ADCs while incurring high circuit complexity.

In order to address the above problems, we present a new approach to sampling input pulses, which is based on processing the signals in the frequency domain. Our frequency domain approach enables the ADC for an UWB radio to operate at a low sampling rate, which opens the possibility of CMOS implementation for UWB radios.

II. Frequency Domain Processing

Consider a train of pulses at a receiver as shown in Figure 1. Our goal is to obtain sampled values of the individual pulses using a low speed ADC. Let \( \tau_0 \) be the observation window of each pulse and \( T \) be the pulse repetition period. Assume that pulses require over-sampling by a factor of \( N \). The conventional method, which over-samples a pulse by a factor of \( N \) during \( \tau_0 \) with a single ADC, requires the ADC to sample at a rate of \( N/\tau_0 \). For example, the sampling rate is impractical 32 GHz for \( N=8 \) and \( \tau_0=250 \) ps.

![Figure 1: Received Input Pulses](image)

The architecture of the proposed sampling circuit is given in Figure 2. The received signal \( r(t) \) is applied to a bandpass filter bank, which outputs the frequency spectrum of \( r(t) \). The center frequency of a bandpass filter is \( n\omega_0 \), where the fundamental frequency \( \omega_0 = 2\pi/\tau_0 \), and \( n = 1, 2, \ldots N \) for \( N \) filters. The outputs of the filters are sampled simultaneously at the end of the observation period by \( N \) sample-and-hold units. Note that the sampling rate of the sample-and-hold unit is \( 1/ T \). The ADC sweeps through each filter output once every \( T \) seconds, and hence the sampling rate of the ADC is \( N/T \). The ADC output for each filter is applied to an
N-point Inverse Fast Fourier Transform (IFFT), and the N outputs of the IFFT correspond to the N sampled values of the input pulse during the observation period $\tau_0$.

As noted earlier, the sampling rate of an ADC in the conventional method is $N/\tau_0$, while the sampling rate of the proposed method is $N/T$. Thus, the sampling ratio for the proposed method is reduced by $T/\tau_0$ when compared with the conventional method. For example, the sampling rate of the proposed method is 80 MHz for a data rate of 10 MHz (i.e., $T=100$ ns, $N=8$, and $\tau_0=250$ ps), while the rate for the conventional method is 32 GHz.

### III. Implementation and Simulation Results

In order to demonstrate the feasibility of the proposed approach, we have implemented a bank of five bandpass filters, in which each filter consists of a pair of passive LC filters, one for the real term and the other for the imaginary term. (So two ADCs are necessary to sample the two outputs.) The fundamental frequency $\omega_0$ of the filter bank is set to 1 GHz, which implies the observation time $\tau_0$ is 1 ns. The IFFT has 128 taps, so that the 123 input points for the 128-point IFFT were padded with 0. The pulse repetition rate $T$ is set to 100 ns. The shape of the input pulses is a Gaussian mono-pulse, whose peak-to-peak width is about 160 ps. Since there are 128 sampling points in the observation window of 1 ns, the pulse itself is over-sampled by a factor of about 41. (Note that the exact over-sampling rate depends on the duration of the pulse, which cannot be determined precisely due to the shape of Gaussian mono-pulses.) Additive Gaussian white noise (AWGN) is added to the original signal to yield an SNR of 0 dB at the receiver.

Our simulation result is shown in Figure 3. To aid the visual inspection of the result, a low-pass filter was added at the output of the IFFT. The dashed line indicates the original pulse, the dotted wavy line is the noisy signal, and the solid line is the recovered signal at the output of the IFFT after lowpass filtering. Our experimental result shows that the recovered signal matches the original signal closely enough to validate our approach. In addition, it rejects out-of-band noise to improve the performance.

### IV. Conclusion

We have investigated a new approach to reduce the sampling rate of ADCs for UWB radios. Our simulation result demonstrates that the proposed method is viable and feasible. Further research is needed in several areas including CMOS implementation of LC filters and optimal system design.

### References


![Figure 2: Proposed ADC Structure](image1)

![Figure 3: Waveforms of the Original and Recovered Signal](image2)