A SUBSAMPLING UWB RADIO ARCHITECTURE BY ANALYTIC SIGNALING

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ABSTRACT

This paper describes a signal processing technique which allows a reduction in the complexity of a transceiver for a 3.1-10.6 GHz Ultra-Wideband radio. The proposed system transmits passband pulses using a pulser and antenna, and the receiver front-end downconverts the signal frequency by subsampling, thus, requiring substantially less hardware than a traditional narrowband approach. By exploring the properties of analytic signals, the system allows hardware reduction and a time resolution finer than the sampling period, which is useful for locationing or ranging applications.

1. INTRODUCTION

UWB technology was approved by the FCC in 2002 [1], and has since drawn considerable attention for a variety of applications, including imaging, surveillance, and locationing. One of the most attractive is for indoor communication systems which are allowed to operate in the frequency band from 3.1 to 10.6 GHz. The interest in these indoor systems extends from high-speed and short-range systems to low data rate communications and precision ranging, as seen in the standard efforts of IEEE 802.15.3a/4a. Regardless of application, it is very crucial to design with low cost and low power; especially as many of these applications intend to deploy a large volume of inexpensive UWB mobile devices that must operate with the longest possible battery life.

The main goal of this work is to improve the efficiency of UWB implementation using signal processing techniques. In particular, an approach will be described that takes advantage of UWB pulses which are narrow in time (with a timescale on the order of a nano-second) to achieve a fine time resolution for ranging. The system architecture will be described in Section 2. Section 3 will explain the challenges of implementing a subsampling front-end and the reasons why it is advantageous to UWB implementation. Digital analytic signal processing will be highlighted in Section 4. Finally, Section 5 discusses synchronization, detection and resolving timing information using the proposed system. The system simulation results are presented which use pulses and noise from actual measurements.

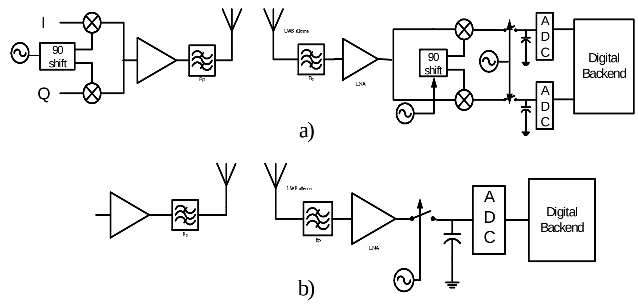

Figure 1. Transceiver of (a) one stage heterodyne (b) subsampling radio.

2. TRANSCEIVER ARCHITECTURE

The proposed system follows the concept of impulse radio [2][3] rather than heterodyne transceiver, the most popular architecture in narrowband systems. Shown in Figure 1, a heterodyne transmitter modulates the baseband information onto a sinusoidal carrier through analog mixers. In contrast, an impulse radio simply uses a pulser to drive the antenna, and radiates a passband pulse (between 3.1-10.6 GHz) shaped by the impulse response of the wideband antenna. The hardware elimination of mixers and local oscillators imply lower complexity implementations.

On the receiver side, the heterodyne system utilizes one or two mixing stages to downconvert the passband signal back to baseband prior to the ADC. The proposed receiver directly samples the incoming signal after amplification. The sampled data will be processed by a digital matched filter in order to reach the matched filter bound for optimal detection [4]. The proposed system avoids wideband analog processing by adding more processing to the digital backend, which results in a more efficient solution.

3. ANALOG: SUBSAMPLING FRONT END

Subsampling directly samples the passband signal at twice the signal bandwidth instead of the maximum signal frequency. The signal is bandlimited from F_l to F_h (Hz), and the sampling frequency is F_s (Hz), as shown in Figure 2. By carefully choosing F_s, F_l and F_h, a non-aliased sampled spectrum can be derived [5]. In order to avoid aliasing and sample at the minimal possible rate, 2B, the lower or upper frequency bound, i.e. F_l and F_h, has to be a non-negative integer multiple of the signal bandwidth, B.

$$F_l = k \cdot (F_h - F_l) = k \cdot B, \text{ where } k \in \mathrm{N}$$

The integer k will be called the undersampling ratio.

There are several key challenges about subsampling that prohibit it from being popular in existing narrowband systems. First of all, the noise spectrum from $-F_l$ to $+F_l$ will alias into the signal band and thus deteriorate the passband SNR, assuming the receiver bandwidth is F_h. Even if the receiver can afford a perfect anti-aliasing filter, the circuits after the filter, such as the sample and hold, still contribute thermal noise. Therefore, if the receiver noise is dominated by these circuit noise sources, the passband SNR degradation is proportional to the undersampling ratio, which can be on the order of hundred's for a narrowband system. Secondly, it is more difficult to design a good bandpass filter (anti-aliasing) at RF than IF or baseband. As mentioned above, less attenuation to out of band noise causes more degradation in signal SNR. Traditionally, these high Q bandpass filters are implemented off-chip, which unavoidably increases the cost and power consumption. Finally, the incoming signal frequency is actually higher than the sampling frequency; therefore the receiver's tracking bandwidth needs to cover the maximum signal frequency range and sampling jitter causes severer degradation of the SNR. If one models the jitter-induced error as another noise source, the noise power increases with input signal frequency [6]. Therefore, directly sampling at RF frequency introduces more noise power than IF or baseband frequency.

Interestingly, the challenge of subsampling is relaxed in the UWB case. For example, if a 1 GHz wide pulse is transmitted between 3 to 4 GHz, and sampled at 2 Gsa/s, then the undersampling ratio is only two, much smaller than a narrowband system. Moreover, as the UWB signal has such a wide bandwidth; at least 500 MHz by FCC regulations, the noise spectrum will be most likely dominated by in-band ambient noise and interference received from other wireless systems, as illustrated in Figure 2. Therefore, the SNR degradation due to aliasing will be much smaller compared to narrowband systems. Next, the difficulty of implementing a RF bandpass filter is also relaxed by the lower filter Q, which is less than ten in this system, simplifying the integration of a CMOS implementation. Lastly, the sampling jitter constraint will be reduced by the lower ADC dynamic range since the received SNR is low due to large in-band noise and interference.

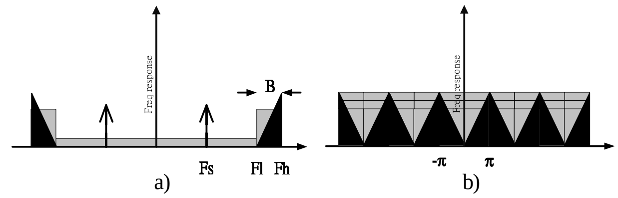

Figure 2. Signal and noise spectrum (a) before and (b) after subsampling. The black color is wanted signal, and the gray one represents noise.

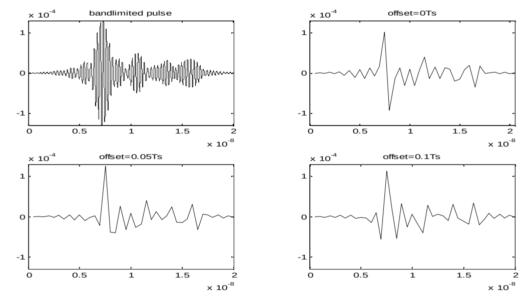

Figure 3. Bandpass pulse (top left) and subsampled waveform with 0*Ts* (top right), 0.05*Ts* (bottom left), 0.1*Ts* (bottom right) sampling offset.

While most of the challenges of subsampling become reasonable for UWB, the architecture still suffers from sampling offset, which can be introduced by frequency mismatch between the TX and RX oscillators or changes of pulse arrival times. The sampled waveform will change dramatically due to this sampling offset, which can deteriorate the SNR of matched filter outputs because of the mismatch between the incoming pulse shape and matched filter response. The impact of the matched filtering mismatch on system performance was studied in [7] and showed a vulnerability to the sampling offset, which will be called timing sensitivity. In fact, the timing sensitivity is more significant in subsampling than Nyquist sampling given the same sampling frequency, since the variation in the waveform significantly complicates the design of digital matched filter. Using measured UWB pulses that are bandlimited to 3-4 GHz, the impact of sampling offset on sampled waveform is shown in Figure 3. An analytic signaling approach described in the next section is thus proposed to alleviate this problem.

4. DIGITAL: ANALYTIC SIGNAL PROCESSING

The main function of the digital backend is to perform synchronization and data recovery. In narrowband communications, synchronization is comprised of timing recovery and carrier frequency compensation. Since there is no sinusoidal carrier in an impulse radio, the synchronization issue becomes only timing recovery. Timing recovery is typically done via oversampling or interpolation [8]. But, the cost of these approaches in

UWB radio is very high due to the large bandwidths involved.

By understanding the relationship between sampling offset and the sampled waveform, the solution to reduce timing sensitivity will become clear. If a bandlimited signal, s(t), is sampled at 1/*Ts*, any sampling offset, *To*, of the sampling sequence will transform into a phase shift in frequency domain, as seen,

$$s(t)\cdot\sum\delta(t-k\cdot Ts-To)\overset{F.T.}{\Leftrightarrow}S(\omega)*\sum\delta(\omega-k\frac{2\pi}{Ts})\cdot e^{-jk\frac{2\pi}{Ts}To}$$

Since the sampling offset is only a phase shift, a naïve approach would be to throw away the phase term by calculating the magnitude of signal's FFT. In other words, a power spectral density matched filter can be more insensitive to the sampling offset. However, the scheme is far from optimal particularly when the sampling offset is small due to the loss of phase information. Therefore, a better approach needs to be proposed for achieving both optimality and timing insensitivity.

Without the loss of signal information, energy detection can also be decoupled from the phase shift, if we formulate an analytic signal. The real and imaginary parts of the analytic signal are orthogonal, and the phase information can be studied on the Euler plane. As sampling offset varies, the signal energy moves between these two orthogonal dimensions. In the next section, we will show that with appropriate usage of the analytic signal, one can reduce the timing sensitivity or resolve timing information. We will also compare the results to real-valued signal processing.

Figure 4 is the block diagram of the proposed digital backend. The main components are pulse shape estimator, analytic signal transformer, correlators, analytic matched filter, and detection block. The first task of signal detection is to determine the matched filter response through the pulse shape estimator. It requires a training sequence to learn the received pulse shape. Assuming the training period is within the channel coherence time, it is shown that doing a running average is the best unbiased linear estimator to estimate the mean of the received signals, i.e. the pulse shape, in the sense of minimizing mean square error [9]. The correlation block is used to provide additional processing gain or despread any possible coding that is modulated on the pulses.

The analytic matched filter response is similar to a real-valued matched filter except with complex conjugation on the time-reversed signal. The filter response is,

$$h_{MF}(t)=s^*(T-t)$$

Therefore, the analytic matched filter takes the pulse estimator results and convolves with the incoming analytic signals. The outputs of matched filter are then followed by a detection block, which is used for synchronization and data recovery. A Coordinate Rotation Digital Computer (Cordic) block can be used to calculate the phase and magnitude of the complex signal.

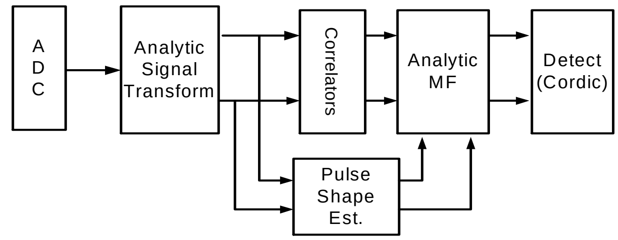

Figure 4. Digital backend block diagram.

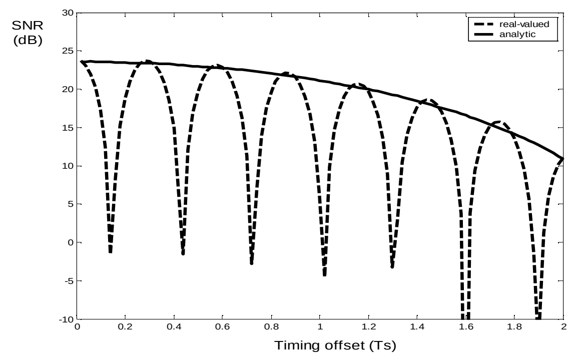

Figure 5. SNR comparison of real-valued processing (dashed) and the magnitude of analytic signal processing (solid) with 0-2*Ts* timing offset.

5. PERFORMANCE EVALUATION

A system simulation is used to evaluate the performance and illustrate how to achieve synchronization, detection; and ranging capabilities based on the proposed system. The simulation takes measured pulses generated from a pulser and TEM horn antenna, whose frequency response is flat between 3 to 10 GHz. For interference and noise, sixty million samples were acquired by the TEM horn antenna and Agilent DSO (54855A), which is capable of sampling at 20 Gsa/s. The measured pulse shape and noise samples were post-processed in Matlab. The pulse is bandlimited to 3-4 GHz and subsampled at 2 Gsa/s.

5.1. Performance Comparison of Real-Valued and Analytic Signal Processing

As illustrated by Figure 5, the output SNR of the real-valued matched filter has periodical null points as the timing offset increases; while, the magnitude of the analytic filtering approach avoids null SNR points. In other words, the vulnerability to the timing offset is thus reduced through analytic signal processing; therefore, the limitations of using a subsampling front-end are alleviated.

The result can be explained from Figure 6, where the analytic matched filter outputs are plotted on Euler coordinates. 10,000 experiments were simulated with and without signal existence. Each graph differs only in sampling offset. The results show that an offset of even 5% of the sampling period could rotate the complex signal about 30 degrees. If a real-valued matched filter was used rather than an analytic one, the real-valued matched filter output is essentially the projection onto the real axis. The reason why timing sensitivity is particularly high for subsampling can be understood by the phase shift term, exp(-2πk*To*/*Ts*). The rotation angle is proportional to the ratio of *To*/*Ts*, and undersampling ratio, k. Note that the Euclidean distance between the two clouds remains about the same, as opposed to the real axis projection always going to zero when the signal energy resides in imaginary component. Thus, the magnitude of analytic signal avoids nulls with respect to timing offset.

5.2. Usage of Analytic Signal for Timing Information

For the purpose of synchronization or data recovery, the timing sensitivity should be kept as small as possible. However, for ranging purpose, the high sensitivity to timing offset implies a high time resolution for the system. As the timing offset can be caused by pulse delay, we may utilize this to measure the distance between transmitter and receiver. In order to determine the pulse delay, the receiver must learn the oscillator mismatch in the training sequence. Also notice that the high time resolution is derived from the following: the wide bandwidth of UWB signal, the interpolation effect from the center oscillation frequency, and the phase information extracted from analytic signal processing. Therefore, the higher the frequency band that UWB pulses use, the finer the time resolution this system can achieve. Finally, the accuracy of ranging depends on how fine the system can resolve the angle, which is directly related to the SNR of analytic matched filter output. In this particular example we used, 90-degree phase shift corresponds to one-inch accuracy.

6. CONCLUSION

A subsampling analog front-end combined with analytic signal processing has been proposed for passband UWB communications. The architecture minimizes the building blocks for a low-complexity implementation with the potential for full integration. By exploiting the analytic matched filter output, timing recovery can be done without oversampling or interpolation. The proposed system also achieves a high time resolution, which implies high accuracy ranging capability.

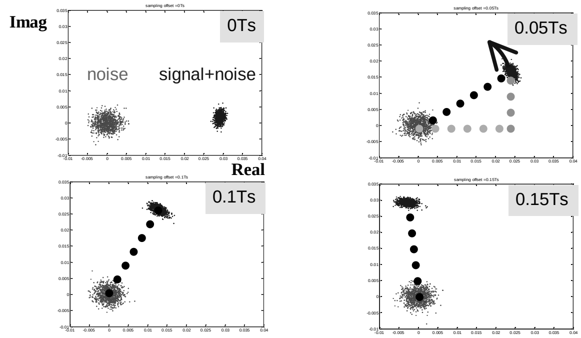

Figure 6. Plots of analytic matched filter outputs corresponding to {0, 5, 10, 15}% of *Ts* timing offset.

7. ACKNOWLEDGEMENT

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