Researches on the Wideband Spectrum Sensing Prototype System Based on MWC

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***Abstract*—Recently, researches based on the Compressed Sensing (CS) become an important direction in the field of signal processing. The newly proposed system called Modulated Wideband Converter (MWC), which breaks through the limit of traditional Nyquist sampling, has presented huge advantages in sub-Nyquist sampling technique. It is composed of several analog channels that alias the signal intentionally before sampling it at a low rate. Then the signal can be reconstructed from the samples by using some commonly used algorithms in compressed sensing theory. It is one of the most successful compressive sampling hardware architectures, but in real hardware implementation, the digital board is fairly difficult to realize. In this paper, we made exploration of the MWC sampling theory and designed a three channels prototype system. To realize the ± 1 logic waveform, we present using Rocket I/O on Xilinx FPGA. In this way, the hardware system can be simplified so as to implement quickly. Through hardware experiments and algorithm reconstruction, we verify the feasibility of MWC system in sub-Nyquist sampling theory.**

***Index Terms*—compressed sensing, sub-Nyquist sampling, modulated wideband converter, prototype system, FPGA**

I. Introduction

With the advancement of modern communication technology, the demand for spectrum resources is growing heavily. However, the signal bandwidth which carries the information becomes more and more wide. Under Shannon/Nyquist sampling framework, the sampling rate should be at least two times the signal bandwidth. A major challenge in wideband spectrum sensing is the requirement of a high sampling rate which may exceed today's best Analog-to-Digital Converters (ADCs) front-end bandwidths. In general, for the perspective of frequency domain, the communication signal only has value within several continuous intervals spreading over a wide spectrum. Besides, we are only interested in a part of the spectrum. In other words, it is usually sparse in frequency domain, which gives opportunity to sample at a low rate. Compressive sampling is an attractive way to reduce sampling rate and data size.

In 2006, E. Candes and T. Tao *et al.* proposed compressive sensing theory [1]. Under this theory system, the sampling rate is no longer determined by the highest frequency of the signal, but decided by the valid information. It means that if the signal is sparse, it is possible to sample at a low rate. In order to achieve this goal，a number of CS structures have been proposed for sub-Nyquist sampling of continuous-time signals in recent years [2], [3]. Such as Random Demodulation (RD), Multi-coset sampling, modulated wideband converter, and so on. However, RD system can only treat multi-tone signal models. But in reality, the analog signals we usually use require a prohibitively large number of harmonics to approximate them well within the discrete model, which in turn renders the reconstruction computationally infeasible and very sensitive to the grid choice [3]. Multi-coset sampling system has a good performance of the multiband signal, but this scheme is fairly hard to implement. This structure has two limitations. First, one analog front-end must fit all the signal bandwidth. Second, maintaining accurate time shifts, on the order of the Nyquist interval, is extremely difficult to implement [4]. In 2008, Mishali *et al.* proposed MWC sampling structure [5]. By using a special analog front-end, the signal can be reconstructed from low rate samples, using the known relation between samples and the original signal. It's the most striking structure and can be realized by commercial components. However, the original architecture has poor results owing to its high complexity. We introduce Rocket I/O on FPGA to replace the digital board which is used to generate periodic sequences. By using some existing discrete circuit modules, we have done some simulations with great results.

This paper is organized as follows. In Section II, we present the MWC theoretical background. In Section III, we describe the MWC prototype system. Section IV shows hardware simulation results.

II. Sampling Theory

Assuming the signal model $x(t)$ is a continuous wideband sparse signal as shown in Fig. 1. According to the Nyquist sampling theorem, its Fourier transform is $X(f)$, and the bandwidth of the signal is then $\left[-f_{nyq}/2, f_{nyq}/2\right]$, in which f_{nyq} means Nyquist sampling rate of the signal. The spectral range of the input signal is $\left[0, f_{nyq}/2\right]$, which contains several narrowband primary signals. The center frequency of

Manuscript received December 2, 2016; revised April 4, 2017.

doi: 10.18178/ijsps.5.2.70-74

each signal is unknown, but its possible maximum bandwidth is B. The number of primary signals is assumed to be no more than $N/2$ where N is a positive even number. To use the compressed sensing theory, we must make sure that neighboring signals are disjointed and total bands are far less than the Nyquist rate. That is $NB \ll f_{nyq}$.

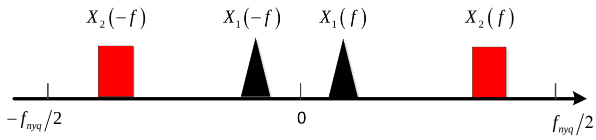

Figure 1. The schematic diagram of multiband signal.

As depicted in Fig. 2, the MWC system consists of an analog front-end with m channels, where each channel is made of a mixer, a low-pass filter and a low-rate ADC. The original signal enters several channels simultaneously. In the ith channel, the input signal is multiplied by a periodic waveform $p_i(t)$ and its period is f_p. The output of the mixer is low–pass filtered by $h(t)$, and then sampled at a low rate $f_s = 1/T_s$. The periodic waveform and the frequency response of the low-pass filter are shown in Fig. 3.

In [6], the frequency band we sensing is divided into L consecutive channels. The bandwidth of each channel is no more than B, and the channel number is within $[1, L]$. The number called support is the key result we want to get.

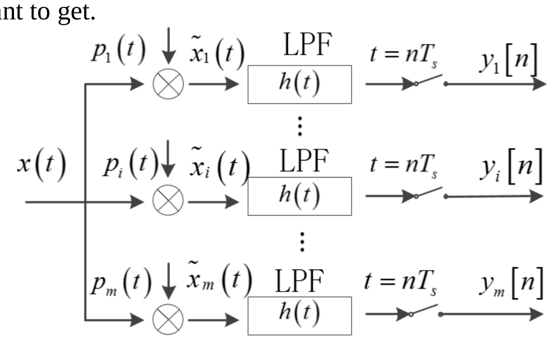

Figure 2. The Schematic Diagram of Modulated Wideband Converter.

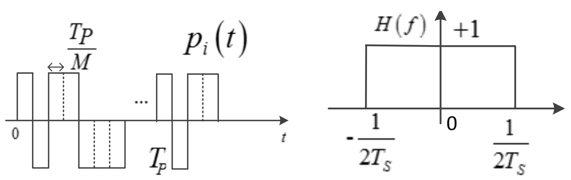

Figure 3. Periodic Waveform and Frequency Response of LPF.

Since $p_i(t)$ is periodic, its Fourier expansion is

$$p_i(t) = \sum_{l=-\infty}^{\infty} c_{il} \exp\left(j2\pi lt / T_p\right) \tag{1}$$

in which $c_{il} = \frac{1}{T_p}\int_0^{T_p} p_i(t)\exp(-j2\pi lt / T_p)dt$. The spectrum of mixing signal $x_i(t) = x(t)p_i(t)$ is

$$\tilde{X}_i(f) = \sum_{l=-\infty}^{\infty} c_{il} X(f - lf_p) \tag{2}$$

where $X(f)$ denotes the spectrum of the input signal.

After passing the LPF, it is sampled at the rate f_s, the DTFT of the samples $y_i[\mathrm{n}]$ is

$$Y_i(e^{j2\pi fT_S}) = \frac{1}{T_s}\sum_{l=-L_0}^{L_0} c_{il} X(f - lf_p) \tag{3}$$

For our purposes, it's convenient to write (3) in matrix form as

$$\mathbf{y}(f) = \mathbf{Az}(f), |f| \le f_s / 2 \tag{4}$$

where, $\mathbf{y}(f)$ is a vector of length m with ith element $y_i(f) = \mathrm{Y}_i\left(e^{j2\pi fT_s}\right)$. The unknown vector

$$\mathbf{z}(f) = [z_1(f), \ldots, z_L(f)]^T, L = 2L_0 + 1 \tag{5}$$

with

$$z_i(f) = \mathrm{X}(f + (i - L_0 - 1)f_p), 1 \le i \le L, |f| \le f_s \tag{6}$$

The $m \times L$ matrix $\mathbf{A}$ contains the coefficients c_{il}

$$\mathbf{A}_{il} = c_{i,-l} = c^*_{il} \tag{7}$$

Equation (7) is called Multiple Measurement Vectors (MMV) problem. The algorithm solving this is to reconstruct the unknown multiband signal $x(t)$ (or its discrete denotation $\mathbf{z}(f)$) from the measured data $\mathbf{y}(f)$ and the measured matrix $\mathbf{A}$. In [6], a continuous-to-finite (CTF) block has been proposed to recover the support S of the sparse signal $\mathbf{z}$ first. Then it can be easily calculated by $\mathbf{z}_S = \mathbf{A}^{\dagger}{}_S\mathbf{y}, (\mathbf{z}_i = 0, i \notin S)$, where $\mathbf{z}_S$ and $\mathbf{A}_S$ are the submatrices comprised of the rows of $\mathbf{z}$ and $\mathbf{A}$ indexed by S. $(.)^{\dagger}$ denotes the pseudo inverse. Finally, we reconstruct the signal $x(n)$.

III. MWC Prototype System Design

A. Hardware Specifications

The system we design is a verification of the theory. To simplify the design process of hardware, we make use of several modules in hand to implement. We next motivate these choices.

The MWC is a flexible system with various parameters, of which we chose to realise the specifications that appear in Table I.

TABLE I. Prototype Specifications

Paraments Symbol	
$N = 4$	Number of signal frequency band
$B = 0.25MHz$	Signal band upper limit
$f_{nyq} = 50MHz$	Nyquist rate of signal
$m = 3$	Folding coefficient
$q = 5$	Number of signal frequency band
$M = 195$	Nunnber of pseudo random code
$f_s = 1.28MHz$	AD sample rate

The multiband signal model, we assume $N = 4, B = 0.25MHz$. The sensing bandwidth is $25MHz$, The Nyquist rate $f_{nyq} = 50MHz$, it also means the clock rate of periodic waveform. In [7], the basic parameter choice is $m \geq 4N$ and $f_s = f_p \geq B$. This means the total sampling rate mf_s is significantly smaller than f_{nyq}. However, if we choose $m = 4N = 12$, the hardware size is pretty big. In order to save size and price, also in [6], in which the physical channels is collapsed by a factor q. To achieve this, the expense is increasing the sampling rate of each channel by the same factor. In our design, we set $q = 5$, and the total sampling rate is $mf_s = 3.84MHz$, which is about 7.68% of the Nyquist rate.

B. Circuit Design

In designing an analog circuit to realize the MWC in [6], they encountered two main difficulties. One is analog mixing with spectrally rich waveforms $p_i(t)$, and another is constructing the periodic waveforms with the required alternation speed of 2.075 GHz. They design a complex system divided into two parts. The digital board aims to generate periodic waveform, and analog board achieves the rest.

Different from the design in [6], our MWC prototype system consists of analog front-end, analog mixer module, periodic sequences generating module, ADC module and PC data processing module. In this system, the analog front-end is used to preprocess the input signal. Then the three-channel analog mixer module is used to mix the input signal and periodic wave form which is generated by FPGA. Because of the data synchronization, we use another channel to generate synchronous clock. The low-pass filtered signal is sampled and transferred to PC processing system. We next demonstrate all modules of the prototype system.

Fig. 4 presents a block diagram of the whole system. For the first difficulty referenced above, $p_i(t)$ spans a wide spectrum. So the mixer should allow a wide range of LO frequencies. We choose the wideband active mixer chip AD831 produced by Analog Device. This mixer includes an LO driver and a low noise output amplifier. Besides, its 500 MHz RF and LO Input Bandwidths is enough for our design. It provides both user-programmable power consumption and third order intercept point.

For the second difficulty, so as to generate high rate periodic waveforms, a straightforward approach to realise this would be to program the desired pattern into a field programmable gate array (FPGA) device. Unfortunately, this approach is difficult for realising a high-rate SR. Conventional FPGA devices do not stand a clock rate of GHz. In [6], they exploited a specific property of shift-register. This way uses 96 shift-registers, and they are connected in series becoming a shift-register circle. Before using this digital board, they set the initial value of each register. Then four channels of sequences output from four different taps. However, this way can generate sequences, but the sequences has high correlation, which may leads to reconstruction failure.

Recently, Rocket I/O GTZ transceivers are capable of running up to 28.05 Gb/s [7]. Based on this, we come up with an idea to use several transceivers to transfer the sequences without receivings. In our previous work, we use GTX module on Virtex-5 FX30T to generate four channels of sequences, the waveform of which is so good that more than what we expect. But now, in our experiment, we don't need such high rate, so we choose Xilinx nexys 3 board instead. The sequences are written in a virtual ROM firstly. Then the board circularly outputs the sequences. Beside, using FPGA has many other advantages, especially when we want to change the sequence value. It is proved that Gold, Kasami and even randomly chosen binary patterns are suitable for the MWC system. We finally choose pseudorandom sequences generated by MATLAB.

Finally, the low-pass filter is the last module of the MWC. As we know, the input to the filter contains energy spread all across the spectrum, with non-eligible power beyond the cut-off. For this reason, we realised a sharp cut-off around 0.6 MHz using two elliptic filters.

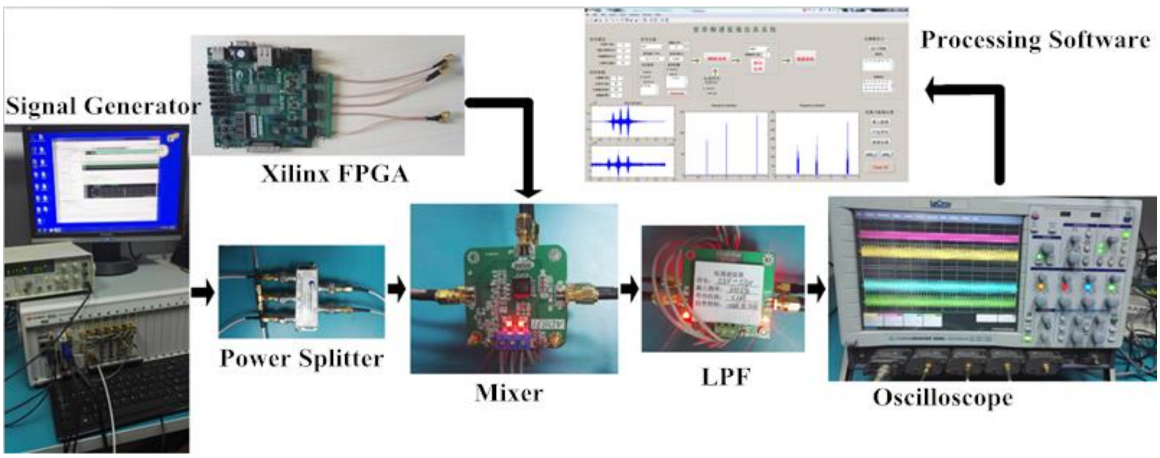

Figure 4. Photos of prototype system

IV. EXPERIMENTS

As seen in Fig. 4, The RF input $x(t)$ is generated using the Arbitrary Wave Generator (AWG)-Keysight M8190A and EE1641 signal generator. The M9180A generates ASK (Binary Amplitude Shift Keying Modulation) of which the bandwidth is 0.2MHz and the carrier is 7.5 MHz. The EE1641 generates single tone signal, and the carrier is 2.45 MHz. Firstly, these two signals are combined, so the $x(t)$ is a two bands signal. Its frequency and time domain is shown in Fig. 5 which is sampled by Lecroy 8500A oscilloscope.

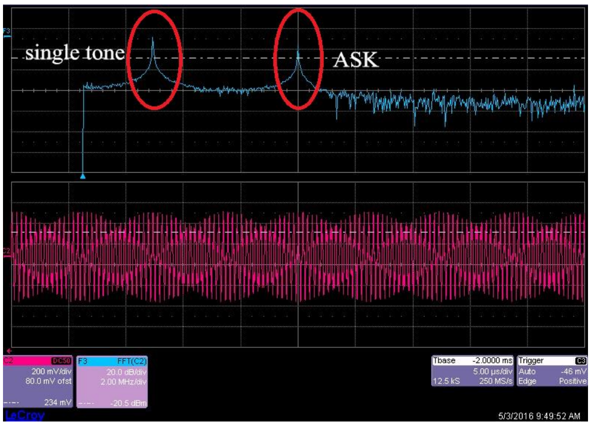

Figure 5. Spectrum and time-domain appearance

Similarly, the $p_i(t)$ and synchronizing sequences are sampled and shown in Fig. 6 and Fig. 7.

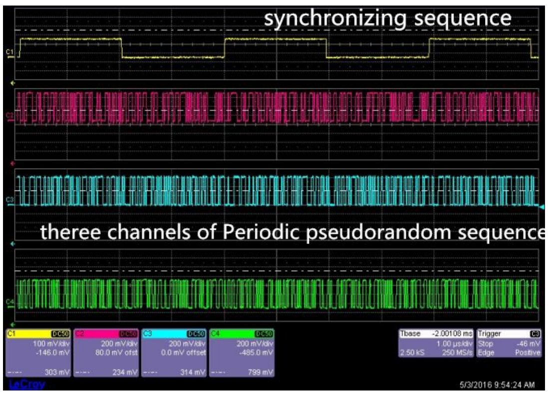

Figure 6. Time-domain appearance of sequences

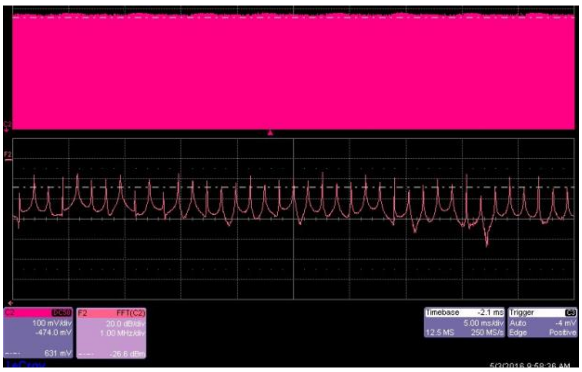

Figure 7. Spectrum of period pseudorandom sequences

As can be seen from Fig. 7, the spectrum of $p_i(t)$, consists of highly concentrated energy peaks called Diracs, as expected for periodic waveforms. The spacing between the Diracs was measured as 0.256MHz, validating the design choice of f_p. Periodicity is the only essential requirement of the MWC system, so we don't need to generate ideal waveform of sequences.

The signal is then low-pass filtered and sampled. Because of the non-ideal analog components such as the LPF filters, we compensate for it using a digital FIR correction scheme [8]. In order to simplify the design process, we use LeCroy 8500A oscilloscope to sample three channels of the output signal. The sampling rate we set is integer multiples of f_s, then we sample at digital-domain to guarantee the actual sampling rate equals to f_s. Finally, it comes to reconstruction. We process the data sampled before on MATLAB 2011a. For reconstruction, OMP (Orthogonal Matching Pursuit) algorithm [9] is used because of its quick speed and high accurate. The spectrum of reconstructed signal is shown in Fig. 8.

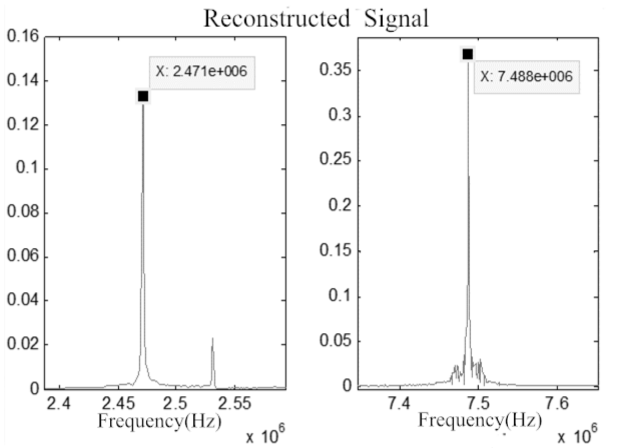

Figure 8. Spectrum of reconstructed signal

As can be seen, although beside the single tone signal appears a clutter, the accuracy of signal frequency is pretty high. To verify the theory, we have done plenty of experiments and its frequency accuracy is stable. Reconstruction proof for a sampling strategy is a certificate that the theoretical principles are sufficiently robust to accommodate circuit non-idealities. And the cost is just 7.68% of traditional Nyquist sampling rate.

V. CONCLUSION

In this paper, we deeply investigated the MWC sampling theory and designed a hardware system prototype. We introduced to use Rocket I/O to generate periodic pseudorandom sequences for the hardware implementation of the MWC. This way is easy to implement and configured flexibly. Experiments show that the prototype system has a good behavior. We are able to recover each band of signals from very low rate samples, at only 7.68% of their Nyquist rate.

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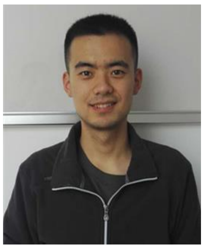

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