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Compensation of RF impairment in multi-band receiver based on RF sub-sampling

HuiKyu Lee*, Heung-Gyoon Ryu

Department of Electronic Engineering, Chungbuk National University, Cheongju, Republic of Korea

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ABSTRACT

This paper is studied about multi-band receiver using sub-sampling and time division multiplexing (TDM) techniques. Software defined radio (SDR) has a goal that places the analog to digital converter (ADC) as near the antenna as possible. But current technique actually cannot process ADC about radio frequency (RF) band signals. So one method is being studied that samples RF band signals to intermediate frequency (IF) band. As one of the ways, sub-sampling technique can convert signals from RF band to IF band without oscillator. If sub-sampling technique is used, over 2 bands can convert signals from RF band to IF band. However, due to the filter performance in RF band, it is possible that interference is generated between signals that are converted to low frequency band. And the problem degrades performance. In this paper, we propose one method that uses TDM technique as a solution to avoid interference between signals. By processing TDM and sub-sampling at the same time, the method can get signals without large changes from the conventional structures.

1. Introduction

A flexible receiver structure is needed to get multi-band and multi-mode signals in next generation. But there are problems to receive multi-band and multi-mode signals in conventional analog system. As a solution, SDR is emerging, which is a method to digitize signals at RF part using ADC [1]. But currently it is difficult because the speed of ADC. So Digital IF method has been studied. That is to digitize signals at IF band after down-converting from RF band to IF band. If signals are digitized in IF band after down-converting with oscillator, it is possible to digitize a multi-mode signal. But it may be difficult that one receiver is to get non-contiguous signals like one contiguous signal. To meet requirement of IMT-Advanced, carrier aggregation (CA) is used in long term evolution-advanced (LTE-A) [2]. CA is used to get up to 100 MHz bandwidth. That is one method that combines carrier component (CC). So in this paper to get some CC, sub-sampling technique is used.

Sub-sampling is a technique that samples high data rate signals with smaller sampling rate than Nyquist sampling rate. There have been studied about sub-sampling [3–5]. Especially, this method is that one sampler down samples some signals that have different center frequency and bandwidth. At that time, interference not occurs between received signals. After down sampling about over 2 band signals using sub-sampling, the signals are digitized, and then over 2 band signals can be received.

Sub-sampling technique need higher speed ADC than heterodyne or homodyne receiver because digitization is done in IF band or near baseband. So it is possible to generate sample timing offset (STO) between digital to analog converter (DAC) and ADC. And Doppler Effect also can be generated in mobile condition. Moreover, higher center frequency and wider bandwidth makes more phase noise. Especially, orthogonal frequency division multiplexing (OFDM) system has a disadvantage of inter-carrier interference (ICI) which is caused by Doppler Effect and phase noise [7].

In this paper, we propose a sub-sampling technique with TDM. In previous system, although over 2 signals can be down-sampling without interference between signals, it is possible to generate interference due to RF filter characteristic. RF filter cannot cut adjacent band signals so the remaining adjacent band signals, undesired signals, can affect desired signals. So we propose sub-sampling with TDM that can avoid previous problems by separating over 2 signals in time.

2. System model

In this paper, we consider two signals that have different center frequency. Transmitted signals are based on OFDM. Eq. (1) is the signals in time domain.

$$x(t)=\begin{cases}\displaystyle\sum_{k=0}^{N-1}X_k^A\exp\left\{j\left(\frac{2\pi k}{N}+f_A\right)t\right\}, & x_A(t)\\ \displaystyle\sum_{k=0}^{N-1}X_k^B\exp\left\{j\left(\frac{2\pi k}{N}+f_B\right)t\right\}, & x_B(t)\end{cases}\tag{1}$$

* Corresponding author.
E-mail address: hanson0403@hanmail.net (H. Lee).

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In this paper, we assume that there are two received bands. Each band has different signals from each transmitter where X_k^A and X_k^B are transmitted signals, respectively. As (1), the signals are represented after inverse fast Fourier transform (IFFT) in time domain. Each band is represented as $x_A(t)$ and $x_B(t)$ in time domain and has center frequencies as f_A and f_B.

Fig. 1 is receiver structure.

Two signals $x_A(t)$, $x_B(t)$ are received through one antenna. After amplification of received signals in low noise amplifier (LNA), each signal passes through RF filter. After band pass filter (BPF), each signal is sampled in time. At this point, our proposed system process sub-sampling and TDM at the same time.

Sub-sampling technique has smaller sampling frequency than one that uses Nyquist rate. Thus, we can receive signals using sub-sampling technique without oscillator. The proposed one also uses TDM technique, when multi-band signals are received. Therefore, the sampling frequency of the proposed one will be more increased than one without TDM technique, the conventional one.

This system processes TDM by sampling $x_A(t)$, $x_B(t)$ alternately and sub-sampling by sampling lower frequency than f_A or f_B. After ADC, sampling frequency of each band is 110 MHz. And using digital oscillator, the center frequency of converted signal is moved to direct current (DC). The Digital oscillator is processed in digital part, so there are no problems that are generated by analog oscillator in receiver. And then the signals are synchronized using synchronization signal. After fast Fourier transform (FFT), the signals are received and then done by frequency domain equalizer (FDE).

3. Offset analysis and compensation

3.1. Doppler Effect

It is possible to have mobility of receiver in mobile communication. So Doppler Effect degrades performance. Doppler Effect is shift in frequency domain and phase rotation in time domain. However, that makes ICI because of broke othogonality in OFDM system.

Signal $x(t)$ is changed like (2) due to Doppler Effect in time domain.

$$x(t') = x(t) \cdot e^{-i2\pi \frac{\varepsilon t}{N}} \tag{2}$$

Transmitted signal $x(t)$ has phase rotation in time domain due to Doppler Effect in time domain. And the transmitted signal $x(t)$ and changed signal $x(t')$ have same sample number.

When signals are affected by Doppler Effect, the received signal is like (3).

$$Y(k') = \sum_{t'=0}^{N-1} x(t') \cdot e^{-i2\pi \frac{t'}{N} k'} = \sum_{t=0}^{N-1} x(t) \cdot e^{-i2\pi \frac{\varepsilon t}{N}} \cdot e^{-i2\pi \frac{t}{N} k'} \tag{3}$$

Transmitted signal $X(k)$'s subcarrier number is k. And the signal that is affected by Doppler Effect has same number of subcarrier. But it is represented as k' to separate the difference of whether there is Doppler Effect. Eq. (3) shows effect of Doppler Effect in time domain and (4) shows it in frequency domain.

$$\begin{aligned} Y(k') &= \sum_{t=0}^{N-1} \sum_{k=0}^{N-1} X(k) \cdot e^{i2\pi \frac{t}{N} k} \cdot e^{-i2\pi \frac{\varepsilon}{N} t} \cdot e^{-i2\pi \frac{t}{N} k'} \\ &= \sum_{\substack{t=0 \\ k=k'}}^{N-1} \sum_{k=0}^{N-1} X(k) \cdot e^{-i2\pi \frac{\varepsilon}{N} t} + \sum_{\substack{t=0 \\ k \neq k'}}^{N-1} \sum_{k=0}^{N-1} X(k) \\ &\quad \cdot e^{i2\pi \frac{(k-k')}{N} t} \cdot e^{-i2\pi \frac{\varepsilon}{N} t} \end{aligned} \tag{4}$$

If signals are affected by Doppler Effect in based on OFDM, the signal is multiplied as three steps. One is IFFT, second is phase rotation, and third is FFT. And the equation is changed as two parts. The one is that one subcarrier affects itself $\sum_{\substack{k=0 \\ k=k'}}^{N-1} X(k) \sum_{t=0}^{N-1} e^{i2\pi(-\varepsilon/N)t}$, and another is that one subcarrier affects the other subcarriers $\sum_{\substack{t=0 \\ k \neq k'}}^{N-1} \sum_{k=0}^{N-1} X(k) \cdot e^{i2\pi((k-k')/N)t} \cdot e^{i2\pi(-\varepsilon/N)t}$, where second is called as ICI.

Normalized Doppler Effect is represented as ε in (5).

$$\varepsilon = \frac{f_d}{\text{carrier spacing}}, \quad f_d = \frac{v \cdot f_c}{c} \tag{5}$$

Normalized Doppler Effect value is divided by carrier spacing. We consider that direction of receiver is heading toward to transmitter. It means that Doppler Effect has the largest scale by direction. And c is light speed, v is receiver speed, and f_c is carrier frequency. In this system, we compensate those problems with pilot and assume that the receiver speed is constant.

The equations of Doppler Effect consider one band because the each band has same effect with different normalized Doppler Effect.

3.2. Compensation of Doppler Effect

The problem of Doppler Effect is compensated with pilot. And we assume that the receiver speed is not dramatically changed.

Phase rotation due to Doppler is like $\sum_{n=0}^{N-1} e^{-i2\pi(\varepsilon/N)n}$ of (4) and n is sample after ADC from time t.

First, phase rotation is estimated using received pilot signals.

$$P(i) = \frac{1}{N_{\text{pilot}}} \left\{ \sum_{n=\text{pilot}}^{N} Y(n) \right\} \tag{6}$$

And then fine P that is average of pilot, where i is index of pilot.

$$\tilde{Y}(k) = \frac{Y(k)_{\text{received}}}{P(k)} \tag{7}$$

To compensate Doppler Effect, it uses pilot. Therefore it just can compensate phase rotation and there is still residual ICI.

3.3. Sample timing offset

A data that has the number of subcarriers of N before IFFT is sent as a signal that has the number of samples of N after IFFT in OFDM system. A signal that has N samples is a symbol. If the symbol is delayed, the symbol affects next a symbol. So, inter-symbol interference (ISI) occurs. But ISI is removed by cyclic prefix (CP).

$$y(t) = h(t) \otimes x(t+\delta) \tag{8}$$

Eq. (8) represents STO δ. If h is 1, STO in frequency domain is like (9).

$$\begin{aligned} Y(k') &= \sum_{t=0}^{N-1} \sum_{k=0}^{N-1} X(k) \cdot e^{i2\pi \frac{t}{N} k} \cdot e^{i2\pi \frac{\delta k'}{N}} \cdot e^{-i2\pi \frac{t}{N} k'} \\ &= \sum_{\substack{k=0 \\ k=k'}}^{N-1} X(k) e^{i2\pi \frac{\delta k'}{N}} + \sum_{\substack{t=0 \\ k \neq k'}}^{N-1} \sum_{k=0}^{N-1} X(k) \\ &\quad \cdot e^{i2\pi \frac{k-k'}{N} t} \cdot e^{i2\pi \frac{\delta k'}{N}} \end{aligned} \tag{9}$$