

Automatic Tracking of Bandpass Sampling OFDM Signals in Mobile Environment for Security Communication Systems

Xin Wang^{1,1}, Heung-Gyoon Ryu¹

¹ Department of Electronic Engineering Chungbuk National University
Cheongju, Korea 361-763
wxzf007@naver.com, ecomm@cbu.ac.kr

Abstract. Angle of Arrival (AOA) estimation is basic procedure required by several monitoring and tracking systems for security applications and other defense purposes. In this paper, a combined OFDM system and bandpass sampling method using Multiple Signal Classification(MUSIC) algorithm for automatic (angle of arrival) AOA tracking is discussed. And we propose a new method that adding (time division multiplexing)TDM with bandpass sampling in the same time to avoid interference due to RF filter characteristics. Also, we consider Doppler effect for the targets' movement in mobile environment and after compensating the Doppler effect with a valid range , the system performances well. Computer simulation results show that the performances of MUSIC spectrum for AOA due to various conditions and demonstrates the accuracy of AOA estimations.

Keywords: MUSIC, AOA, Bandpass sampling, OFDM, Doppler effect

1 Introduction

Countless security and defense systems require an accurate Angle of Arrival (AOA) detection for incoming signals. Smart antenna is one of the possible solutions to increase the channel capacity due to an increase in the number of mobile units and the need for high-speed digital communication in mobile communication. Smart antenna utilizes the beamforming technique to spatially direct the electromagnetic power to an intended mobile unit while spatially null the signal power along other mobile units. The system needs the process of angle of arrival estimation to locate the mobile units before beamforming can be performed. Angle of arrival estimation technology play an important role in enhancing the performance of adaptive arrays for mobile wireless communications[1]. A number of angle of arrival estimation algorithms have been developed. For the most recent ones being MUSCI[2] and ESPRIT[3] algorithms, who both utilizing subspace-based on exploiting the eigen structure of the input covariance matrix and thus requires a higher computation effort. Although ESPRIT needs less computation, the MUSIC algorithm is found to be more stable and accurate[4]. Bandpass sampling can be used for direct down conversion without analog mixers. In practice, the required sampling rate for ADC can be too high to be achieved if the Nyquist sampling theorem is to be satisfied[5]. So we use bandpass sampling which is

a technique that samples high data rate signals with smaller sampling rate than Nyquist sampling rate to relax the demand for ADCs. After down-sampling about over two band signals using bandpass sampling, the signals are digitized and then two band signals can be received [6].

In this paper, we propose a bandpass sampling technique with time division multiplexing (TDM). In previous system, although over two signals can be down-sampling without interference between signals, it is possible to generate interference due to RF filter characteristics. RF filter cannot cut adjacent band signals so the remaining adjacent band signals (undesired signals) can affect desired signals. So we propose bandpass sampling with TDM that can avoid previous problems to separate over two signals timely.

2 System Model

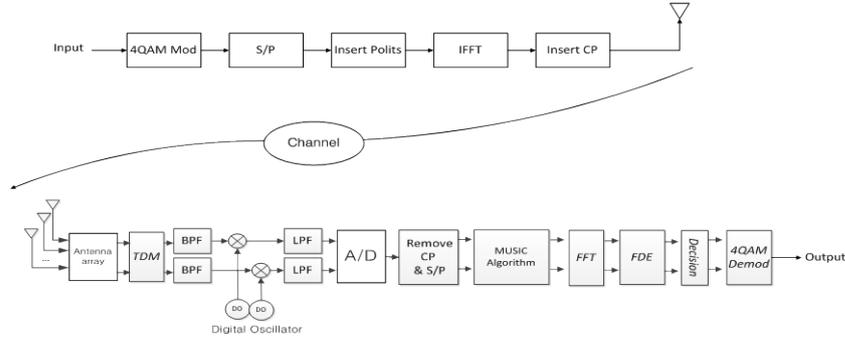


Fig.1. System Model

In this paper we consider two signals that have different center frequency. Transmitted signals are based on OFDM.

$$\mathbf{x}(t) = \begin{cases} \frac{1}{\sqrt{N}} \sum_{k=0}^{N-1} X_{k,m}^A e^{j(\frac{2\pi k}{N} + f_A)t}, & \mathbf{x}_A(t) \\ \frac{1}{\sqrt{N}} \sum_{k=0}^{N-1} X_{k,m}^B e^{j(\frac{2\pi k}{N} + f_B)t}, & \mathbf{x}_B(t) \end{cases} \quad (1)$$

MUSIC stands for Multiple Signal Classification. The covariance matrix, \mathbf{R} , is the collected data for each of the array receivers in the time domain. The correlation matrix is given as[7]

$$\mathbf{R} = E[\mathbf{X}\mathbf{X}^H] = \mathbf{A}\mathbf{R}_s\mathbf{A}^H + \sigma^2\mathbf{I} \quad (2)$$

where R_s is the $P \times P$ signal correlation matrix. σ^2 is the white noise power. The noise subspace E_N used in MUSIC can be obtained from eigenvalue decomposition of R , and the spatial spectrum of MUSIC is given by

$$P(\theta) = \frac{1}{a(\theta)^H E_N E_N^H a(\theta)} \quad (3)$$

3 Doppler Effect and Compensation

The orthogonality among subcarriers is often destroyed by the CFOs due to oscillator mismatches. So Doppler effect was generated and degrades performance. Doppler effects cause shifting in frequency domain and phase rotation in time domain.

$$y_n = \sum_{k=0}^{N-1} H_k \cdot X_k \cdot e^{i2\pi \frac{k+\varepsilon}{N}} + z_n \quad (4)$$

Signal $x(t)$ is like (8) due to Doppler effect in time domain.

Channel H is represented as product of X . Doppler effect is represented phase rotation in frequency domain. k , n , ε are sub-carrier, symbol, normalized Doppler frequency respectively in (5).

$$\begin{aligned} Y_p &= \sum_{m=0}^{N-1} \sum_{k=0}^{N-1} H_{k,m} \cdot X_{k,m} \cdot e^{i2\pi \frac{(k+\varepsilon)}{N}} \cdot e^{-i2\pi \frac{m}{N}} + Z_p \\ &= H_p \cdot X_p \cdot e^{i2\pi \varepsilon p} + \sum_{m=0}^{N-1} \sum_{\substack{k=0 \\ m \neq k}}^{N-1} H_{k,m} \cdot X_{k,m} \cdot e^{i2\pi \frac{(k-m)}{N}} \cdot e^{i2\pi \frac{\varepsilon}{N}} + Z_p \end{aligned} \quad (5)$$

4 Simulation and Discussion

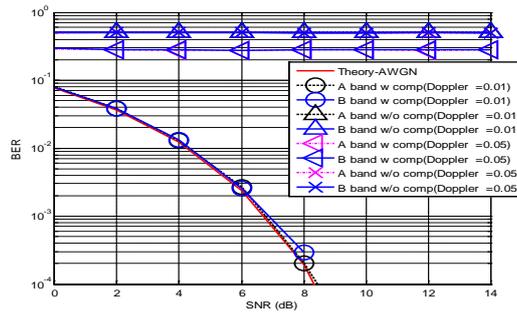


Fig.2. BER performance with Doppler Effect.

Figure 2 indicates BER performance when Doppler effect occurs. We can see the performance according to Doppler scale. The two bands have no difference according to Doppler scale. The two bands have no difference due to TDM. In the case of Doppler effect $\varepsilon = 0.01$, for that both A band and B band without compensating, we can't communicate because of the phase rotation. And after compensating phase rotation, there is small performance degradation comparing to the theory curve because of existing ICI. And when the Doppler Effect is give $\varepsilon = 0.05$, we can't communicate as we use block type pilot and do linear interpolation which is difficult to estimate fast phase rotation.

5 Conclusions

In this paper, we discussed and performance a automatic AOA tracking method using MUSIC algorithm by bandpass sampling method. And we also proposed an adding TDM with bandpass sampling method which can avoid interference. By considering the Doppler effect and compensating the effect, system using proposed method performances well. And simulation shows using MUSIC algorithm to estimate the AOA under different conditions.

Acknowledgments. "This research was supported by Basic Science Research Program through the National Research Foundation of Korea(NRF) funded by the Ministry of Education, Science and Technology(No. 2012017339)"

References

1. Schmidt, R.: Multiple emitter location and signal parameter estimation. In: IEEE Trans. Antennas Propag., vol. AP-34, pp. 276-280,(1986)
2. Paulraj, A., Roy, R.,Kailath, T.: A subspace rotation approach to signal parameter estimation. In: Proceedings of the IEEE, vol. 74, no. 7, pp. 1044 – 1046, (1986)
3. Lavate, T., Kokate,V.,Sapkal, A.: Performance analysis of MUSIC and ESPRIT DoA estimation algorithms for adaptive array smart antenna in mobile communication.In: International Journal of Computer Networks (IJCN), vol. 2, no. 3, pp. 152 – 172, (2010)
4. Walden, R. H.: Performance trends for analog-to-digital converters. In: IEEE Commun. Mag., vol. 37, no. 2, pp. 96-101, (1999)
5. Akos, D. M., Stockmaster, J., Tsui, B. Y., Caschera, J.: Direct bandpass sampling of multiple distinct RF signals. In: IEEE Trans. Commun., vol. 47, no. 7, pp. 983-988, (1999)
6. Wang, J.,Zhao, Y.J.,Wang Z.G.: A MUSIC like DOA estimation method for signals with low SNR. GSSM 2008,pp.321-324, (2008)
7. Tseng,C.H.,Chou, S.C.: Direct Downconversion of Multiband RF Signals Using BandPass Sampling. In: IEEE Trans.Commun, vol. 5, no.1,pp.72-76, (2006)