Automatic Estimation of Moving Targets By MUSIC Algorithm Considering Doppler Effect

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***Abstract* — For the multiple users communication or multiple target detection, it is important to separate multiple targets or users that are included in the same communication or radar antenna beam. In this paper, a combined OFDM (orthogonal frequency division multiplexing) system and bandpass sampling method using MUSIC (multiple signal classification) algorithm for automatic AOA (angle of arrival) tracking is discussed. Also, we propose a new method of the time division multiplexing with bandpass sampling at the same time to avoid the interference due to the imperfect RF (radio frequency) filter characteristics. The main purpose of our proposed structure is to avoid the interference generated from the multi-band signals processing. Next, we consider the Doppler effect caused by the target movement in mobile environment. After compensating the Doppler effect with valid range, the system performance is improved. Computer simulation results show the performances of MUSIC spectrum for AOA in various conditions and demonstrate the accuracy of AOA estimations.**

Keywords-MUSIC; AOA; Bandpass sampling; OFDM; Doppler effect

I. INTRODUCTION

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. The most recent ones are MUSIC [2] and ESPRIT (estimation of signal parameters via rotational invariance techniques) [3] algorithms; both utilize subspace-based exploitation of the Eigen structure of the input covariance matrix and thus require higher computation efforts. Although the ESPRIT needs less computation, the MUSIC algorithm is found to be more stable and accurate [4]. In this paper, we use the MUSIC algorithm combined with the OFDM bandpass sampling signal model to perform the antennas sensing for allowing the accurate azimuth. The accuracy of the estimation in azimuth increases proportionally to the number of antenna elements.

Bandpass sampling can be used for direct down conversion without analog mixers. In practice, the required sampling rate for ADC (analog to digital converter) can be too high to be achieved if the Nyquist sampling theorem is considered [5]. So, we want to use bandpass sampling which samples the signals with smaller sampling rate than Nyquist sampling rate to relax the demand for ADC. After down-sampling about over two band signals using the bandpass sampling, the signals are digitized. Then, two band signals can be received [6].

In the case of the conventional bandpass sampling receiver architecture, which we also refer to it as RF bandpass front-end receiver architecture, the SDR (software defined radio) receiver design has been widely treated in the literature. In particular, in [7] and [8], RF bandpass sampling frontend for SDR is designed to place analog-to-digital converter as near the antenna as possible. From the point of view of receiver design, due to previous system [9], 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, when the existing receiver architecture try to receiver the multi-band signals, it inevitably generates interference among the multi band signals, which can seriously damage the performance of the communication system.

In this paper, we propose a novel TDM (time division multiplexing) [10] based on the bandpass sampling RF(radio frequency) front-end receiver structure [5], and we compare it with the state-of-the-art systems. Our proposed structure can avoid the interference generated when doing the multi-band signals processing. The main motivation of our proposed structure is that we can reduce the complexity of the operation by using the TDM structure comparing with the existing structure.

This paper is organized as follows. In Section II, the system model is introduced. In Section III, we consider the Doppler effect of the system and compensation. Section IV describes the existing structure as well as our proposed

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structure. Section V gives the simulation results, which show the performances of MUSIC spectrum for AOA in various conditions and demonstrate the accuracy of AOA estimations. In the end, Section VI gives the conclusions.

II. SYSTEM MODEL

A. OFDM Signal and Multi-Antenna Receiver

In this paper, we consider two signals that have different center frequencies. Transmitted signals are based on OFDM. Eq. (1) is the signal in time domain. Let us assume that there are two received bands and there are both transmitted signals. Each band has different signals from each transmitter and $X^A_{k,m}$ and $X^B_{k,m}$ are the transmitted signals, respectively. After an IFFT (inverse fast Fourier transform) processing, each band is represented in time domain as $x_A(t)$ and $x_B(t)$ and has center frequencies of f_A and f_B.

$$x(t) = \begin{cases} \dfrac{1}{\sqrt{N}} \displaystyle\sum_{k=0}^{N-1} X^A_{k,m} e^{j(\frac{2\pi k}{N} + f_A)t}, x_A(t) \\ \dfrac{1}{\sqrt{N}} \displaystyle\sum_{k=0}^{N-1} X^B_{k,m} e^{j(\frac{2\pi k}{N} + f_B)t}, x_B(t) \end{cases} \quad (1)$$

The receiver is equipped with ULA (uniform linear array) of *M* elements located along a straight line. The system can be viewed as the multiplication of each received ray by a steering vector considering the direction of arrival of each multipath. Assume that there are P (P <M) uncorrelated narrowband signals $x_p(t)$ received by ULA from different direction θ_p, corrupted by AWGN (additive white Gaussian noise), where p=1,2… P. The observation is given as

$$X(t) = \sum_{p=1}^{P} a(\theta_p) * x_p(t) + n(t) \quad (2)$$

where $a(\theta)$ is the array steering vector given by

$$a(\theta) = [1 \;\; \mathrm{e}^{-j2\pi d \sin\theta/\lambda} \cdots \mathrm{e}^{-j2\pi d \sin\theta(M-1)/\lambda}]^T \quad (3)$$

where d is the inter-element spacing, λ is the signal wavelength. When we take a snapshot at time k=1,2…K, we can get

$$X(k) = \sum_{p=1}^{P} a(\theta_p) * x_p(k) + n(k) \quad (4)$$

where noise $n(k)$ is assumed to be both temporally and spatially white, and uncorrelated with signal $s_p(k)$.

In Figure 1, two signals are received through multi-antenna receiver. 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. The sampling frequency is smaller than Nyquist rate but larger than signal bandwidth of twice shown in Figure 1. TDM and bandpass sampling are processed at the same time. The digital oscillator is processed in the digital part; so, there are no problems that are generated by analog oscillator. After ADC and removing the cyclic prefix (CP), signals are processed through the MUSIC algorithm and the frequency domain equalization.

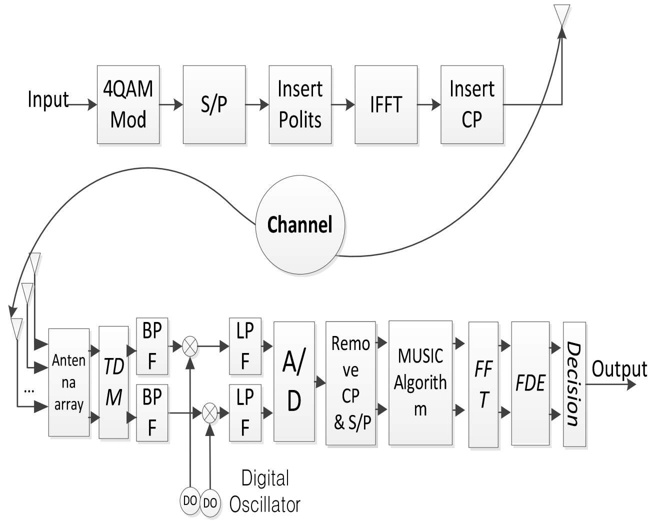

Figure 1. System Model.

B. Angle of Arrival Estimation using MUSIC Algorithm

MUSIC stands for MUltiple SIgnal Classification. The covariance matrix, R, is the collected data for each of the array receivers in the time domain. The correlation matrix is given as [6]

$$R = E\left[XX^H\right] = AR_sA^H + \sigma^2 I \quad (5)$$

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

$$P(\theta) = \frac{1}{a(\theta)^H E_N E_N^H a(\theta)} \quad (6)$$

III. DOPPLER EFFECT AND COMPENSATION

The orthogonality among the subcarriers is often destroyed by the carrier frequency offset due to the mismatching of oscillators between transmitters and receivers. Therefore, the Doppler effect is generated and degrades the communication performance. Doppler effect results in the frequency shift in frequency domain and is represented phase rotation in time domain. The signal x(t) is similar to Eq.(7) due to Doppler effect.

$$y_n = \sum_{k=0}^{N-1} H_k \cdot X_k \cdot e^{i2\pi\frac{k+\varepsilon}{N}} + z_n \quad (7)$$

The 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 as phase rotation in frequency domain. In Eq. (7), k, n, ε are sub-carrier, symbol, and the normalized Doppler frequency, respectively.

$$\begin{aligned} Y_p &= \sum_{m=0}^{N-1}\sum_{k=0}^{N-1} H_{k,m}\cdot X_{k,m}\cdot e^{i2\pi\frac{(k+\varepsilon)}{N}}\cdot e^{-i2\pi\frac{m}{N}} + Z_p \\ &= H_p\cdot X_p\, e^{i2\pi\varepsilon p} + \sum_{\substack{m=0\\ m\neq k}}^{N-1}\sum_{k=0}^{N-1} H_{k,m}\cdot X_{k,m}\cdot e^{i2\pi\frac{(k-m)}{N}}\cdot e^{i2\pi\frac{\varepsilon}{N}} + Z_p \end{aligned} \quad (8)$$

In (8), the first stage is phase rotation and the second stage is inter carrier interference (ICI), where p is symbol in frequency domain and k, m are sub-carrier before IFFT in transmitter and sample before FFT in receiver. Phase rotation of Doppler is different per symbol and ICI is generated when one sub-carrier affects other sub-carriers.

The normalized offset value is found by division with the carrier spacing. We consider the direction of the receiver to be heading towards the transmitter.

$$\varepsilon = \frac{f_d}{carrier\ spacing}\ ,\ f_d = \frac{v\cdot f_c}{c} \quad (9)$$

In this system, we compensate those problems with synchronization signal and block type pilot and it is assumed that the receiver speed is constant. fd, c, v are Doppler frequency, the velocity of light, and the speed of receiver, respectively.

The problem of Doppler effect is compensated with block type pilot. The phase rotation is estimated by doing interpolation between pilots, because the receiver speed is not dramatically changed.

$$Y_p = H_p \cdot X_p\ e^{i2\pi\varepsilon p} + Z_p \quad (10)$$

The phase rotation is estimated by using the received pilot signals.

$$P(i) = \sum_{i=1}^{N} mean\left\{\sum_{n=1}^{64}\ Block\_Pilot(i+n-1)\right\} \quad (11)$$

$$\frac{angle\{P(i)\} - angle\{P(i+1)\}}{pilot\_interval}\cdot\left([1:pilot\_interval-1]\right)$$
$$i = 1, 2, \ldots \quad (12)$$

P is average of block type pilot. Eq. (12) represents linear interpolation using P. So, the symbols that have no pilots are estimated. But, if there is too long pilot interval or Doppler frequency, it is difficult to compensate the effect.

IV. PROPOSED BANDPASS SAMPLING METHOD

A. Existing Structure

The existing multi-band system with bandpass sampling finds sampling frequency that does not overlap signals between multi-band signals according to (7). But, to select multi-band signals, RF filter is used. Although RF filter has good Q value, the RF filter cannot remove all adjacent signals. So, the remaining adjacent signal is able to be overlap when multi-band signals are converted at low frequency band. Bandpass sampling about multi-band of over 2 bands meet condition like (13) [9]. To convert the two signals in low frequency band without interference between signals, $F_{IF,A}$ and $F_{IF,B}$ have to meet (13).

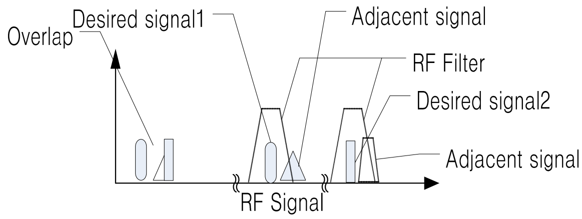

Figure 2. The problem when signals are sub-sampled from RF band.

$$\begin{aligned} &0 < F_{IF,A} - BW_A/2,\quad F_S > F_{IF,A} - BW_A/2 \\ &0 < F_{IF,B} - BW_B/2,\quad F_S > F_{IF,B} - BW_B/2 \\ &if\ F_{IF,B} > F_{IF,A} \\ &\qquad F_{IF,B} - BW_B/2 > F_{IF,A} + BW_A/2 \\ &if\ \ F_{IF,A} > F_{IF,B} \\ &\qquad F_{IF,A} - BW_A/2 > F_{IF,B} + BW_B/2 \end{aligned} \quad (13)$$

At first, the signals that are converted into low frequency band are larger than 0 and smaller than Fs/2, respectively. Secondly, the low frequency part of $F_{IF,A}$ is larger than the high frequency part of $F_{IF,B}$ ($F_{IF,B} < F_{IF,A}$) or, the low frequency part of $F_{IF,B}$ ($F_{IF,A} > F_{IF,A}$) is larger than the high frequency part of $F_{IF,A}$ ($F_{IF,B} > F_{IF,A}$).

B. Proposed Structure

We propose a method that adds TDM method into the bandpass sampling system.

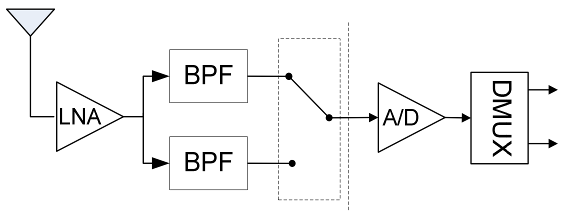

Figure 3. A multi-band receiver structure with bandpass sampling and TDM method.

The proposed structure is shown in Fig.3.

$$\begin{aligned} &0 < F_{IF,A} - BW_A/2,\quad F_S > F_{IF,A} - BW_A/2 \\ &0 < F_{IF,B} - BW_B/2,\quad F_S > F_{IF,B} - BW_B/2 \end{aligned} \quad (14)$$

Multi-band signals are received with antennas and the signals pass through LNA. Afterward, the multi-band signals are divided into two signals by filter. Each signal is sampled two times faster than the existing bandpass sampling frequency. After sampling processing in front of the ADC, TDM and bandpass sampling are performed at the same time. The signals that are received by TDM have no interference between the receiving signals because the signals are divided in time. Therefore, the converted signals just satisfy Eq. (14) instead of Eq. (13). So, it is possible to give an low sampling frequency.

Owing to our proposed algorithm, which adds TDM method using bandpass sampling in Eq.(14), we can reduce the complexity of operation by using the TDM structure comparing with the existing structure. Secondly, while by reducing the complexity of operation comparing Eq. (13) and Eq. (14), the system will relax its burden of hardware requirement and provide a faster processing speed than the existing structure.

V. SIMULATION AND DISCUSSION

Table 1 shows the simulation parameters. OFDM signals using 4-QAM modulation are used in the simulation. The number of OFDM subcarrier is 64 and the number of antenna is 8.

TABLE I. SIMULATION PARAMETERS

OFDM system	
The number of Subcarriers	64
Bandwidth	20MHz
Symbol Period	4us
Subcarrier Spacing	312.5KHz
CP Length	0.8us
Modulation	4-QAM
Channel	AWGN
Antenna Number	8
Antenna Type	ULA
Target A azimuth	10°
Target B azimuth	50°

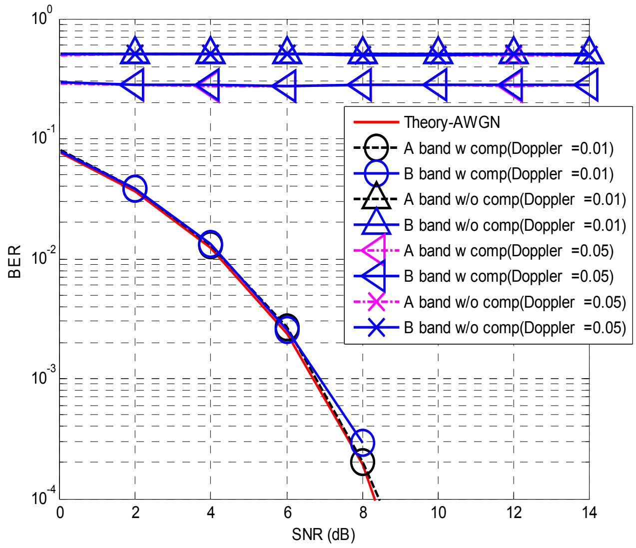

Figure 4. BER performance with Doppler effect.

Fig. 4 shows the BER performance under the Doppler effect with two different normalized offset values 0.01 and 0.05. Due to our proposed system, TDM can divide the multi-band signals into time one and time two then we can see performance according to only one band at a time. For the situation of normalized offset value given as 0.01 according to both band A and B without compensation, we cannot communicate because of the heavily damaged phase rotation. Then, after compensating the phase rotation, our proposed system shows that comparing with the theory curve, the degradation is very small, which is caused by remaining ICI. So, we can recover the communication performance. When the normalized offset value is 0.05, the pilot signal is of block type and the linear interpolation is performed. It is difficult to estimate fast phase rotation. We cannot communicate well when both band A and B are too much seriously damaged in the phase rotation.

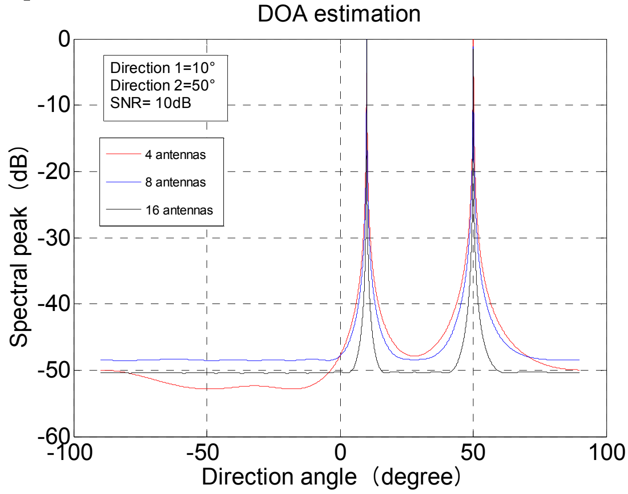

Figure 5. AOA estimation with array antenna numbers at SNR=10dB.

Fig. 5 and Fig. 6 show the MUSIC AOA performance with the different array antenna numbers under the same SNR environment. We can see that, the larger the array antenna number is, the more accurately the system can separate the signals. The better performance shows the sharper shape.

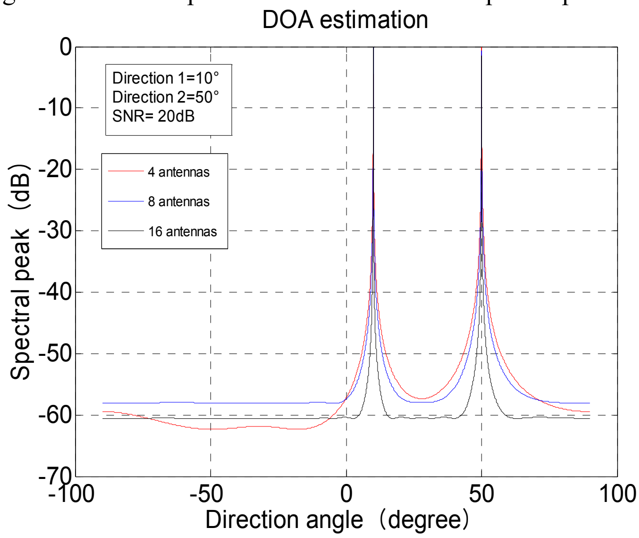

Figure 6. AOA estimation with array antenna numbers at SNR=20dB.

We evaluate the AOA estimations for two scenarios. One is to estimate the two mobile targets in a low speed environment, while the other considers the fast speed environment. The parameters of both cases are shown in Table 2 and Table 3.

TABLE II. PARAMETERS OF TARGETS IN LOW SPEED ENVIRONMENT

Target No	Power(dB)	AOA(degree)	Velocity(m/s)
Target 1	20	25°	20 （72km/H）
Target 2	15	24°	18 （64.8km/H）

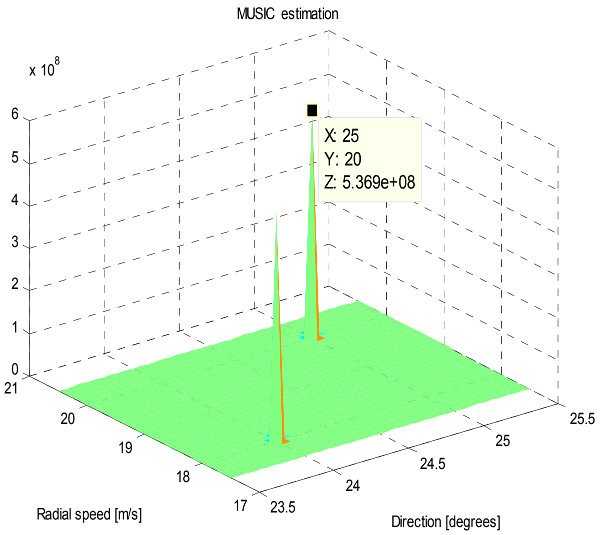

Figure 7. Estimation of Target 1 by MUSIC algorithm in low speed.

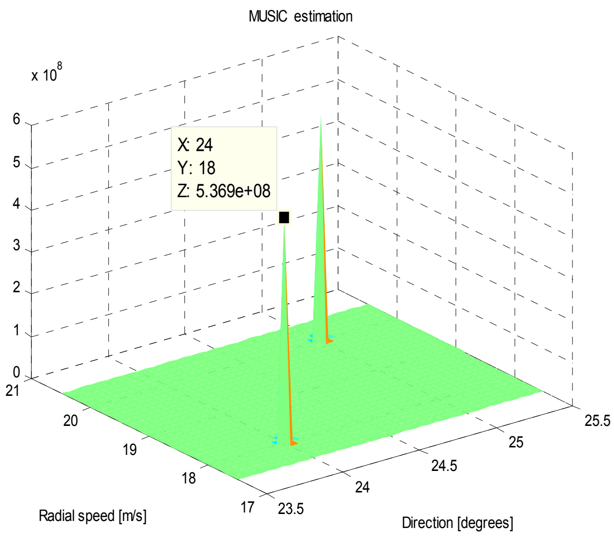

Figure 8. Estimation of Target 2 by MUSIC algorithm in low speed.

Fig. 7 and Fig. 8 show the estimation performances of two targets in low speed environment. And we can see that both the two targets are estimated by MUSIC algorithm which indicates their DOAs and velocities in the labels.

Fig. 9 and Fig. 10 show the estimation performances of two targets in the fast speed environment. We can see that both the two targets are estimated by MUSIC algorithm, which indicates their DOAs and velocities in the labels.

TABLE III. PARAMETERS OF TARGETS IN FAST SPEED ENVIRONMENT

Target No	Power(dB)	AOA(degree)	Velocity(m/s)
Target 1	20	35°	41.6 （150km/H）
Target 2	15	30°	44.4 （160km/H）

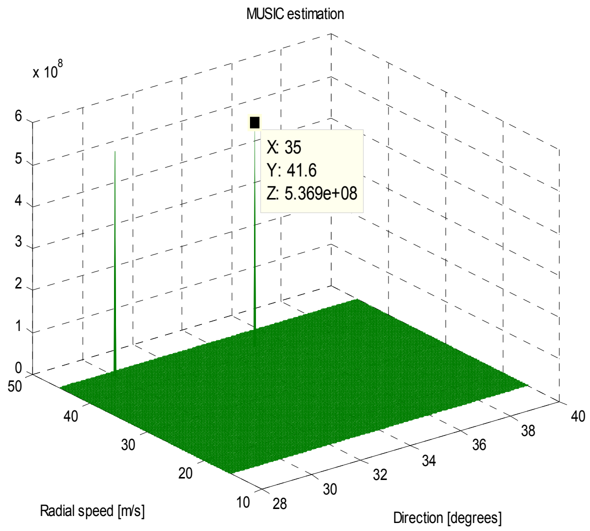

Figure 9. Estimation of Target 1 by MUSIC algorithm in high speed.

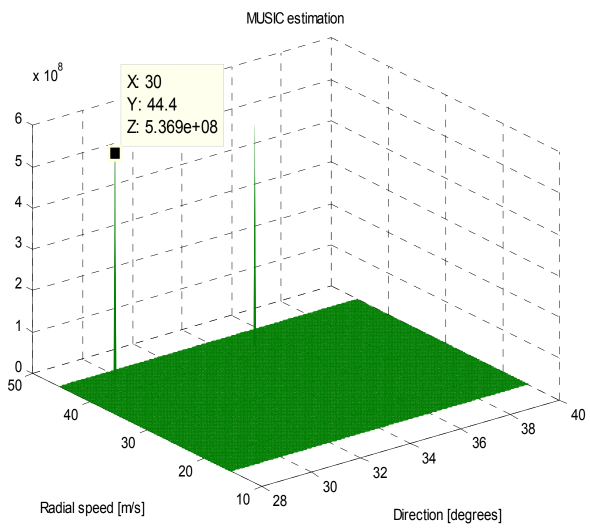

Figure 10. Estimation of Target 2 by MUSIC algorithm in high speed.

VI. CONCLUSIONS AND FUTURE WORK

In this paper, we proposed a new TDM based bandpass sampling method for the mobile targets tracking using MUSIC algorithm. The proposal consists of time division multiplexing with bandpass sampling at the same time to avoid the interference due to the imperfect RF filter characteristics in the receiver side. The main motivation of our proposed structure is that it can reduce the complexity of operation by using the TDM structure comparing with the existing structure. By using MUSIC algorithm, we can estimate the AOA of the targets in both low speed environment and fast speed environment. When we consider the Doppler effect, the system using the proposed method improves the communication performances by compensating the Doppler effect. Finally, computer simulations show that MUSIC algorithm can accurately estimate the AOA of the two targets under different conditions.

There are several directions that need to be investigated in the future. One of the directions is to improve the computational efficiency for covariance matrix of the conventional MUSIC algorithm. Other possible direction we want to explore is to improve the accuracy of the AOA estimation in the more complicated situation by considering other domain-related beamforming technology.

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