MUSIC Extracted Direction of Arrival Estimates Using spatial smoothing improvement for indoor localization using RFID

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***Abstract*—Due to its cheap price and no line of sight requirement, the RFID has emerge the field of indoor localization. Many algorithms have been developed, the classical ones have used the received signal strength informations as a metric, but due to the low degree of accuracy, we propose in this paper an other algorithm known as MUSIC algorithm which estimate the direction of arrival to determine the coordinates of the tag, then we will apply a spatial smoothing to eliminate the problem of interference.We will show through some simulations on Matlab software the most improvements like the accuracy that this technique has brought.**

Keywords—Indoor localization; RFID; MUSIC algorithm; Spatial smoothing.

I. INTRODUCTION

Nowadays, with the extension of buildings like museums and hospitals, people find a lot of difficulties to locate themselves and move in. In that case Intelligent localization systems are more than a luxury devices, As a GPS have many problems in such indoor environments like diffraction, refraction,absorption and attenuations of the electromagnetic wave coming so far from a satellite, the best solution is to use a wireless sensor network like the WIFI, ZigBee and RFID, this last one is the best due to its no line of sight requirement and its cheap price, comparing with the other ones.

Any Radio frequency identification system is composed of tags and readers. The Tag send its identification code to the reader, this last extract from the received signal some properties that are called metrics, the most used are RSSI (Received signal strength), TOA (time of arrival), D-TOA for Differential TOA and AOA for angle of arrival.

One of the methods is to use an array of antennas, then to measure the distance and the direction of arrival between the tag and the array. For the distance it's easy to calculate it,and it exist many works describing the manner to determine it from the RSSI.

The second step is to estimate the Direction of arrival, which is a parameter that reflect roughly the accuracy of the results. As all this interest we have dedicate all this article for the study of an algorithm of estimation of DOA called MUSIC, as well as an extension of this last one by using a spatial smoothing which improve the precision especially in the case of interferences.

The remainder of this paper is organized as follow: we give a mathematical representation of the signal incoming from the tag to the array in section II. Section III describes the Multi signals classification algorithm. Then we adapt the spatial smoothing to this one in section IV, the section V describe some simulations with comments about this method, also we compare the basic MUSIC with the improved one.Finally we give a conclusion in section VI.

II. INCOMING SIGNALS MODEL

First of all we model the mathematical expression for the incoming signals. In real life electromagnetic waves are diffused from the tag's antenna in all directions, moreover the antenna of the reader in reception is not an isotropic one. So to simplify the study we consider the distance between the antenna's array smaller than the distance between the tag and the array, we assume also that the antennas in reception are isotropic and the array sees only a finite number of plane waves.

A. case of one tag

The signal in reception $x_1(t)$ is the same signal emitted by the tag to the first antenna $s(t)$ with a certain delay τ_1 expressed as a phase shift (cause the signal is a sinusoidal wave):

$$x_1(t) = s(t)e^{-j2\pi f\tau_1} + n_1(t) \qquad (1)$$

Where f is the frequency of the carrier, generally for UHF passive tag $f = 867MHz$, and n is a noise caused by the propagation channel and the receiver front end components.

The delay τ_i is proportional to $d\sin\theta$.As the antennas array are spaced with the same distance d, we have:

$$c\tau_i = id\sin\theta \qquad (2)$$

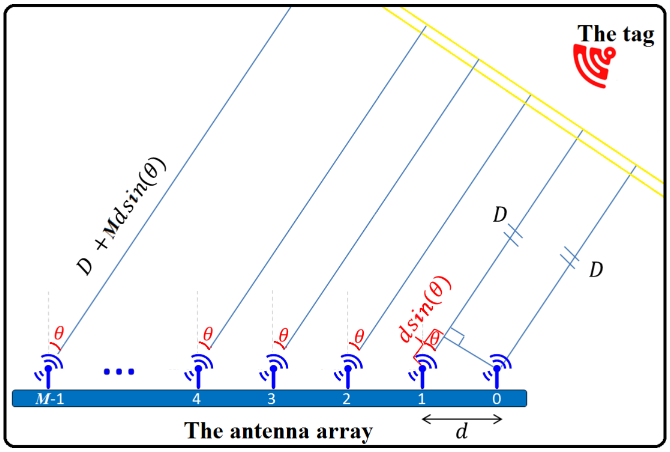

Fig. 1: Incoming signals from one tag

where c is the celerity of light and i is the number of antenna, assuming that $i = 0$ for the first antenna, as shown in Fig. 1. Extracting τ_i from (2) and submitting in (1) gives the expression of the received signal at the i^{th} node of the array:

$$x_i(t) = s(t)e^{\frac{-j2\pi f i d \sin\theta}{c}} + n_i(t) \tag{3}$$

Knowing that $\frac{f}{c} = \lambda$, (3) becomes:

$$x_i(t) = s(t)e^{\frac{-j2\pi i d \sin\theta}{\lambda}} + n_i(t) \tag{4}$$

B. case of multiple tags

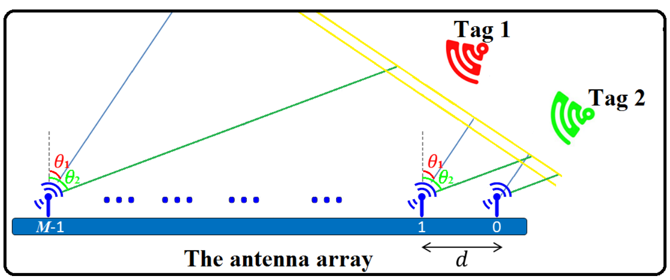

Fig. 2: Incoming signals from multiple tags

In the case of more than a tag in the area of the reader, as illustrated in Fig. 2, the signal in reception is the sum of all the signals sent by each mobile node :

$$x_i(t) = \sum_{k=1}^{P} s_k(t)e^{\frac{-j2\pi i d \sin\theta_k}{\lambda}} + n_i(t) \tag{5}$$

The latter equation can be written in its matrix form as follows:

$$[X] = [A]\,[S] + [N] \tag{6}$$

where $[X]$ is the received signal by each antenna of the array: $\begin{bmatrix} x_0(t) & x_1(t) & \dots & x_{M-1}(t) \end{bmatrix}^T$. $[S]$ is the transmitted signal from each tag : $\begin{bmatrix} s_1(t) & s_2(t) & \dots & s_P(t) \end{bmatrix}^T$.

And without forgetting the most important steering matrix $[A]$:

$$\begin{bmatrix} 1 & 1 & \dots & 1 \\ e^{-j\frac{2\pi d \sin\theta_1}{\lambda}} & e^{-j\frac{2\pi d \sin\theta_2}{\lambda}} & \dots & e^{-j\frac{2\pi d \sin\theta_P}{\lambda}} \\ \vdots & \vdots & \vdots & \vdots \\ e^{\frac{-j2\pi(M-1)d\sin\theta_1}{\lambda}} & e^{\frac{-j2\pi(M-1)d\sin\theta_2}{\lambda}} & \dots & e^{\frac{-j2\pi(M-1)d\sin\theta_P}{\lambda}} \end{bmatrix}$$

The additive noise is represented by the vector $[N]$: $\begin{bmatrix} n_0(t) & n_1(t) & \dots & n_{M-1}(t) \end{bmatrix}^T$.

III. MUSIC ALGORITHM

The principle of this algorithm which has been reinterpreted by Z. Guo in [1] is based on the orthogonality between the signal space and the noise space, so The signal in reception must be decomposed into two subspaces: The noise one and the signal one.To do that we calculate the correlation matrix, considering K is the number of samples:

$$R_{xx} = E[X.X^*] = \frac{1}{K}\sum_{t=1}^{K} x(t).x(t)^* \tag{7}$$

Devolopping the last equation using (6) gives:

$$R_{xx} = A.E[S.S^H].A^H + E[A.S.N^H] + E[N.S^H.A^H] + E[N.N^H] \tag{8}$$

As the signal and the noise are orthogonal, their correlation is null, so the middle terms of (8) are omitted so we obtain :

$$R_{xx} = A.E[S.S^H].A^H + E[N.N^H] = R_{ss} + R_{nn} \tag{9}$$

If we consider that the signals $s(t)$ are decorrelated, $E[S_i, S_j^H] = 0$ if $i \neq j$ and $E[S_i, S_j^H] = PW_i$ if $i = j$, where PW_i is the power of the signal $s_i(t)$, so:

$$R_{ss} = \begin{bmatrix} PW_1 & 0 & \dots & 0 \\ 0 & PW_2 & \ddots & \vdots \\ \vdots & \ddots & \ddots & 0 \\ 0 & \dots & 0 & PW_P \end{bmatrix} \tag{10}$$

As we can observe R_{ss} is a diagonal matrix where its diagonal elements represent the power of the signal which has been sent by the tag, so for the case of uncorrelated signals this matrix is non-singular.

If we consider the noise is a Gaussian white one of variance σ^2, the covariance noise matrix is given by:

$$R_{nn} = \sigma^2.I \tag{11}$$

Where I is the identity matrix of size M.

In the other side assuming that the signals $x(t)$ are totally decorrelated, as a consequence R_{xx} is hermitian, thus as the spectral theorem, explained in [2] it can be written as: $R_{xx} = [V].[DD].[V]^H$, where $[DD]$ is the eigenvalue diagonal matrix, and $[V]$ is the eigenvector matrix. As a result:

$$R_{xx} = [A].R_{ss}.[A]^H + \sigma^2.I = [V].[DD].[V]^H \tag{12}$$

Resolving this latter equation conducts us to define the eigenvalue as the power matrix (PW) seen before, and the eigenvector matrix as a matrix that contains two subspaces : The signal one that contains vectors related to the P big eigenvalues, when the other $M - P$ vectors related to the small eigenvalues form an orthogonal basis for noise space.

Finally, the direction of arrival $(\widehat{\theta})$ can be easily found using the resulting noise vectors (V_N), by testing all angles from -90 to 90, the ones that minimize the inner product : $S^H(\phi).V_N.V_N^H.S(\phi)$. With other manner, $\widehat{\theta} = \phi$ where Pmusic is maximized:

$$Pmusic = \frac{1}{abs(S^H(\phi).V_N.V_N^H.S(\phi))} \tag{13}$$

IV. Spatial smoothing

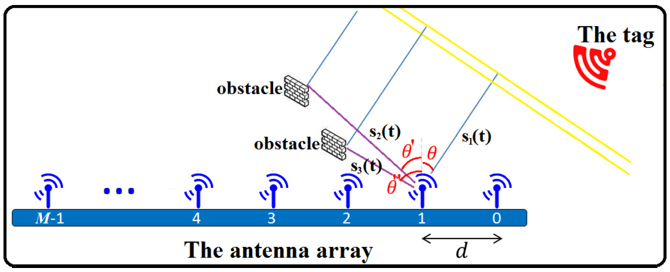

Fig. 3: Multipath problem

In real life it is impossible to have decorrelated signals due to multipath problem explained in [3], this issue is caused by reflections of the electromagnetic wave on the walls of a room as example as illustrated in Fig. 3, in such case every signal has some copies mitigated in power and delayed in time :

$$s_k(t) = \alpha_k s_1(t) \tag{14}$$

where α_k is the factor of mitigation and $s_1(t)$ is the original signal incoming from the tag. As a result, If we assume that $||s_1(t)||^2 = 1$ the source covariance matrix R_{ss} given in (15), is singular and not of full rank so some eigenvalues are null, thus it is impossible to extract the noise subspace as done in the previous section.

$$R_{ss} = \begin{bmatrix} 1 & \alpha_1 & \alpha_2 & \dots & \alpha_k \\ \alpha_1 & \alpha_1^2 & \alpha_1\alpha_2 & \dots & \alpha 1\alpha_k \\ \vdots & \vdots & \vdots & & \vdots \\ \alpha_k & \alpha_k\alpha_1 & \alpha_k\alpha_2 & \dots & \alpha_k^2 \end{bmatrix} \tag{15}$$

That kind of barrier is similar to the one encountered in the ATC navigation and surveillance systems exposed and solved by Evans in [4] and reinterpreted by Qing Chen in [5], where the first one separate the ghost signals and the original ones, by dividing the whole sensors array into $L = M - m + 1$ overlapping sub-arrays, each one includes m sensors as illustrated in Fig. 4.

The received signals at the l^{th} sub-array can be expressed as:

$$x_l(t) = A_m \psi^{l-1} s(t) + n_l(t) \tag{16}$$

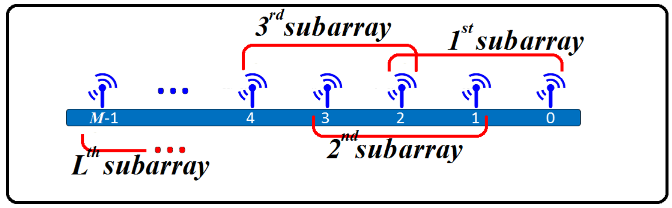

Fig. 4: Sub-arrays of the Spatial smoothing scheme

Where A_m is the same one in (6) but including just the m-first antennas so it is a $m \times P$ Vandermonde matrix, and ψ is the shifting matrix given by:

$$\psi = \begin{bmatrix} e^{-j\frac{2\pi d \sin\theta_1}{\lambda}} & 0 & \dots & 0 \\ 0 & e^{-j\frac{2\pi d \sin\theta_2}{\lambda}} & \ddots & \vdots \\ \vdots & \ddots & \ddots & 0 \\ 0 & \dots & 0 & e^{-j\frac{2\pi d \sin\theta_P}{\lambda}} \end{bmatrix} \tag{17}$$

Also it is possible to obtain the received signals covariance matrix of the $l - th$ sub-array by replacing A in (12) by its equivalent from (16):

$$R_l = A_m \psi^{l-1} R_{ss} (\psi^{l-1})^H A_m^H + \sigma^2 I \tag{18}$$

Then the global received signals covariance matrix is given by means of the L covariance matrix of every sub-array:

$$R_{xx} = \frac{1}{L}\sum_{l=1}^{L} R_l = A_m R_{ss}^L A_m^H + \sigma^2 I \tag{19}$$

Where R_{ss}^L is the smoothed source covariance matrix defined as:

$$R_{ss}^L = \frac{1}{L}\sum_{l=1}^{L} \psi^{l-1} R_{ss} (\psi^{l-1})^H \tag{20}$$

And by decomposing R_{ss} into its square root and the hermitian square root matrix, it is possible to write:

$$R_{ss}^L = GG^H \tag{21}$$

Where:

$$G = [\alpha, \psi\alpha, \psi^2\alpha, ..., \psi^{L-1}\alpha]$$

$$= \begin{bmatrix} \alpha_1 & 0 & \dots & 0 \\ 0 & \alpha_2 & \dots & 0 \\ \vdots & \vdots & & \vdots \\ 0 & 0 & \dots & \alpha_k \end{bmatrix} \begin{bmatrix} 1 & D_1 & D_1^2 \dots & D_1^{(L-1)} \\ 1 & D_2 & D_2^2 \dots & D_2^{(L-1)} \\ \vdots & \vdots & & \vdots \\ 1 & D_K & D_K^2 \dots & D_K^{(L-1)} \end{bmatrix} \tag{22}$$

Where $D_k = e^{-j\frac{2\pi dk \sin\theta_1}{\lambda}}$.

From (22) two things was observed by Kailath in [6]: The first is that the left matrix containing mitigation coefficients reflect the energy of every signal so each row has at least one non zero element. The second is about the right matrix which represent the direction vector matrix, as this last one is a Vandermonde one it is non-singular for $L \leq K$ thus the vectors are linearly independents. The two observations above conduct us to conclude that the matrix G is of rank K, consequently from (21) R^L_{ss} is too of full rank, the thing that allows us to extract the noise subspace then to estimate the directions of arrival.

V. Simulation and Results

In this section, we show all the simulations and give their results to hight-light the improvements brought by the technique of spatial smoothing in conjunction with the MUSIC algorithm and also to chose the best configuration as number of sub-arrays, number of tags.

To do that we take two situations, the first one is about one planar wavefront at 30° as direction of arrival, accompanied by two interfering signals, one of them at 35° attenuated to $1/5$ of the original one, the other one at 50° mitigated to $1/2$ of the expected signal.

In the second we take 2 Tags at 1° and 10° from the array sensors, the first signal has two interfering signals ($\theta'_1 = 7°, Pow'_1 = 0.3$ and $\theta"_1 = 5°, Pow"_1 = 0.6$) and the second signal is accompanied by one interfering signal ($\theta'_2 = 17°, Pow'_2 = 0.7$).

In both of the above situations the signal that have been sent is an amplitude modulated one containing the Id of the tag, transmitted at a carrier frequency of $867MHz$. To be closer from the reality we take 5000 samples for each signal. To simulate the propagation problems due to the environment, -10 dB was taken as signal to noise ratio.

The reception side consist of 10 antennas separated by $0.17m$ which is equal to $\lambda/2$. Differently to many simulations done by other researchers, we have not neglected the undesirable effects of the different blocks of a transmission chain, by simulating the filtering and the demodulation of the received signal following the RFID reader diagram in [7] .

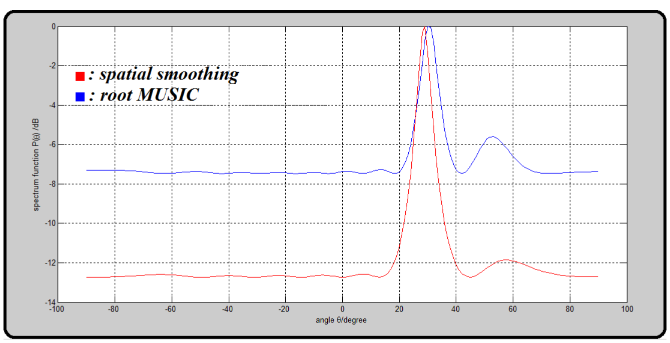

Fig. 5: Results of the first example

As a first step we present the graph of the pseudo-spectrum of both the root music and the one where we applied the spatial smoothing technique as illustrated in Fig. 5 and Fig. 6, and as a clear observation for one pretended Direction of arrival the original MUSIC algorithm detect two DOA (at 34° and at 52°) which means that the reflected signal has been also taken in consideration as a useful signal.

In the second example shown in Fig. 6, The old version of the algorithm neglects a whole direction of arrival, since the

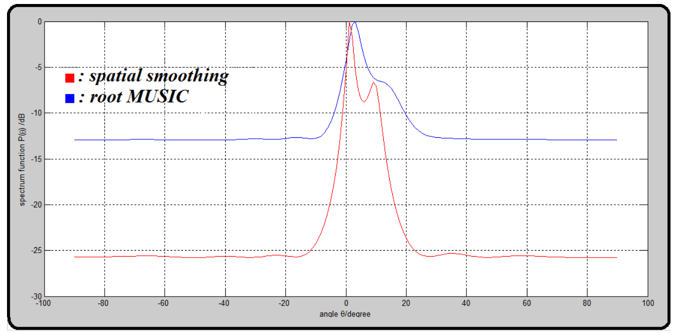

Fig. 6: Results of the second example

spectrum shows there is only one peak. So in this situation and the previous one the spatial smoothing preprocessing method gives MUSIC more accuracy in results, by smoothing the spectrum for the minimum power angles representing ghost signals and focusing on the maximum power angles that represent a direct reception.

To have a good interval confidence, each experiment has been repeated 200 times, then all the mean squared errors defined by (23) Have been averaged.

$$RMSE = \sqrt{\frac{1}{P}\sum_{i=1}^{P}\left(\theta_i - \hat{\theta}_i\right)^2} \tag{23}$$

Where $\hat{\theta}_i$ is the estimated DOA, θ_i is the real DOA and P is the number of TAG in the environment of the array-sensors.

The spatial smoothing can be more precise by choosing the right number of sub-arrays, To see the impact of this criteria, we change every time the number of antenna of each sub-array, in other words we changed the number of sub-arrays than we calculate the mean error, at the end we plot the graph shown in Fig. 7.

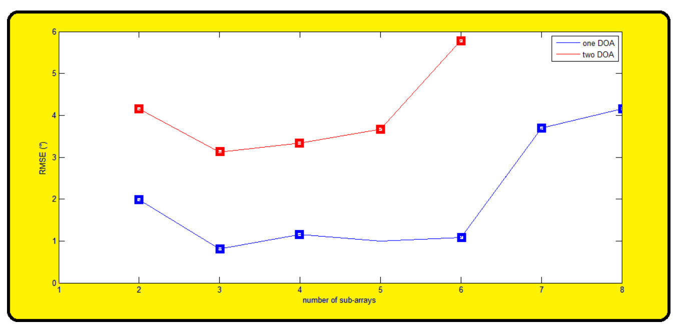

Fig. 7: Impact of the number of sub-arrays on the accuracy

As the number of the largest coherent sources group is 2 in both cases, so the spatial smoothing technique gives more accurate results when the number of sub-arrays is greater than 2, in our case the minimum RMSE is resulted when the sensor array is divided into 3 groups as shown in Fig. 6. This observation can be demonstrated theoretically that for $L > P'$, the averaged covariance matrix is of full rank, considering that P' is the number of the largest coherent sources group.

However it is good to increase the number of sub-arrays, it is important to keep in mind that going up over some number will deteriorate the results due to the reduction in the aperture, something we notice significantly in Fig. 6 ($L > 5$ and $L > 6$ respectively for the 1^{st} and the 2^{nd} example of this simulation).

VI. CONCLUSION AND FUTURE WORKS

In this paper, we present the spatial smoothing scheme and we observe its contribution in the field of indoor localization based on RFID technology. We proposed some simulation that have allow us to see the improvement brought to MUSIC algorithm using this preprocessing method proposed by Evans.

Also we adjust some parameters like the number of sub-arrays, However this parameter is important, there is some other parameters that must be studied in the future to get more accuracy and less latency.

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