A Novel Method to Implement the Matrix Pencil Super Resolution Algorithm for Indoor Positioning

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RECEIVED ON 11.03.2011 ACCEPTED ON 07.06.2011

ABSTRACT

This article highlights the estimation of the results for the algorithms implemented in order to estimate the delays and distances for the indoor positioning system. The data sets for the transmitted and received signals are captured at a typical outdoor and indoor area. The estimation super resolution algorithms are applied. Different state of art and super resolution techniques based algorithms are applied to avail the optimal estimates of the delays and distances between the transmitted and received signals and a novel method for matrix pencil algorithm is devised. The algorithms perform variably at different scenarios of transmitted and received positions. Two scenarios are experienced, for the single antenna scenario the super resolution techniques like ESPRIT (Estimation of Signal Parameters via Rotational Invariance Technique) and theMatrix Pencil algorithms give optimal performance compared to the conventional techniques. In two antenna scenario RootMUSIC and Matrix Pencil algorithm performed better than other algorithms for the distance estimation, however, the accuracy of all the algorithms is worst than the single antenna scenario. In all cases our devised Matrix Pencil algorithm achieved the best estimation results.

Key Words: Matrix Pencil, Indoor Positioning, OFDM, Time of Arrival, Super Resolution Algorithims.

1. INTRODUCTION

1.1 Time of Arrival for Mutlipath Indoor Position Systems

Estimation of the TOA (Time Of Arrival) and DOA (Direction Of Arrival) are typical issues in multi-path wireless communication systems, specially in indoor positioning systems [1-2]. In order to locate a particular object in indoor positioning systems, accurate estimation of TOA and DOA are required. Multi-path highly faded channels disperse the actual CIR (Channel Impulse Response) by noise effect causing an inaccurate estimate of TOA. IFFT (Inverse Fast Fourier Transform) and correlation based conventional techniques normally result in an inaccurate estimation [1,3]. Super Resolution Techniques based algorithms like ESPRIT , ROOT MUSIC and Matrix Pencil use the eigen values of the decomposed noise and signal subspaces [4]. This opts to eliminate the effects of noise in the received corrupted signal and to generate an accurate estimate of the TOA and DOA for indoor portioning systems [5]. This paper presents the short introduction to the algorithms used, along with

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the analysis of the actual estimates obtained using these algorithms and derives a novel method for implementing Matrix Pencil algorithm after initial analysis. Twenty Five (25) different data sets are processed. For each data set the estimates of the delay for the presented algorithms are evaluated and compared amongst. Performance of all conventional and super resolution techniques based algorithms are evaluated. Our presented algorithm achieve performnce significance in distance estimation amongst all the compared algorithms.

The rest of this paper is organized as following, Section 2 introduces the system setup and signal parameters used to capture the real data for the indoor positing system. Section 3 describes the conventional and super resolution algorithms and presents the novel method for matrix pencil algorithm used for the estimation of time of arrival and the distances. Section 4 presents the data sets description prepared for the analysis and Section 5 highlights the measurement and estimation results and validates the method implemention. Finally Section 6 concludes the article.

2. SYSTEM DESCRIPTION

Fig. 1 shows the block diagram of the system setup used to obtain the data values for the transmitted and received signals. The reference branch is directly connected to the digitizer system (Channel 1). The other branch is connected over an antenna connection. The left transmit antenna stands outside the building on a fixed point. The receive antenna is inside the building and is moved from point to point. The transmit antenna is connected to a signal generator which produces a continuous multicarrier signal. The receiver antenna is connected to the second channel of the digitizer system. The reference channel is used to reproduce the time of transmit between channel 2 and channel 1 and to estimate the delay of channel 2 and that of the channel 1. The delay difference of these two channels is used to obtain the TOA delay of the transmit and receive antenna. The reference channel is directly connected to the channel 1 of the digitizer card for the initial data sets. Therefore, in this branch no additional delay is considered. For the later measurements reference channel is removed and delay is calculated by transmitting two different signals separately and then by comparing their difference of TOA. The other antenna branch have additional delay through the cable, amplifier, and so on, for all measurements. The extra delay is subtracted from the estimated delays to get the actual estimates.

2.1 Description of the Transmitted Signal

Table 1 describes the multi-carrier signal properties, generated and transmitted through both reference and data channels.

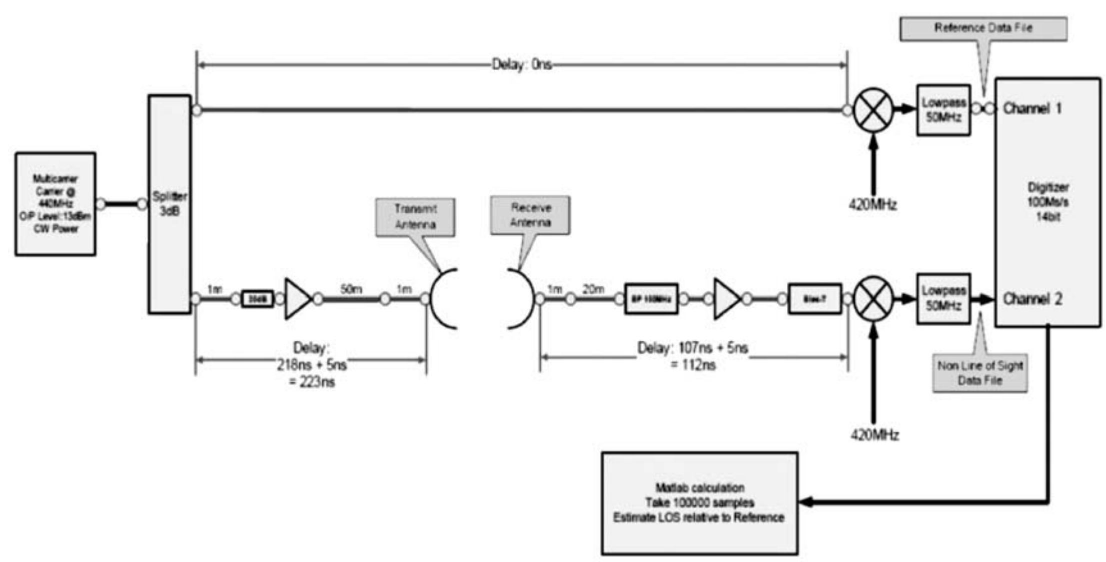

FIG. 1. BLOCK DIAGRAM OF THE SYSTEM

2.2 Signal Generation

Multicarrier signal is processed by the Matlab script with a fixed random phase distribution between the subcarriers. The length of the data files are 1e6 samples with a sampling rate of 100MS/s. The signal data vector has the same length of 1e6 samples. This signal data vector is transferred to a memory buffer of a IQ-Signal Generator from Rohde & Schwarz (AFQ100A, SMJ100A). The signal is constantly transmitted. The random phase vector is not changed during the transmission, rather it is fixed after the signal generation by the Matlab script.

2.3 Signal Reception

There are two channels in the digitizer system. The first channel is used for the reference data. The second channel is used for the NLOS (Non Line Of Sight) data. The initial target is to estimate the direct path delay in the reference data file (Channel 1), then the estimate of the direct path delay in the NLOS data files (Channel 2) is calculated. The system delays (Cable, Amplifier, and so on) are subtracted from the NLOS data files, and then the NLOS delay is subtracted from the reference delay. The result, is conventional TOA delay of the 1D-Range between transmit and the receive antenna. The data samples in the reference files are not corrupted, because the signal line is directly connected via cable from signal generator to the digitizer. The NLOS data files are corrupted with multi-path, NLOS conditions, attenuation and dispersion. The super resolution algorithms described in Section 4 are applied to combat these corruptions.

2.4 Data File Parameters

Table 2 describes the numbers and properties of the data measurement files used for capturing both the reference and NLOS (received) signals.

TABLE 1. DATA FILE MEASUREMENT PARAMETERS

Parameter	Value
Carrier Frequency of the First Transmitter	420MHz
Frequency Offset of the First Subcarrier	2MHz
The First Subcarrier Lays at	422Mhz
Number of Subcarriers	200
Frequency Spacing	200kHz
Normal Random Phase Distribution Between the Carriers	2*pi
Signal Range	422-462MHz

3. ALGORITHMS

In this section super resolution based algorithms are presented. Generally we can say that the IFFT and Xcorr use the peak detection method, because their accuracy is directly limited by the sampling rate. These methods give strict estimation for the distances by rounding the estimated distances and an error of 1.5 meter is normally expected. On the other hand super resolution algorithms are also affected by the sampling rate (signal bandwidth in general), but the estimation accuracy is better than the IFFT and Xcorr based algorithms [1]. This section briefly describes these algorithms, the details for these algorithms can be found in [3,6-10].

3.1 Channel Impulse Response

We can model the CIR based technique in multicarrier signal by expressing k as the carrier frequency index, the transmitted signal as X(k), corresponding additive noise as W(k) and the channel transfer function as H(k), all in frequency domain, then the received signal can be written as [2]:

$$R(k) = H(k)\,X(k) + W(k) \tag{1}$$

where H(k) is the CTF (Channel Transfer Function) of the time invariant multi-path dispersive channel.

In order to estimate the time delays we need first to estimate the CTF, since both transmitted and received signals are known, we can estimate H(k) by simply distinguishing the received signal from the transmitted one as:

$$H(k) = \frac{Y(k)}{N(k)} \tag{2}$$

TABLE 2. SIGNAL PROPERTIES

Property	Value
Number of Reference Files per one Indoor-Point	21
Names of the Reference Data Files	Reference0.txt, Reference1.txt,..
Number of Measured Points per NLOS File	200
Number of NLOS Files per one Indoor-Point	21
Names of the NLOS Data Files	LOS-MP0.txt, LOS-MP0.txt..
Sample Rate	100 Ms/s
Number of Data Samples	100000 Samples

The (CIR) can be calculated by taking IFFT of the CTF H(k) as:

$$CIR(\tau) = \sum_{K=0}^{N-1} H(k) e^{j1\pi\tau k} \tag{3}$$

where τ's represents the delays. By taking the first appearance of CIR(τ), we can get the first time of arrival τ_0. This method is very effective by noise since in many cases, the LOS signal is very weak and can not be distinguished well from the noise and from the other parts. The CTF and the corresponding CIR for 200 carries are shown in Figs. 2-3 for our application.

3.2 Cross-Correlation

The time delays can be calculated by taking the cross-correlation between the received and the transmitted time domain signals, which can be written as:

$$R_{\tau} = \sum_{n=-\infty}^{+\infty} y(n) \times (n+\tau) \tag{4}$$

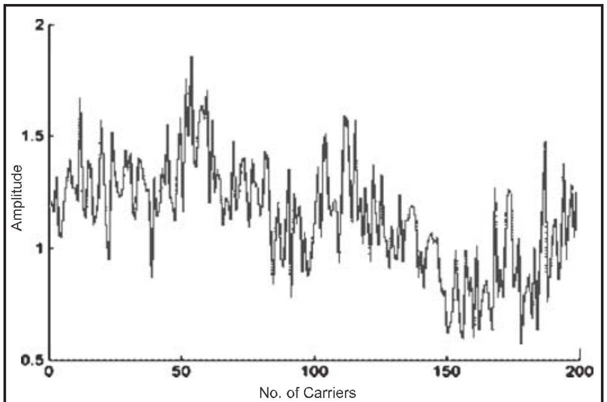

FIG. 2. THE CHANNEL TRANSFER FUNCTION

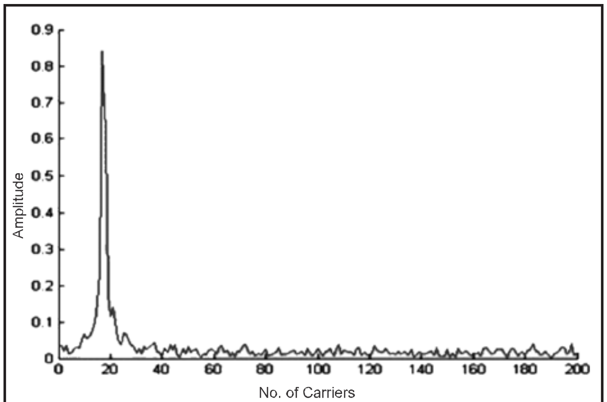

FIG. 3. THE CHANNEL IMPULSE RESPONSE

where n is time independent and y, x are the received and transmitted time domain signals respectively and can be achieved by simply applying the IFFT operations on X_n and Y_n, the frequency domain counterparts of the earlier, respectively. The cross correlation R_τ for the current application is shown in Fig. 4. The maximum value of R_τ appears at the time delay of the received signal. The problem with this method is that, the first estimated delay is in fact the total delay of the received signal and may not correspond to the first path, which causes an error if the reflected parts are strong. An alternate expression for R_τ is R_{xy} which can written as $R_{xy}=E[X_n Y_n^h]$ where E[.] is the expectation operator and h is the Hermitian transpose.

3.3 Estimation of Signal Parameters via Rotational Invariance Technique

ESPRIT is a subspace based algorithm for estimating parameters in a sinusoidal model [7] which is shown in Fig. 5.

Unitary ESPRIT algorithm is a subspace based algorithm for estimating the frequencies in a sinusoidal model. Unitary ESPRIT algorithm is extended form of the ESPRIT algorithm.

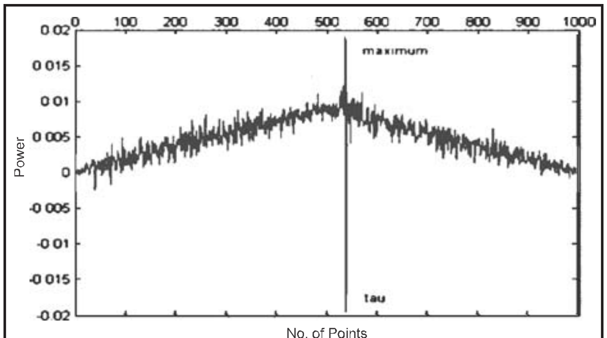

FIG. 4. CORRELATION FUNCTION(R_τ)

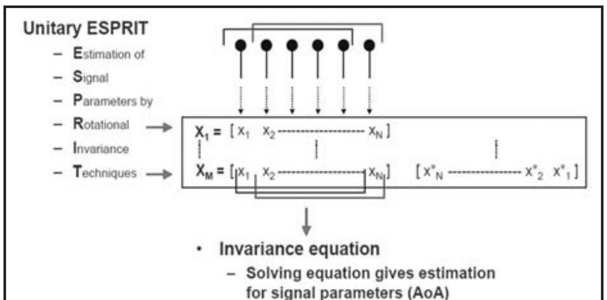

FIG. 5. THE ESPRIT ALGORITHM

The ESPRIT algorithm can be summarized in the following steps [7]:

(1) Obtain an estimate of the signal subspaces for the two sub-arrays

$$\hat{\hat{E}} = \begin{pmatrix} \hat{E}_1 \\ \hat{E}_2 \end{pmatrix}$$

(2) Solve the overdetermined system of equations

$$\hat{E}_1 \hat{\psi} \approx E_2$$

(3) Compute the signal poles by the Eigenvalue decomposition

$$\hat{z}_i = \lambda_i(\hat{\psi})$$

where E_1, E_2 are signal subspaces, z_i are signal poles, $\lambda_i(\psi)$ denote the i[th] Eigenvalues of ψ. ψ is the unique non-singular matrix relating E_1 and E_2, and [.] operator denote the estimation

Once Eigenvalues λ_i are computed using above procedure, the time delays $\tau_0 > \tau_1, \ldots, > \tau_{L-1}$ are estimated by finding the largest L peaks of $f(\tau)$ using Equation (7) given in Section 3.4.

3.4 Root-MUSIC

Root-MUSIC (Multiple Signal Classification) spectral estimation algorithm performs Eigenspace analysis of the channel correlation matrix in order to efficiently estimate the channel delay contents by making use of the estimated channel samples [8]. Taking the correlation of the estimated CTF we have:

$$R_N = E[HH^h] = A(\tau) R_d A^h(\tau) + \sigma_2 I \tag{5}$$

where R_d is the covariance matrix of the channel amplitudes d, $A(\tau)$ is the (NxL) time delay matrix with columns $A(\tau)$, where Equation (6) presents $A(\tau)$:

$$A(\tau_i) = \left(\begin{pmatrix} 1 & 0 & 0 & \cdots & 0 \\ 0 & e^{\frac{j2\pi\tau_i}{N}} & 0 & \cdots & 0 \\ 0 & 0 & e^{\frac{j2\pi\tau_i}{N}} & \cdots & 0 \\ \vdots & \vdots & \vdots & \cdots & \vdots \\ 0 & 0 & 0 & 0 & e^{\frac{(N-1)j2\pi\tau_i}{N}} \end{pmatrix} \begin{pmatrix} x_0 \\ x_1 \\ \cdot \\ \cdot \\ x_{N-1} \end{pmatrix} \right) \tag{6}$$

Solving the following Eigen system:

$$R_N E = E\Lambda$$

where $\Lambda = \text{diag}(\lambda_0, \lambda_1, \ldots, \lambda_N)$ contains the Eigenvalues of R_N, and E is a matrix contains the corresponding Eigenvectors E_k.

The number of paths is estimated from the fact that $\lambda_0 > \lambda_1, \ldots, > \lambda_L > \lambda_{\sigma W}^2$ where $\lambda_{\sigma W}^2$ is the noise variance. The time delays $\tau_0 > \tau_1, \ldots, > \tau_{L-1}$ are estimated by finding the largest L peaks of $f(\tau)$ where $f(\tau)$ can be calculated as:

$$f(\tau) = \frac{1}{A^h(\tau) \left[\sum_{K=L}^{N-1} E_k E_k^h \right] A(\tau)} \tag{7}$$

3.5 Matrix Pencil

MP (Matrix Pencil) is an efficient kind of super resolution technique. It uses a generalized pencil function used to obtain the exponents of a sum of complex exponentials. It is a variation of Eigen-structure approach and uses input MP instead of the correlation matrix [3,5]. The signal components can be distinguished exactly by mapping the noise components to the null space. It is an accurate algorithm for calculating time domain parameters of the network analyzer. It is used to obtain the poles of the system in one step process.

MP of a function can be defined as followings, if two functions g(t) and h(t) are combined on a common interval with scalar parameter Ψ such that $f(t,\Psi) = g(t) + \Psi h(t)$ then, $f(t,\Psi)$ is called the pencil of g(t) and h(t) parameterized by Ψ. When g(t) and f(t) and Ψ are appropriately selected, the pencil of functions contains very important features that are helpful in extracting information about the poles present in the particular function [10]. MP is used for multi-path parameters used for channel modeling. The MP algorithm can be summarized as:

The parametric model for the discrete complex frequency domain channel response can be written as

$$H_n = H(f_n) = \sum_{I=0}^{L-1} h_1 \exp - j2\pi f_n \tau_1, n = 0,1,\ldots N-1$$

where L is the number of multi-path components, N is the number of measurement points, f_n is the frequency. Time domain sampled values can be written as:

$$H_n = \sum_{I=0}^{L-1} \hat{h}_1 z_1^n$$

where

$$\hat{h}_l = h_l e^{-j2\pi f_o \tau_l}$$

$$Z_l = e^{-j2\pi \Delta_f \tau_l}$$

Δ_f is the frequency spacing between the adjacent channel samples and f_0 is the starting measurement frequency .

To apply MP for the measured frequency response the following vector is constructed [9]:

$$D_i = [H_i, H_{i+1}, \ldots, H_{i+N-P-1}]^T \tag{8}$$

where N is the number of measurement points and P is the pencil parameter chosen between N/3 to 2N/3 to get good performance. The value of P should be greater than that of L, which is the number of paths. We present the novel method for implementing this algorithm in which we construct $\breve{Y}_1, \breve{Y}_2$ and from following equations

(9)

then the following MP is considered

(10)

where Λ is same as defined under Equation (7) in section 3.4. To find the number of multi-path components L, a singular decomposition of matrix is carried out as:

(11)

where

$$\breve{Y} = \left[D_0, D_1, \ldots, D_P\right]$$

and V and U are unitary matrices and Δ is a diagonal matrix containing Eingenvalues of . The parameter L is chosen such that the singular value beyond L are set to zero.

The time delays are estimated from the Λ_l as:

(12)

The amplitudes of h_l of the multi-path components can be calculated by solving the system $HX = \hat{h}$ using linear least squares as:

$$\hat{h} = \left(\left[X\right]^h \left[X\right]\right)^{-1} \left[X\right]^h H \tag{13}$$

where

$$X = \left[X\left(\tau_0\right), X\left(\tau_1\right), \ldots, X\left(\tau_{L-1}\right)\right] \tag{14}$$

$$X\left(\tau_k\right) = \left[1, e^{-j2\pi f_s \tau_k}, \ldots, e^{-j2\pi\left(N-1\right) f_s \tau_k}\right]^T \tag{15}$$

and

$$\hat{h} = \left[\hat{h}_0, \hat{h}_1, \ldots, \hat{h}_{L-1}\right]^T \tag{16}$$

The method explained above improves the performance of the matrix pencil method. The validation of this statement is confirmed by the analysis of the estimated results presented in the following section.

$$\breve{Y}_1 = \left[D_0, D_1, \ldots, D_{P-1}\right]$$

$$\breve{Y}_2 = \left[D_0, D_2, \ldots, D_P\right]$$

4. ANALYSIS

This section describes the two different scenarios for the recorded data sets. For each scenario different data packs are processed. Each data pack consists of the transmitted (reference) and the received (data) signals. For each signal 21 different recording are captured, each in a separate data file. Every data pack is recorded from different locations and distances according to the setup shown in Fig. 1.

4.1 Test Data for Single Antenna Scenario

We formed different data packs for the captured data through the system setup described in section 2. As specified earlier that, the data transmitted through channel 2 of Fig. 1 from the antenna stands outside the building on a fixed point is received by the antenna inside the building and is moved from point to point. For different points different sets of data packages are captured. The same signal is transmitted through channel 1 and is used as a reference channel. The circuit delay for the first two data packages is 335ns and the original distances are 21.35m and 24.45m respectively. Using conventional CIR and correlation techniques large errors in the estimated distances are found, however the super resolution techniques algorithms described in the earlier section efficiently reduced the amount of error in the delay and distance estimations. It is noticed that 1000 samples of the measured signal are enough to estimate the distance in this case. Another scenario for the actual data packages is with the two antennas case. In this scenario the reference channel is removed and two different antennas are used to transmit the signal, while both transmitted signals are received by single receiver antenna. We further explain this scenario in the subsequent subsection.

4.2 Test Data for Two Antenna Scenario

The second scenario for recording the data sets is such that we have two transmitters and one receiver. Both transmitters send their multi-carrier signals (assuming at the same time). Each transmitter has a GPS timing receiver with a GPS isochronous 10MHz signal output. These 10MHz reference signal is used for the 10MHz reference input from the Agilent signal generator. The GPS timing receiver has a 1PPS signal, for triggering the baseband output signal one time, for the beginning and starting the first sample. The disadvantage is, that in this case of 2 transmitters, the inaccuracy of the PPS and 10MHz output between each GPS receiver can be +/-10ns. These +/-10ns are slowly shifted in time and are not predictable. These 10ns are also cause the additional range estimation failure. The spectrum bandwidth for both singal is 40MHz, one spectrum is shifted 20KHz from the other, so that each carrier is shifted 20KHz. If we receive the right carriers for both signals then the logic scenario regarding calculating the distances is described in sequel. We will have two channels (using the transmitted signals as references), that give two delays. The distances between each transmitter and the receiver can be calculated from these two delays . Since the channel is static (nothing is moving), we expect the two CIRs do not change between file to file, of a recorded data set. These CIRs would be used to estimate the distance between the transmitters.

Table 3 summarizes all the data sets processed, with their respective distances and delays for the single and multiple antennas.

5. DESCRIPTION OF ESTIMATION CONSIDERATIONS AND RESULT ANALYSIS

5.1 Description and Considerations

We have applied the algorithms described in section 3 for estimating the distance, directly from the CIR. The number of effective paths is estimated from the Eigenvalues of the correlation matrix according to a predefined threshold, which is related in fact to the noise variance. For distance

TABLE 3. LIST OF PROCESSED DATA SETS WITH ORIGINAL DISTANCES AND DELAY

Single Antenna				Multiple Antenna			
No.	DP Name	Distribution (m)	Delay (nSec)	No.	DP Name	Distribution (m)	Delay (nSec)
1.	DP1	21.35	335	1.	TDOA DP5 MSG11	6.08	0
2.	DP2	24.45	335	2.	TDOA DP5 MSG27	2.11	0
3.	DP3-18		480.7	3.	TDOA DP5 MSG47	5.79	0
4.	DP3-19		480.7	4.	TDOA DP5 MSG58	5.35	0
5.	DP3-20		480.7	5.	GPS PL4xPL1 P1	26.661	0
6.	DP4	27.2	377	6.	GPS PL4xPL1 P9	26.661	0
7.	Coax1x2	30.242	335	7.	GPS PL4xPL1 P11	26.661	0
8.	Coax1x4	26.661	335	8.	GPS PL4xPL1 P22	26.661	0
9.	Coax2x1	30.242	335	9.	GPS PL4xPL1 P27	26.661	0
10.	Coax3x1	36.226	335	10.	GPS PL4xPL1 P47	26.661	0
11.	Coax3x2	17.942	335	11.	GPS PL4xPL1 P58	26.661	0
12.	Coax4x1	26.661	335	12.	PL1xPL2 P47	5.80	0
				13.	PL2xPL3 P11	6.08	0

estimation, we are interested in the first tap only. For the IFFT and cross-correlation, the distance estimation accuracy was limited by the sampling rate, since one sample causes distance shift of 3m (1/100E6 *c). That entails even for very clean signal, an error of 3/2=1.5m is normal. The circuit delay for different distances, as shown in Fig. 1, was subtracted from the estimated distance in order to get the proper distance. Fig. 6 shows the first arrival of path with the CIR and Fig. 7 shows the corresponding CTF and the estimated number of paths for valid estimation from the CTF. It is evident that we do not require the whole CTF and the algorithms play the vital role. In order to get the accurate estimation we have taken median and the mean of the measured distances from 21 files of each data set.

Since the measurements taken for most of the files of a package, almost has a constant offset except one or two, that have very different values, the right averaging for an accurate estimate is the median instead of the mean. Regarding the accuracy of the super resolution, we have applied a known virtual channel with some paths to the transmitted signal by simple delay shifting and summation. The algorithms give very accurate distance estimation.

5.2 Result Analysis

Fig. 8 shows the comparison of performance of distance estimation for all the algorithms for every file in a data set

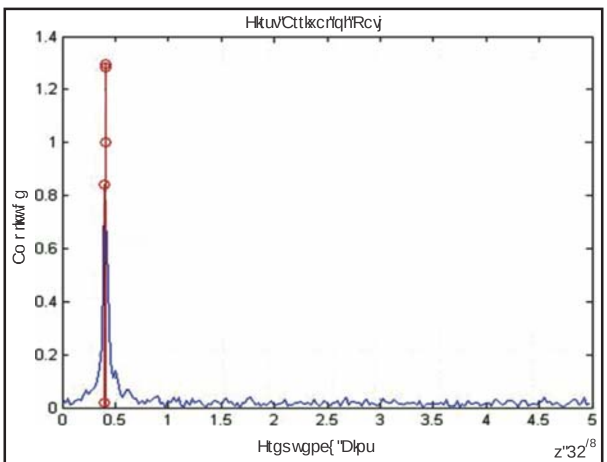

FIG. 6. FIRST ARRIVAL OF PATH

for single antenna case. It is clear from the figure that the Matrix Pencil algorithm gives optimal results. Correlation and IFFT have the deviations from to 4-8m from the original distance, ESPRIT has up to 3m while RootMUSIC and Matrix Pencil have deviation within 2m and 1m range respectively.

Figs. 9-10 show the estimates of distances file by file for two further data pack using normal and coaxil cabel is applied for synchronization respectively. The optimal estimation results are achieved by ESPRIT and Matrix Pencil algorithms in both cases again. The similar estimates are shown in Fig. 11 for the two antennas case.

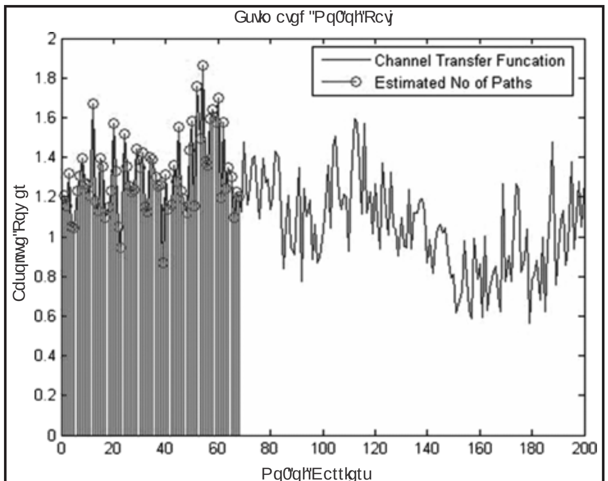

FIG. 7. ESTIMATED NUMBER OF PATHS FROM CHANNEL TRANSFER FUNCTION

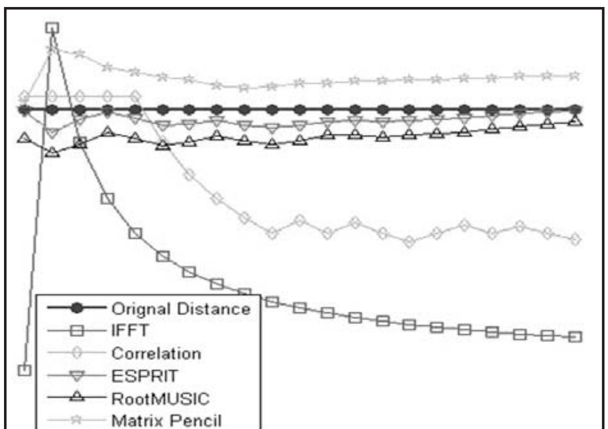

FIG. 8. ESTIMATES OF DISTANCES FILE BY FILE FOR A DATA PACK

The averages for the recorded data estimations are shown in Figs. 12-14 respectively. It consists of the mean, median and standard deviation values of the distances estimation for the conventional and super resolution techniques based algorithm. It can be noticed that in case of coaxil cable synchronization the ESPRIT and Matrix pencil algorithms perform better than the others. We can see that the performance of ESPRIT is bit worse than the other super resolution techniques. It achieves 3m to 4m deviations from the original distance, the other two techniques achieve 1-2m deviations. Thus Matrix Pencil again achieved the optimal results as expected. For the two antenna case we have the similar. results. It can be clearly evident from the figures that the performance of the ESPRIT algorithm is worst in the case of two antennas. It achieves a deviation of up to 5m while both RootMUSIC and Matrix Pencil have deviations up to 1.5m from the original range. Both RootMUSIC and Matrix Pencil again supersede the traditionnal techniques along with the ESPRIT algorithm.

6. CONCLUSION

We have simulated super resolution techniques along with conventional techniques in order to estimate the distance/ and time of arrival for the indoor positioning applications. Two different scenarios were tested. For each scenario

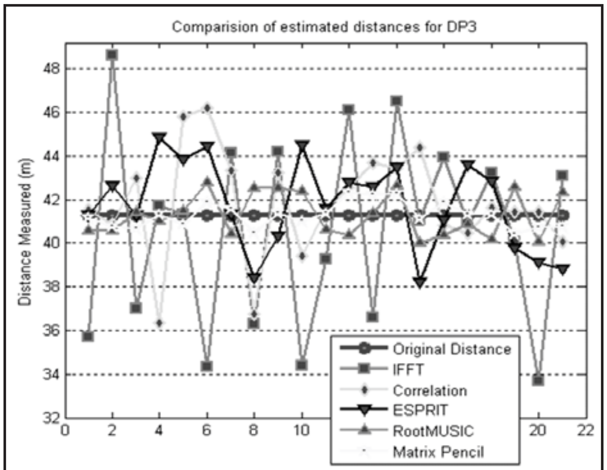

FIG. 9. ESTIMATES OF DISTANCES FILE BY FILE FOR ANOTHER DATA PACK

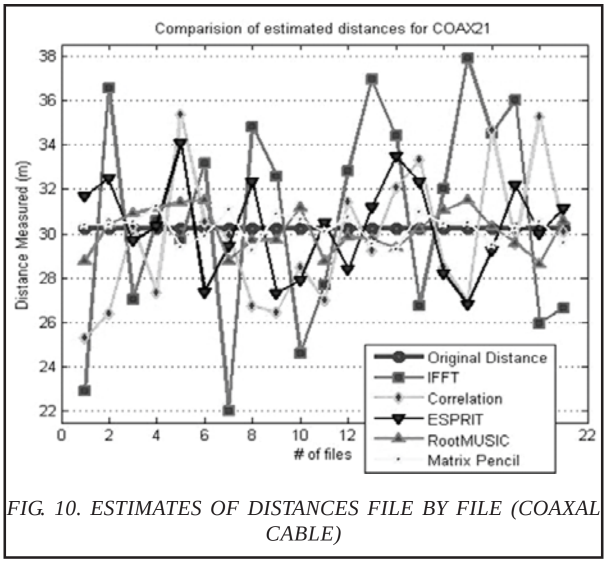

FIG. 10. ESTIMATES OF DISTANCES FILE BY FILE (COAXAL CABLE)

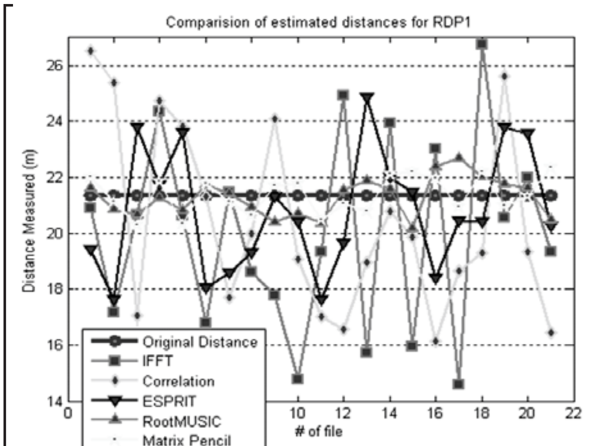

FIG. 11. ESTIMATES OF DISTANCES FILE BY FILE FOR TWO ANTENNAS

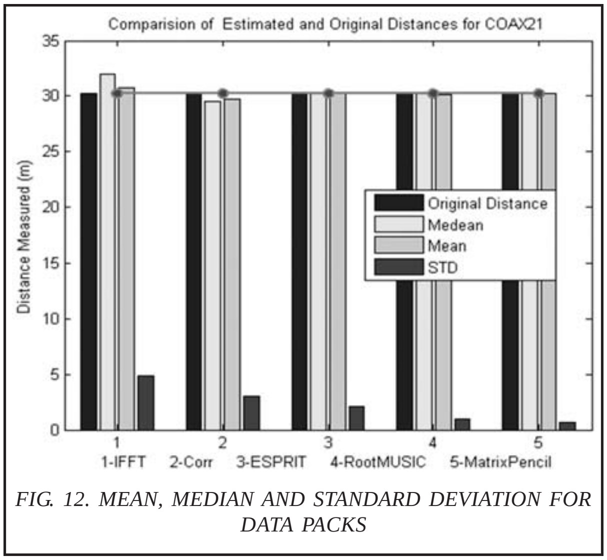

FIG. 12. MEAN, MEDIAN AND STANDARD DEVIATION FOR DATA PACKS

multiple sets of data are recorded, each with 21 different recordings. For single antenna scenario the super resolution techniques like ESPRIT and the Matrix Pencil algorithms implemented using our novel method give optimal performance compared to the conventional techniques. In most of the cases Matrix Pencil archived the estimation distance equal to the original distance, however, ESPRIT got 1-3m deviation which was increased further by 1m to 2m with the RootMUSIC algorithm. In two antenna scenario RootMUSIC and Matrix Pencil algorithm performed better than other algorithms for the distance estimation, however the accuracy of all the algorithms was worst than the single antenna scenario. In all cases Matrix Pencil using our the presented method is concluded as the best estimation technique.

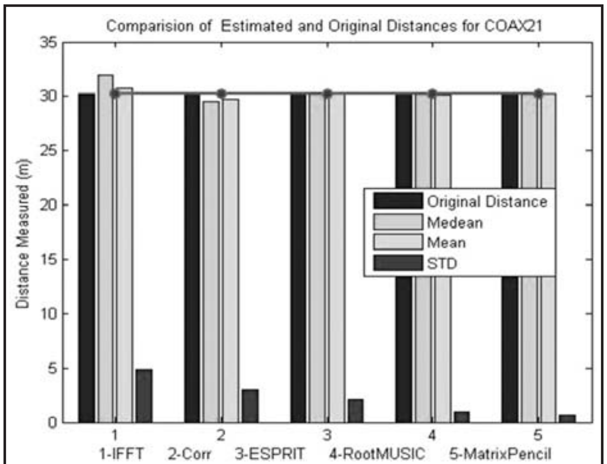

FIG. 13. AVERAGES OF THE ESTIMATED DISTANCES (COAXAL CABLE)

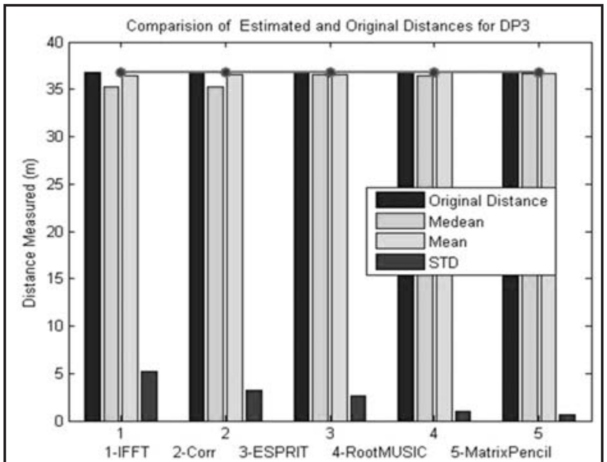

FIG. 14. MEAN, MEDIAN AND STANDARD DEVIATION FOR TWO ANTENNAS

AKNOWLEDGEMENT

The authors are thankful to University of Magdeburg Germany. for providing data caputirng facilities.

REFERENCES

[1] Li, X., and Pahlavan, K., "Super-Resolution TOA Estimation With Diversity for Indoor Geolocation", IEEE Transactions on Wireless Communication, Volume 3, No. 1, January, 2004.

[2] Prasad, R., "OFDM for Wireless Communication Systems", Artech House, 2004.

[3] Haddad, D.H., and Noga, A.J., "The Matrix Pencil and its Applications to Speech Processing" Technical Report, Final Technical Report, February 2004 to December 2006, Air Force Research Lab Rome Information Directorate, Accession Number: ADA466668.

[4] Khanzada, T.J.S., and Ali, A.R., " Super Resolution Techniques for Range Estimation Time Delay of Arrival-A Report for the TOA Estimation", Technical Report, IESK-HF-CMWCE/0508, Chair of Microwave and Communication Engineering University of Magdebug, May, 2008.

[5] Khanzada, T.J.S., Ali, A.R., Napoleon, S., and Omar, A., "Use of Super Resolution Algorithmsfor Indoor Positioning Keeping Novel Designed WLAN Signal Structure", 1st International Workshop on Digital Engineering, Magdeburg, Germany, 14-15 June 2010.

[6] Yuen N., and Friedlander B., "Asymptotic Performance Analysis of Three ESPRIT Algorithms", 29th Asilomar Conference on Signals, Systems and Computers, Volume 2, pp. 717, Asilomar, 1995.

[7] Lemma, A.N., Veen Alle-Jan Van Der, V.A.J., and Deprettere, Ed F., "Multiresolution ESPRIT Algorithm", IEEE Transactions on Signal Processing, Volume 47, No. 6, pp. 1722-1726, June, 1999.

[8] Chai, K.K., Jimaa, S.A., Alukaidey, T., and Sharif, B., "OFDM Frequency Offset Estimation using Root-MUSIC Algorithm", Communication Systems and Networks, Benalmadena, Spain, 2003.

[9] Pesavento, M., Gershman, A.B., and Haardt, M., "A Theoretical and Experimental Performance Study of a Root-MUSIC Algorithm Based on a Real-Valued Eigendecomposition", IEEE International Conference on Acoustics, Speech, and Signal Processing, Volume 5, pp. 3049-3052, Istanbul, Turkey, 2000.

[10] Ali, A.A., and Omar, A.S., "Super Resolution Matrix Pencil Algorithm for Future Fading Prediction of Mobile Radio Channels", Proceedings of the Eighth International Symposium on Signal Processing and Its Applications, Volume 1, pp. 295-298, 2005.