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Practical Implementation and Performance Estimation of MUSIC Method Implemented on Signal Processor TMS 320c30

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Results of practical implementation and performance estimation of MUSIC method implemented on signal processor TMS320c30 in 32-bit floating point arithmetic are presented in this paper. Implementation is performed as part of the research of methods for spatio-frequency signal analyses of radio-frequency spectrum and development of direction finder for VHF/UHF frequency range. Application software for implementation of MUSIC method on signal processor is a part of the simulation model for modeling superposition and parameter estimation of the superposed radio signals and, in the same time, a part of application software of direction finder prototype.

Key words: **MUSIC method, digital signal processing, signal processor, direction finding, radio reconnaissance, antenna arrays.**

Introduction

MUSIC (Multiple Signal Classification) [1] method is a kind of high resolution non - linear spectral method. This method is basically formulated in spatial domain as a method for Direction of Arrival (DOA) estimation and parameter estimation of superposed radio signals on the antenna array.

MUSIC method is of specific theoretic and practical interest to the researchers, due to its high resolution properties and good performances. MUSIC method is a subspace method. The high resolution properties of the MUSIC method are based on its special properties and decomposition of two orthogonal subspaces (signal subspace and noise subspace).

MUSIC method provides asymphtotic unbiased estimation of parameters of the spatial model of multiple signal superposition. It means that, in case of direction of arrival estimation problem, when the number of spatial-time samples of the signals in the asymptotic case tends to infinity, standard deviation of estimation error of the DOA estimation tends to lower Cramer-Rao's bound and mean value of the estimation error tends to zero.

MUSIC method can be applied to the DOA estimation when the antenna array is of non-uniform geometry, so the spatial sampling of the wavefront is non-uniform.

A big advantage of this method compared to the DFT method is that spatial (time) sampling of wavefront (signal) can be non-uniform**.**

By applying this method to the systems for radio-frequency spectrum monitoring, a problem of signal detection and parameter estimation in multiple incident signal scenario can be solved in a qualitative new way. These problems are difficult to solve using classic spectrum analysis, especially in the case when radio signals are, partially or fully, overlapping both in time and frequency. By applying the MUSIC method in the analysis of spatial samples of wavefront, the following unknown parameters of superposed radio signals can be estimated: the number of superposed (active) radio signals in the given frequency band and parameters of each superposed signal such as: spectral bandwidth, direction of arrival (azimuth and elevation) and polarization. The process of estimation of spectral bandwidths of multiple incident signals in open references is known as '*band or spectrum segmentation*' [4].

In the references about high-resolution spectral methods it is usually stated that MUSIC method does not have wide practical application in systems for direction of arrival estimation, because of its numerical complexity (this method requires estimation of eigenvalues and eigenvectors of complex square matrix). Paper [2], published on the 1992 IC ASSP Conference, represents one of the first papers analysing the possibility of the application of MUSIC method in sonar systems. The analysis is based on the estimation of arithmetic, memory and I/O resources needed for sonar immplementation, based on this method. It is shown that requirement of computer resources for the MUSIC method implementation are less than for other parts of the algorithm, which are done in the solution recommended for the sonar locator, and that complexity of MUSIC method is not a barrier for practical application.

Results of the practical implementation and performance estimation of the the MUSIC method implemented on the signal processor TMS 320c30 are presented in this paper. Implementation has been done as part of the research of methods for **spatio-frequency analysis** of radio-frequency spectrum in the development of the prototype of direction finder RGK-2/3, [3-7].

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Implementation of MUSIC Method on the Signal Processor TMS 320c30

Hardware Environment

Implementation of the algorithm for direction finding based on the MUSIC method is done on the signal processor TMS 320c30. It is a 32-bit floating point digital signal processor in the chip. Processor is realised in CMOS technology which executes 33 MFLOPs (Mega Floating Point Operations per second). The executing time of each instruction is 50 ns. The processor architecture is adjusted to structure of algorithms for digital signal processing, especially to digital filter structure and FFT.

The appearance of the digital signal processors in the chip, especially in the floating point arihthmetic (TMS320c30, Motorola 960000,....) represents a very important moment in the digital signal processing area. Implementation of the algorithm for digital signal processing in the floating point arithmetic is much easier than implementation in the fixed point aritmetic, since the problems of scaling and calculation accuracy, which are a specific aspect in the fixed point arithmetic, are overcome. There are C compilers for almost all commercially available processors in the floating point aritmetics, as part of the development software support, which significantly simplifies the development of application programs. Hardware platform for this implementation is the PC compatible signal processor board *ASPI Banshee* produced by *Atlanta Signal Processors*. Additional daughter boards such as board with AD/DA convertor, prototype board or board with external dynamic memory can be mounted on the main signal processor board. The board can be used as quite a useful development resource, but also as a target resource built in into a PC like devices and systems which require real-time signal processing.

Software Environment

There is a good software environment for the available signal processor board. The base element of software support is ASHELL, an upgrade of the DOS operating sistem. ASHELL enables a complete integration of the signal processor applications in a PC program environment, so that the PC behaves like a powerful station for digital signal processing. Application programs for signal processor boards can be developed in C programming language and/or assembler. Standard TI ANSI C compiler and/or assembler, are started by specific MAKROs from ASHELL environment which is used to generate an executing program for the signal processor. It is also possible to generate an executable version of the same C program for PC using Microsoft C compiler and assembler. Numerous tools (HEAR, VIEW, PATCH TI_LOAD) are available in software environment for the signal processor board. The package BAN_BUG is especially useful for the application development, since it represents a window oriented debugger.

Specific element of development support is SPOX. In references, SPOX is conisidered a real-time operating sistem for signal processor applications. Actually, it is not an operating sistem in the strict meaning of the word. SPOX represents a high-level software interface between the program and hardware signal processor platform. SPOX provides a layer between the underlying hardware of a target DSP system and the application program that runs on it. The programmer sees a "virtual DSP machine", a processor-independent model of DSP functionality. There are three categories of functions that the SPOX implements: math, stream I/O and memory management. Each of these areas encapsulates a part of the underlying hardware.

Programmer uses SPOX, by integrating SPOX programming modules, that are developed in assembler, in the application program that is developed in C programming language and/or assembler.

Structure of the Application Program

Implementation of the algorithm for direction finding based on the MUSIC method was a part of a wide research project devoted to the methods and technical solutions for spatio-frequency analyzing of radio-frequency spectrum. The algorithm that is implemented on the signal processor has been previously implemented in MATLAB in the simulation model for spatio-frequency signal analysis and it has been tested using simulated as well as real radio signals in the HF, VHF and UHF frequency band.

The simulation model, as well as the application program, was modularly designed and implemented.

When the algorithm was implemented in MATLAB, all numerical calculations were executed in floating-point arithmetics with double (64-bit) precision, while all numerical calculations on the signal processor were executed in the floating-point arithmetic with 32-bit precision. One of the purposes of implementation was to determine how 32-bit aritmetics will decrease the accuracy of the algorithm in relation to the implementation in 64-bit arithmetic in MATLAB. The part of the algorithm related to the estimation of eigenvalues and eigenvectors has been the most critical.

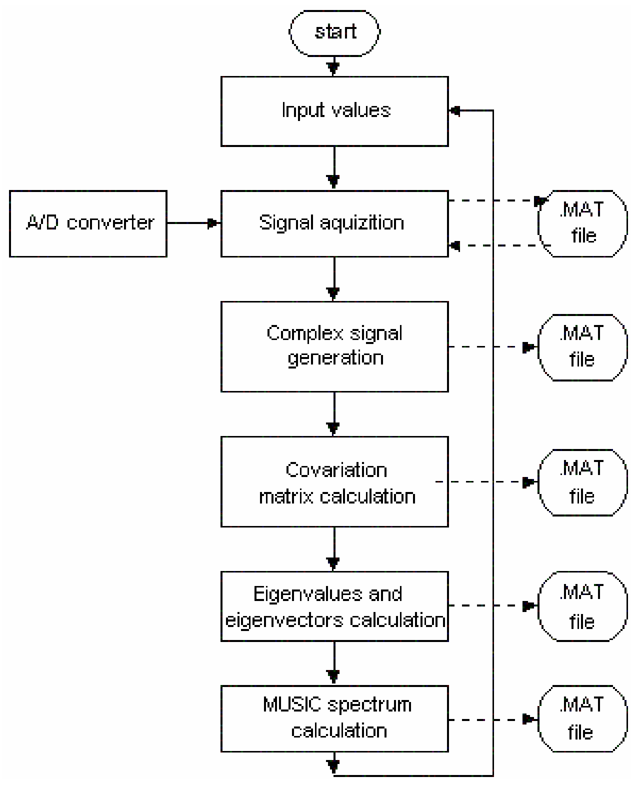

Figure 1. Structure of aplication program

The structure of the application program is presented in Fig.1. The application program was realised in the ANSI C programming language. The SPOX capabilities were mostly used for realisation of vector and matrical operations and for realisation of input/output stream operations. The application program is considered to execute concurrently with the PC application, which is impossible in ASHELL environment (later versions of SPOX make this possible!). It was necessary therefore to develop own programs for communication between the applications on the signal processor and the PC**.** The communication control between these two applications was realised using status and control registers of the signal processor board**.**

The application program is realised as a program loop. Interactive setting of all parameters which are needed for the execution of the program (spatial sector of interest, resolution which is used for the calculation of direction of arrival of the emitter, number and position of the antennas, etc.) is done at the beginning of the program loop. In the final program version these parameters are interactively set in the program which is executed on the PC and are transmitted in application onto the signal processor board in the way that the PC accesses the program memory of the signal processor through 16-kB window, while the signal processor is in the HOLD mode. When the estimation of the direction of arrival is required from the application on the PC, the PC unblocks the signal processor board which causes the execution of the program loop to start. When the signal processor returns to the starting point of the program loop, it sets itself to the HOLD mode and waits to be unblocked by a program from the PC again.

At the end of each program module, the interresults are written in .MAT file in ASCCII flat format, which can be read by MATLAB. In this way, during development of the application program, a comparison of the results obtained in MATLAB and those obtained on the signal processor can be made.

In the final versions, all the functions executed using the PC resources (printf, writing and reading .MAT files, data input from keyboard) are removed from the application program, which leaves the PC free to execute its own applications (interactive communicaton with operator, graphical results display).

Signal Acquisition

Two possiblities for signal acquisition were released in the application program: one from the two-channel AD converter and the other from .MAT files. In both cases, transferring and receiving signal samples was done using SPOX SIG modules.

During program development and testing, signal samples were acquired from .MAT file, in which the signal samples were generated by the simulation model of spatial-time samples of the wavefront which has been previously stored in a file [6]. Acquired data were placed in current acquisition buffer (TAB), which was declared SPOX array object.

Complex Signal Generation

After a signal has been acquired, a complex signal is generated using Hilbert's transformation using FFT method. If L denotes the number of antenna elements in the antenna array and *N* denotes the number of time samples, the generation of the complex signal is executed with approximately $2LN\log_2\ (N)$ multiplication/addition complex operations.

Calculation of Covariance Matrix

Covariance matrix of the spatial samples of a signal is a complex square Hermit's matrix with dimension $(L\times L)$, [3], where L denotes the number of antenna elements in an antenna array. It is a simple for the implementation on the signal processor, because it multiplies the complex vectors only, which is directly supported by SPOX. The covariance matrix is calculated for approximately $N\ \ (L\times L)$ complex multiplication/addition operations.

Calculation of Eigenvectors and Eigenvalues

Calculation of eigenvectors and eigenvalues is the most critical and complex part of the implementation of the MUSIC method. Estimation of eigenvectors and eigenvalues is a base for a class of matrix type algorithms for digital signal processing.

The base of all computer efficient algorithms for calculation of eigenvectors and eigenvalues is the following relation of similarity:

$$\boldsymbol{E}^{-1}\boldsymbol{C}\boldsymbol{E} = diag(\lambda_1, \lambda_2, ..., \lambda_N) \tag{1}$$

where:

- $\boldsymbol{E}$ is a matrix of dimension $(N\times N)$ whose columns are eigenvectors of matrix ***C***
- diag (...) is a diagonal matrix whose diagonal elements are eigenvalues of matrix ***C***

The left part of the equation (1) represents a matrix similar to matrix ***C***.

A very useful theorem of the matrix algebra is: a matrix similar to the starting matrix has the same eigenvalues as the starting matrix.

From relation (1), it follows:

- Each matrix which has eigenvectors can be transformed by similarity operation to diagonal similar matrix.
- If a matrix is transformed in diagonal form, then the values on the diagonal of the transformed matrix are equal to eigenvalues of the starting matrix and columns of transformation matrix are equal to eigenvectors of the starting matrix.

Numerically efficient algorithm for calculating eigenvectors and eigenvalues is based on the matrix iterative digitalization. Elementary transformation of similarity is executed in each iteration. Tridiagonalization of matrix using Householder's transformation is the most computer efficient algorithm for estimating eigenvectors and eigenvalues based on diagonalization of the matrix. The tridiagonalisation of matrix with dimension $(N\times N)$ is executed with approximately $4/3\,N^3$ multiplication/addition operations. The second part of the algorithm (QL decomposition) is executed with approximately $3N^3$ arithmetic operations.

During the implementation of modules for estimating eigenvalues and eigenvectors, the routines *tred2.c* and *tqli.c*, given in [9], are used.

Having calculated eigenvalues and eigenvectors, the eigenvalues are sorted in descending order which is used to form the noise subspace matrix E_n, whose columns are the 'noise eigenvevtors' of the covariance matrix.

Calculation of MUSIC Algorithm

The algorithm for calculation of MUSIC spectrum is a simple of implementation, because only multiplication of complex vectors and matrix is needed, but its execution isn't computer efficient and depends on the number of points when calculating the MUSIC spectrum. For calculating the MUSIC spectrum in one point, approximately $2L^2$ complex multiplication/addition operations are needed.

Results of Implementation

The comparative results of the direction of arrival estimation based on the MUSIC method implemented on the signal processor TMS 320c30 and in MATLAB, are presented in the Figures 2 - 4.

The spatial-time signal samples are generated using the simulation model. It is supposed that antenna array is of circular geometry with four omni directional antenna elements in the array. The azimuth of arrival is 30 degrees, the signal to noise ratio is 30 dB and the number of signal time samples is 1024. The comparative results of the azimuth estimation by the MUSIC method implemented on the signal processor in 32-bit floating point arithmetic and in MATLAB in 64-bit floating point are presented in the Figures 2 and 3.

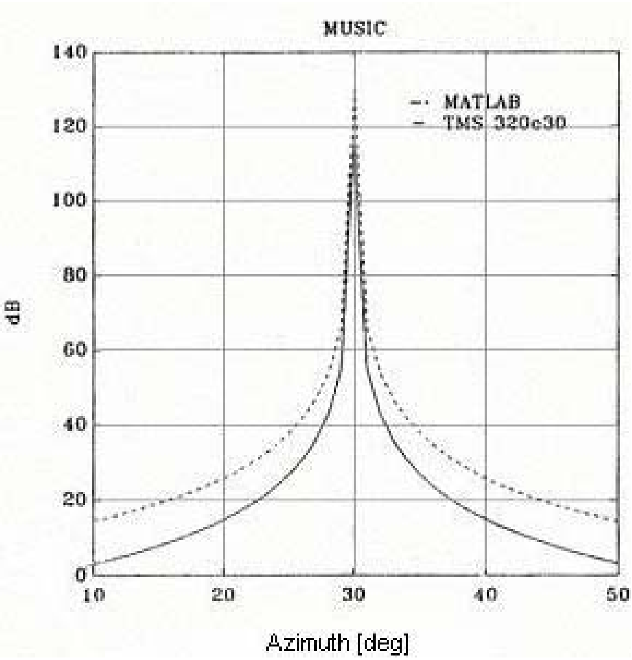

Figure 2. The comparative results of azimuth estimation based on the MUSIC method implemented on the signal processor TMS 320c30 (32-bit FLP) and in MATLAB (64-bit FLP)

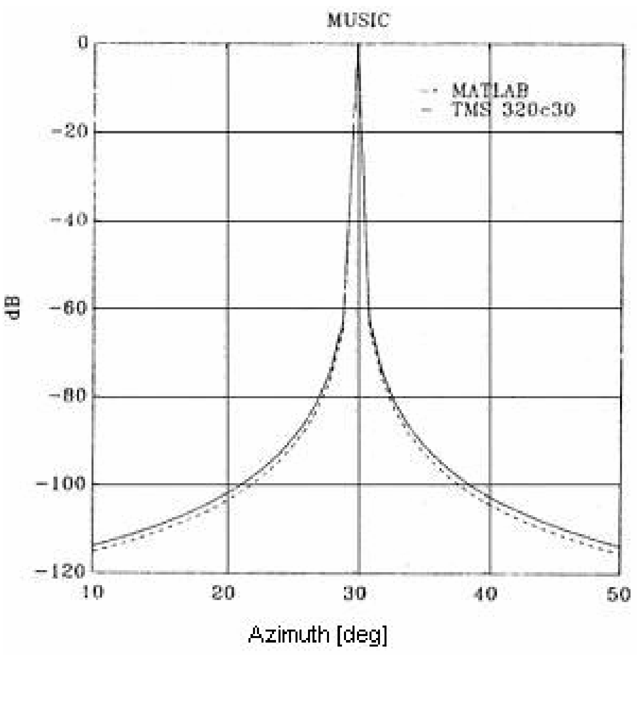

Figure 3. The comparative results of azimuth estimation (normalized); resolution of azimuth calculation is 1 deg

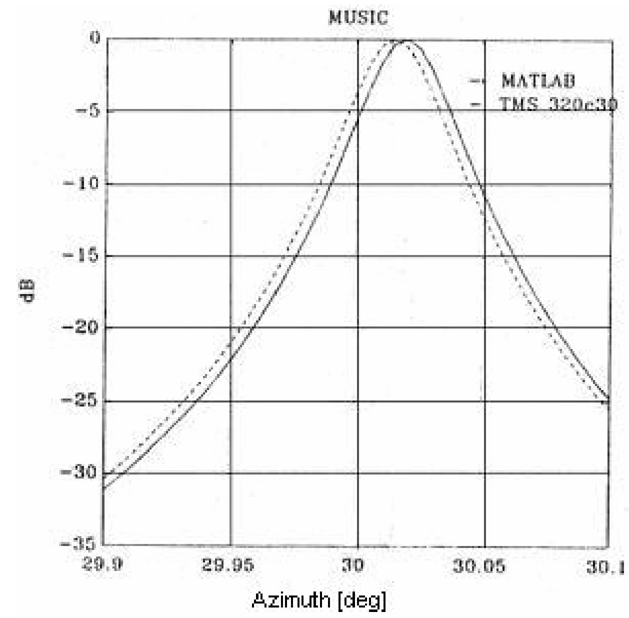

Figure 4. The comparative results of azimuth estimation (normalized); resolution of azimuth calculation is 0.005 deg

The maximum values of the MUSIC spectrum implemented on the signal processor is suppressed for about 10dB related to maximum value of the MUSIC spectrum calculated in MATLAB. But results are qualitatively very similar, which is especially well seen in the Fig.3, where the normalized MUSIC spectrums are presented. From Fig.4 it can be seen that the error of azimuth estimation is 0.015 degrees in MATLAB and 0.020 degrees on TMS, which means that the difference of direction estimation is less than 0.005 degrees in both cases.

Performances of the music method implemented on the signal processor TMS 320c30

The performances of high resolution methods for direction of arrival (azimuth and elevation) estimation are widely theoretically elaborated and evaluated in references. The question is whether the theoretical performances of the methods can be practically achieved.

From the aspect of spectral analysis, resolving very close spatial emitters is equivalent to separating near spectral components in a spatial spectrum. Most frequently, a relatively small number of signal spatial samples is available.

The most frequently used criteria for performance estimation of these methods are: direction estimation accuracy, resolution properties (separating spatial emitters), side lobe levels of the estimation function, sensitivity of the estimation algorithm to the change of parameters of antenna and received systems, sensitivity of the estimation algorithm to the coherency (correlation) of emitters, as well as the numerical complexity of methods. In this paper the direction estimation accuracy is compared in cases when MUSIC was implemented on the signal processor TMS320c30 in 32-bit floating point arithmetic and when implemented in MATLAB in 64-bit floating point arithmetic.

The standard deviation and mean value of estimation error are widely used as quantitative measures of accuracy.

There are three basic approaches for analyzing the performances of methods for direction of arrival estimation by the criteria of accuracy of estimation. The first approach is based on the theoretical analysis of the accuracy of estimation. The methods for direction of arrival estimation can be considered as methods for estimating the unknown parameters of the model of superposed signals on antenna array. The efficiency measure of any estimator of unknown parameters is a bias of the mean value and variance of the estimation error. When unknown parameters of the model are estimated (the direction of arrival is an unknown parrameter in the case of superposed emmiters model) , the estimation is conditionally unbiased if the mean value of the estimation error is equal to zero, [10]. An estimator is called '*estimator of minimum variance*' if the variance of parameter estimation error using this estimator is less than variance of estimation error using any other estimator. For the most of direction of arrival estimators a bias and variance of estimation error can be very difficult to express in an analytical form when the number of spatial and time samples of signals is finite. That's why Cramer-Rao's bound of variance of error for asymphtotic unbiased estimator (if it exists) is calculated instead of the real variance of error. For actual situations the theoretical Cramer - Rao's bound is usually calculated first and after that the variance of estimation error for the actual method (estimator), generated using the simulation, is compared with Cramer-Rao's bound. In the case of of the superposed signals model, the Cramer-Rao's bound can be expressed in an analitical form only for the linear antenna array. In the case of antenna array of arbitrary geometry, the calculation of this bound is very complicated and most often it is calculated numerically.

For the MUSIC method, which represents a specific practical and theoretical interest due to its good resolution properties, the explicit form for the theoretical value of the error variance is given in Porat and Friedlander's [13] and Stoica and Nehorai's [14,15,16] papers. These authors theoretically proved that the MUSIC method provides asymphtotic unbiased estimation of the direction of arrivals in the case of uncorrelated signals. Porat and Friedlander theoretically analyzed the relative asymphtotic efficiency of the MUSIC method, which is defined as the ratio of theoretic value of the standard deviation of the estimation error using the MUSIC method and Cramer-Rao's bound for the standard deviation (squared root of Cramer-Rao's boundary for error variance), [13]. The authors claim that the MUSIC method represents a very efficient estimator (in the most cases the estimation error deviation using the MUSIC method is about 20% bigger than Cramer-Rao's boundary for the standard deviation). Therefore, the theory gives the answer to the question about potential asymptotic performances of methods. The asymptotic characteristics of the methods are usually just a bad prediction of the real performances of methods for finite (small) number of spatial-time samples of signals.

The second approach estimation of the performances is experimental, which is the most expensive one. In order to perform a completely controlled experiment for direction of arrival estimation, it is necessary to control all the conditions of the experiment, which is very difficult to realize in practice. On the other hand, the referent antenna and receiver system need to be realized, which is equivalent to their development. It is interesting to observe that the authors of the MUSIC method have practically verified this method and its performances on the prototype of multichannel measuring system especially projected for the purpose [17].

The third approach to performance estimation is based on simulation. This approach is very useful especially if the theoretic borders of the potential performances are well known. Simulation is based on the mathematical model of the signal superposition on the antenna array and it is a very controllable experiment if the mathematical model is correct.

The Simulation Model of the Superposition of Radio Signals on the Antenna Array in a Given Frequency Sub-band and Time Observation Interval

The simulation model of the superposition of radio signals on the antenna array in a given frequency sub-band and time observation interval was developed and verified in a research project related to the methods and technical solutions for spatio-frequency signal analysis, identification and parameter estimation of radio signals. The model is realized in MATLAB and enables defining all parameters relevant for spatio modeling of signal superposition of radio signals. The simulation model is based on generalized spatial mathematical model of superposition of radio signal on the antenna array in the given frequency sub-band and time observation interval, which is formulated in [3].

The model provides antenna array parameter settings (number of antennas in an array, antenna array geometry), settings of parameters of the superposed radio-signals (number of signals, their spectral bandwidths, direction of arrivals on the antenna array (azimuth and elevation), signal/noise ratios) and settings of multipath parameters (number of replicas, relative delay and attenuation of each replica). The model provides different signal scenarios, which represent the real situation in the radio spectrum, including the situation when radio signals, which can be narrow-band and/or wide-band in spatio-time sense, are active in the same time and frequency subband. All parameters of the model are normalized so the model is applicable for different frequency subbands [3].

Many well known classical methods for direction of arrival estimation are included e.g. Watson Watt, "electronic direction finder", interferometer method in the time and frequency domain, method of correlative interferometry, Bartlet's method, as well as high resolution methods such as the method of Pisarenco, Capon's method of Minimum Energy, Eigen-Value method of Johnson and De Graf, Method of Minimal Norm and the MUSIC method.

The simulation model provides performance analysis of the previous mentioned methods for direction of arrival estimation. Many criteria such as direction estimation accuracy, ability to separate spatially closed radio emissions (the resolution properties), level of side lobs, sensitivity of method on coherency of emitter and sensitivity of methods to the change of parameters of antenna and receiving systems are applied. Most of the simulations for different scenarios and geometry of antenna array are done using the simulation model. The prototype development of MUSIC based direction finder for VHF/UHF range was based on the results of simulation.

Comparative Analysis of Performances of the MUSIC Method

Comparative analysis of performances of direction of arrival methods is especially interesting from the point of accuracy of direction of arrival estimation when a small (finite) number of time samples are available.

The practical points of interest in the phase of development of direction finder for VHF/UHF frequency range were as follows: which accuracy of the direction estimation can be achieved using antenna array with four antenna elements in the array, with signal duration of approximately 10ms, and signal/noise ratios which is expected in real situation in the radio spectrum.

The spatio-time samples of signals are generated using the simulation model. The simulated scenario was: the antenna array is linear and with four omni-directional antennas. One emitter is active, with the following normalized parameters: ratio of central frequency of signal and frequency on which antennas are on one half of wavelength (denote it as f_a) is 2/3; ratio of spectral bandwidths of signal and frequency f_a is 16/30000. The specified parameters correspond to the physical situation when the signal has a spectral bandwidth of 16 kHz, is centered on the frequency of 20 MHz and the distances between antenna elements in the array are equal to $\lambda/2$ of the frequency of 30 MHz. The radio signal is modeled as an MA random process. The test signals are generated using the simulation model. The number of spatio-time samples of wavefront is: 4 spatio x 16384 time samples, the signal to noise ratios (SNR) is 0,5,10 and 20 dB. The SNR is defined as σ_s^2/σ_n^2, where σ_s^2 is a variance of the signal and σ_n^2 is a variance of the noise on the antenna elements. It was presupposed that the noise on the antenna elements is the Gaussian white noise which is un-correlated between the antenna elements and with the same variance on each of the antenna elements. It was also presupposed that the signal of the emitter arrives on the antenna array from azimuth of 30 degrees and the elevation of 0 degrees.

Measurement of comparative accuracy of direction of arrival estimation provided by the MUSIC method implemented on the signal processor TMS320c30 in 32-bit floating point arithmetic and the MUSIC method implemented in MATLAB in 64-bit floating point arithmetic is performed. The mean value and standard deviation of the error of direction of arrival estimation are measured for SNR 0, 5, 10 and 20 dB and for an antenna array with 3 and 4 antenna elements in the array. Measurements are performed for specified SNR and specified number of antennas in an array, for signal duration of 64, 128, 256, 512 and 1024 samples of signals, which is equivalent to time duration of signals of 2, 4, 8, 16 and 32 ms.

The comparative presentations of standard deviation of the error of direction of arrival estimation based on the MUSIC method implemented in MATLAB and on the signal processor TMS320c30 and theoretic Cramer-Rao's bound for standard deviation of error for 4 antennas in an array and SNR of 0, 5, 10 and 20 dB are presented in Figures 5, 9, 13 and 17 respectively. Same values for the same SNR for 3 antennas in an array are presented in Figures 7, 11, 15 and 19. The comparative presentations of the measured relative statistic efficiency, for analyzed signal scenario, for the MUSIC method implemented in MATLAB and the MUSIC method implemented in the same conditions on the signal processor TMS 320c30, for SNR 0, 5, 10 and 20 dB and for 4 antennas are presented in Figures 6, 10, 14 and 18, and for 3 antennas in the Figures 8, 12, 16 and 20 respectively. The relative statistical efficiency is defined in [13] as:

$$\sqrt{\frac{\text{Variance of error of direction estimation by the MUSIC method}}{\text{Cramer Rao's bound for error variance}}} \qquad (2)$$

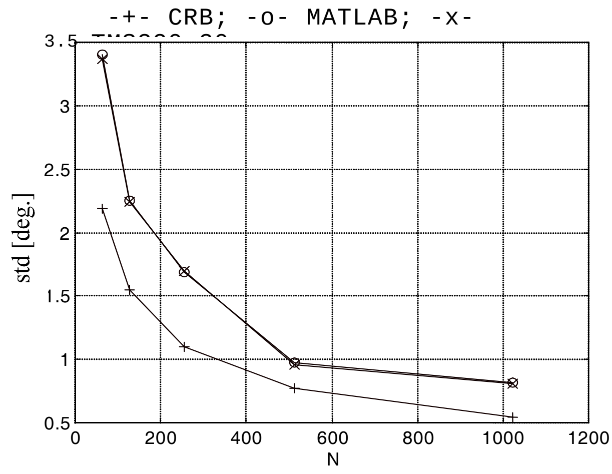

Figure 5. Standard deviation of error of azimuth estimation: 4 antennas; S/N = 0 db

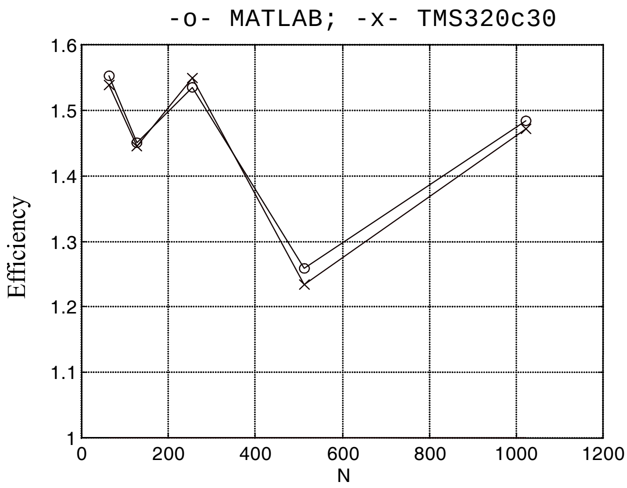

Figure 6. Statistical efficiency: 4 antennas; S/N = 0 dB

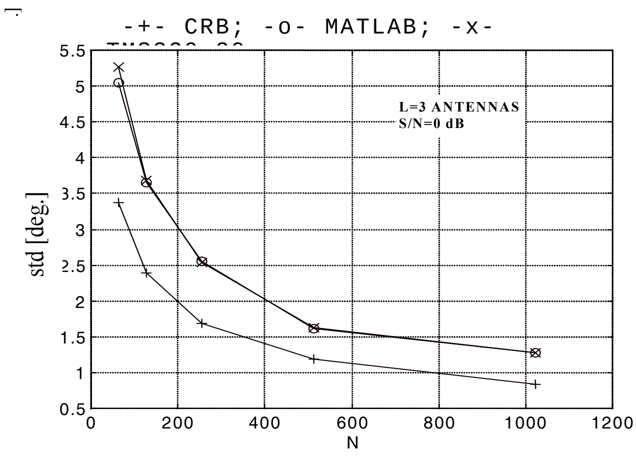

Figure 7. Standard deviation of error of azimuth estimation: 3 antennas; S/N = dB

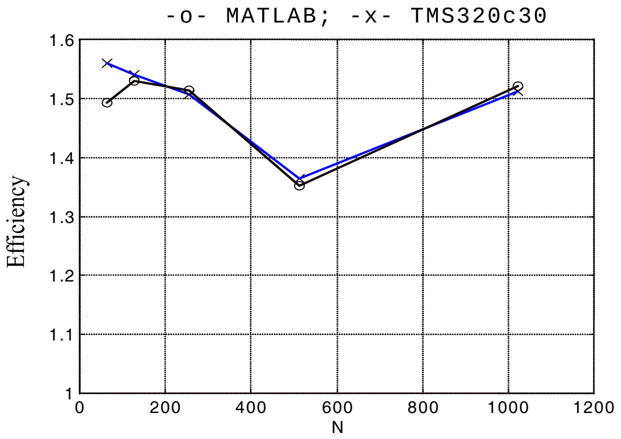

Figure 8. Statistical efficiency: 3 antennas; S/N = 0 dB

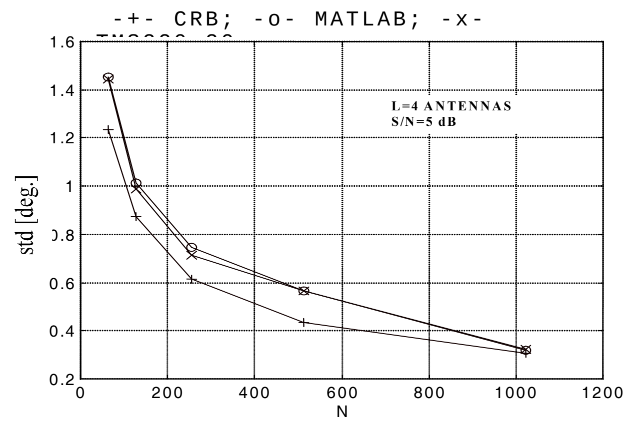

Figure 9. Standard deviation of error of azimuth estimation: 4 antennas; S/N = 5 dB

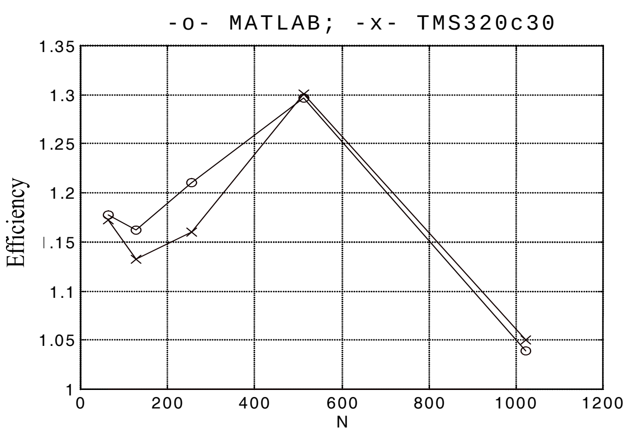

Figure 10. Statistical efficiency: 4 antennas; S/N = 5 dB

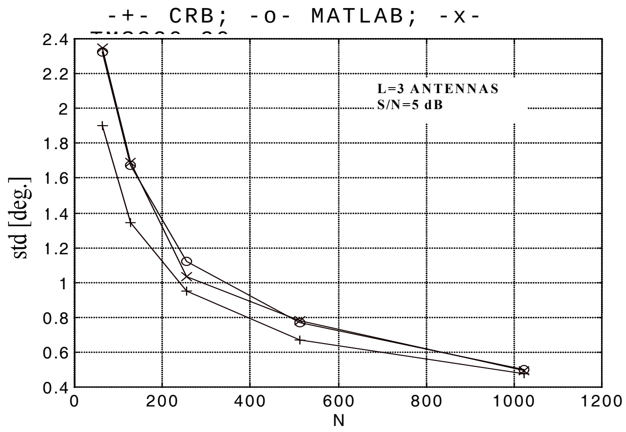

Figure 11. Standard deviation of error of azimuth estimation: 3 antennas; S/N = 5 dB

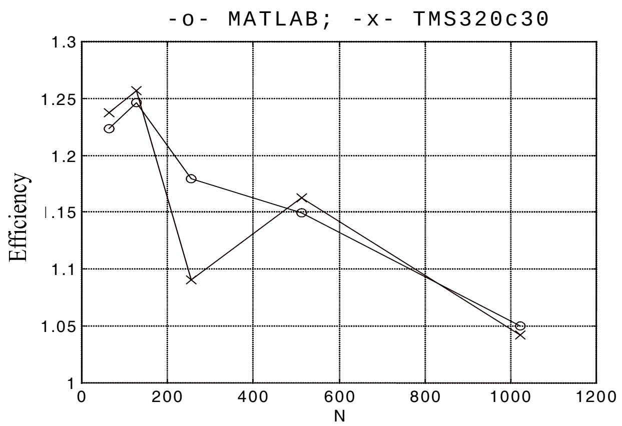

Figure 12. Statistical efficiency: 3 antennas; S/N = 5 dB

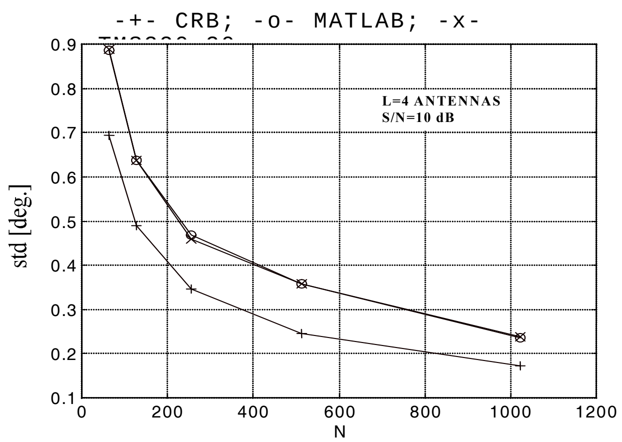

Figure 13. Standard deviation of error of azimuth estimation: 4 antennas; S/N = 10 dB

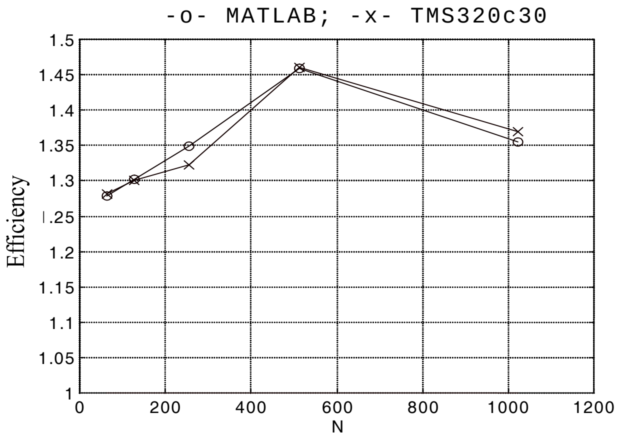

Figure 14. Statistical efficiency: 4 antennas; S/N = 10 dB

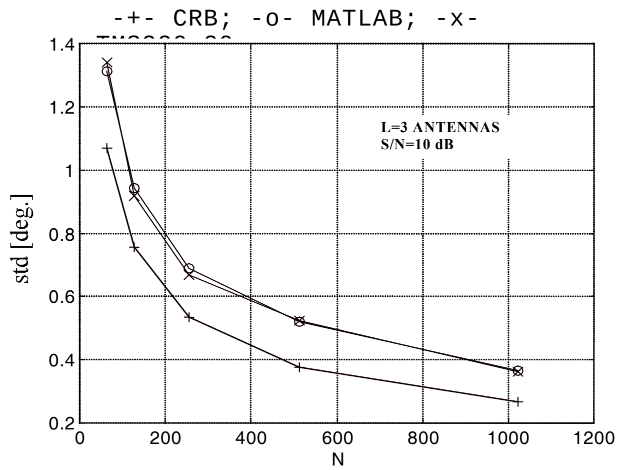

Figure 15. Standard deviation of error of azimuth estimation: 3 antennas; S/N =10 dB

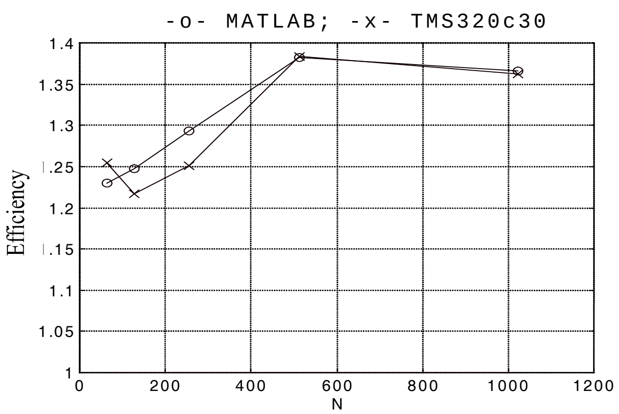

Figure 16. Statistical efficiency: 3 antennas; S/N = 10 dB

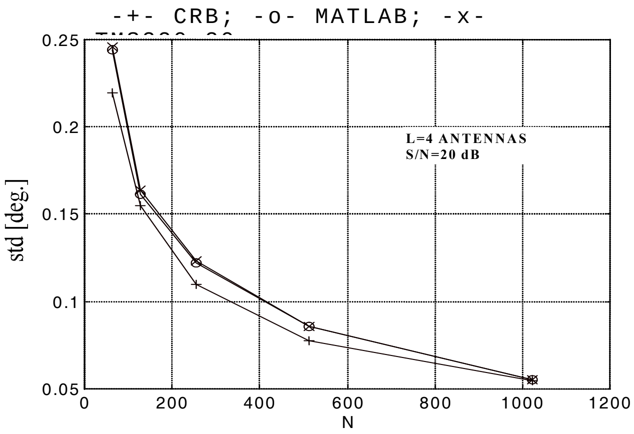

Figure 17. Standard deviation of error of azimuth estimation: 4 antennas; S/N =20 dB

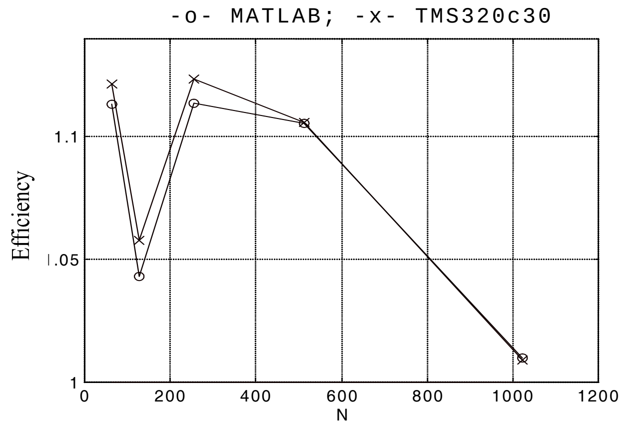

Figure 18 Statistical efficiency: 4 antennas; S/N = 20 dB

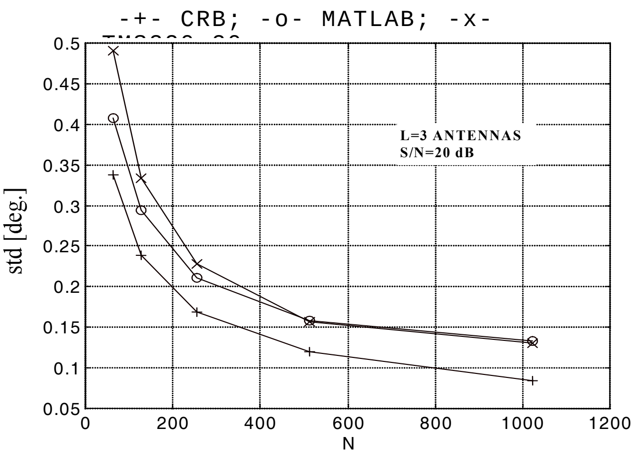

Figure 19. Standard deviation of error of azimuth estimation: 3 antennas; S/N =20 dB

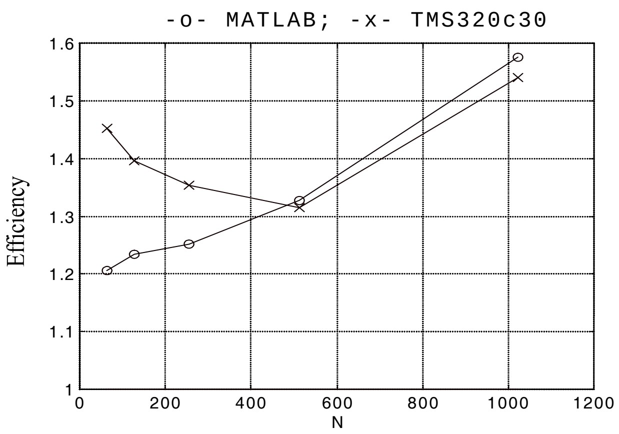

Figure 20. Statistical efficiency: 3 antennas; S/N = 20 dB

Conclusion

From the practical results of implementations of the MUSIC method on the signal processor TMS320c30 it can be concluded that:

- The 32-bit floating point arithmetic is enough for implementation of the MUSIC method. It has already been implemented in hardware in the first generation of digital floating point signal processors (as TMS 320c30). The achieved results of implementation with 32-bit accuracy are stable and qualitatively almost the same as results achieved in 64-bit floating point arithmetic.
- With the appearance of digital signal processors in floating point arithmetic, along with the possibility of the development of application programs in C language, the numerical complexity of the MUSIC method and the whole class of algorithms for signal processing of matrix type based on the MUSIC method, is not a serious barrier for real time implementation.

From the presented results of comparative performance analysis of the MUSIC method implemented on the signal processor TMS320c30, implemented in MATLAB and theoretical Cramer-Rao's bound for variance of error estimation, it can be concluded that:

- The curves of relative statistical efficiency of the MUSIC method, implemented on the signal processor TMS320c30 and implemented in MATLAB have almost perfect matches in all cases. It proves that practical implementation of the MUSIC method on the signal processor is qualitatively and correctly done.
- The curves of deviation of error estimation, when the MUSIC method is implemented both on the signal processor TMS320c30 and MATLAB insignificantly differ from Cramer-Rao's bound. This means that the implementation of the MUSIC method on the signal processor insignificantly degrades the potential performances of the MUSIC method. Relative statistical efficiency of the MUSIC method implemented on the signal processor doesn't overcome the value of 1.6, which means that the standard deviation of direction estimation error with the MUSIC method implemented on the signal processor is at least 60% bigger than Cramer-Rao's bound for the standard deviation of estimation error.
- Implementation of the MUSIC method on the signal processor TMS320c30 is a part of application software of the radio-direction finder RGK-2/3. Practical testing of the RGK-2/3 prototype proves that in practice the estimation error is better than 3 degrees. In the same conditions (3 antennas, signal/noise ratio 20dB and 1024 time samples, spatio-time samples of the signal are mathematically generated using the simulation model) deviation of error estimation of the direction of arrival using the MUSIC method is about 0.13 degrees. This means that the antenna and receiving system (mechanical tolerances of antenna systems, stability of characteristics of the used receivers), in a real system, dominantly degrades both potential performances of the MUSIC method and potential performance of the prototype of radio-direction finder RGK-2/3.
- Increasing the number of antennas in an antenna array and increasing signal/noise ratio dominantly contributes to the increasing accuracy of direction of arrival estimation.

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Praktična implementacija i procena performansi metode MUSIC implementirane na procesoru signala TMS 320c30

U radu su prikazani rezultati praktične implementacije i procene performansi metode MUSIC implementirane na procesoru signala TMS320c30 u 32-bitnoj aritmetici sa pokretnom decimalnom tačkom. Implementacija je realizovana u okviru istraživanja metoda za prostorno-frekvencijsku analizu radio-frekvencijskog spektra i za potrebe razvoja prototipa radio-goniometra za VHF/UHF frekvencijski opseg. Aplikacioni softver u kome je implementirana metoda MUSIC na signal procesoru je u isto vreme deo simulacionog modela superpozicije i procene parametara superponiranih radio-signala i deo aplikativnog softvera prototipa radio-goniometra.

Ključne reči: **MUSIC metoda, digitalna obrada signala, procesor signala, radio-goniometrisanje, radio-izviđanje, antenski nizovi.**

Практическая реализация и оценка характеристик метода MUSIC, реализованного на сигнальном процессоре TMS 320c30

В этой работе показаны результаты практической реализации и оценки характеристик метода MUSIC, реализованного на сигнальном процессоре TMS 320c30 в 32-разрядной арифметике с подвижной десятичной точкой (пунктом).

Реализация проведена в рамках исследования методов для пространственного-частотного анализа радиочастотного спектра и для надобности развития прототипа радиопеленгатора для ВЧ/УВЧ полосу частот. Прикладное программное обеспечение, в котором реализован метод MUSIC на сигнальном процессоре, одновременно является частью имитационного моделирования суперпозиции и оценки характеристик суперпозиционных радиосигналов и частью прикладного програмнного обеспечения прототипа радиопеленгатора.

Ключевые слова: MUSIC метод, цифровая обработка сигнала, сигнальный процессор, радиопеленгация, радиоразведка, антенные матрицы (массивы).