

# Investigation on Space-Time Signal Processing Technology Based on Data Channelization

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**Abstract** - High accuracy of parameter estimation and real-time processing is difficult in electronic support management signal processing with low SNR(Signal -Noise-Ratio). Traditional algorithms can be only effective for high SNR. In this work, classical spatial spectrum estimation is integrated with data channelization and a frequency and DOA (Direction-of-arrival) cascade estimation method based on data channelization is proposed. The simulations show that the algorithm is effective for SNR of  $-10$  dB and the computing speed is improved.

**Index Terms** - Digital Channel, Low Signal-to-Noise Ratio, Spatial Spectrum Estimation, Frequency and DOA Cascade Estimation

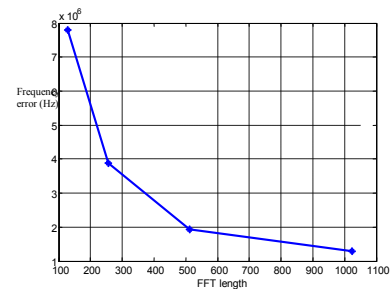
## I. Introduction

In the field of electronic warfare, high-resolution space-time two-dimensional spectral estimation technique for array signal [1] has attracted much attention in recent years.. Compared with the traditional surveillance equipment combing frequency-measure system with an array of direction-finding instrument, it is constructed by a set of array system which can measure the frequency of multiple signals and two-dimensional arrival angle at the same time. It can not only measure frequency and find direction simultaneously, but also has the advantages of high accuracy, high resolution and multi-signal processing capabilities. High-resolution space-time dimensional spectral estimation technology for array signal is based on high-resolution spatial spectrum estimation method. Spatial spectrum estimation algorithm is a high resolution and high accuracy method. But the performance of this algorithm is greatly reduced or failure for target signal with environmental noise at  $-10$  to  $-30$ . Order cumulants [1,2,6], and other algorithm have high performance for low SNR, However, they can only apply to long snapshot data and non-Gaussian signals and cannot apply to real-time processing and Gaussian signals., In this study, a real-time processing program for measuring frequency and finding direction is established on the basis of digital channel. Meanwhile, a frequency measurement and azimuth measurement cascade algorithm is proposed. This algorithm is able to achieve high precision, high real-time frequency measurement and angle measurement at low SNR and limited length of snapshot data.

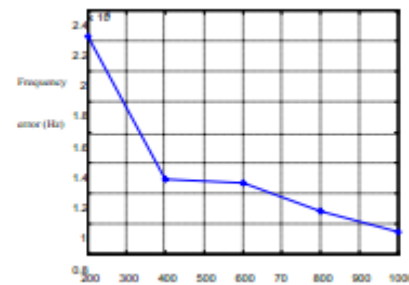
## II. Traditional frequency measurement scheme

Frequency measurement is the first parameter measurement issues to be considered, because the angle parameter estimates must know the frequency of the signal, the measurement of other signal parameters and estimates

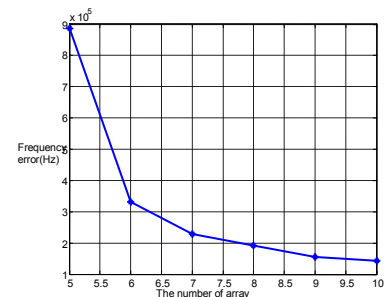
also are affected. In this section, combined with the system, a comparative analysis of indicators on different measurement of frequency. System indicator, the frequency measurement error of single- frequency sine signal should be better at 60KHz. The conventional methods including FFT and MUSIC super-resolution algorithms are unable to reach this target. This description of the simulation results. The simulation has three narrow-band signal frequencies are 128MHz, 266MHz and 442MHz, the SNR 3dB, the sampling frequency of 1GHz. FFT and MUSIC estimation results shown in Figure 1.



(a) FFT method estimates



(b) When the number of array element is 5, the results by MUSIC method



(c) When the snapshots is 1000, the results by MUSIC method

figure1 The result by Conventional frequency measurement method

The figure shows the FFT method in the length of the 1024 error of about 1MHz, MUSIC method in the number of array elements for the five conditions, the low number of snapshots, the estimation accuracy higher than the FFT, the number of snapshots increasingly large precision the convergence of FFT method, to expand the number of array elements, the MUSIC method is accurate to improve accuracy in the number of 10 elements up to 100KHz or so, but as the number of array elements to improve and enhance the precision slow down, the amount of computation has increased significantly. Therefore concluded that the traditional FFT and MUSIC methods are unable to meet the index requirements. The reasons for this situation can be attributed to the high sampling frequency. Whether FFT or the MUSIC frequency measurement of the digital signal is essentially to estimate the phase difference between the signal sampling points, while the two phase difference between the sampling points. Formula:

$$\varphi = \frac{2\pi f}{f_s} \quad (f_s \text{ is the sampling frequency}) .$$

It can be seen from the formula, the phase measurement error and frequency measurement error factor is  $f_s$  multiples relationship. Small phase estimation error will bring greater frequency estimation error. Sampling frequency 1GHz, for example, the frequency measurement error than the phase error to expand about 109 times, which resulted in the high frequency error. It is an effective means to reduce the sampling frequency in order to improve the accuracy of frequency measurement.

### III. Channels of frequency measurement

The channel frequency measurement is carried out on the basis of the channelized, the coarse frequency measurement results is obtained through channelized, while the frequency of the signal after fine measurement. Assuming the signal bandwidth of 500MHz digital signal for 64 channels, each channel width  $500M/64 = 7.8125MHz$ . In other words, channelized, the results of the signal frequency error is less than 4MHz. Known by the channel process, after a number of times the subsampling 1GHz sampling the signal in the channel process, which in turn provides the basis for the frequency fine measurement. But this brings a vague question.

Since the frequency precision measurement can not be separated due to subsampling, subsampling will bring fuzzy. Consider the use of a crude measurement results efuzzification, using under-sampling methods for frequency fine measurement. The principle of the method shown in Figure 2.

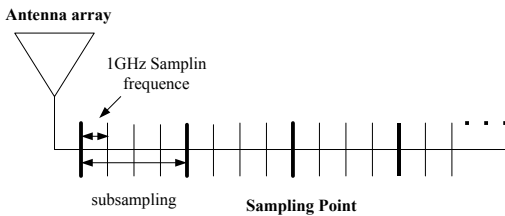


Figure 2 The principle of frequency measurement

Of course, in the presence of multiple signals, this approach to achieve a coarse measurement results paired with precise and accurate results, too complicated pairing process is difficult for practical, so we intend to use the ESPRIT(Estimation of Signal Parameters via Rotational invariance techniques) method to get a rough measurement and precision measurement of the signal frequency at the same time, as shown in Figure 3:

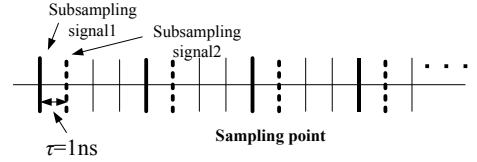


Figure3 Using the ESPRIT to get a rough measurement and precision measurement of frequency

Located subsampling signal 1 for the  $X_1$ , and subsampling signal 2 for the  $X_2$ . Between  $X_1$  and  $X_2$  delay  $\tau = 1ns$ .  $X_1$ ,  $X_2$  is essentially drawn from the 1GHz sampling frequency sampling points under-sampling signal for the frequency of precise and accurate, two sets of sampling points difference between the sampling points can be used for the frequency of rough measurement. Rough measurement and precision measurement results by the ESPRIT technical get automatic matching, thus eliminating the complexity of the matching process. The theoretical analysis is as follows:

Set time-domain element of  $M$ ,  $X_1$  and  $X_2$  written in matrix form:

$$X_1 = [a(\theta_1), \dots, a(\theta_N)]S + N_1 = AS + N_1 \quad (1)$$

$$X_2 = [a(\theta_1)e^{j\phi_1}, \dots, a(\theta_N)e^{j\phi_N}]S + N_2 = A\Phi S + N_2 \quad (2)$$

Where  $A$  is the array manifold of the under-sampling signal in time domain,  $N$  is the number of signal sources,  $S$  is the signal complex envelope,  $N_1$ ,  $N_2$ , for the white Gaussian noise, power is  $\sigma^2$ .in the formula.

$$\Phi = diag[e^{j\phi_1}, \dots, e^{j\phi_N}] \quad (3)$$

$\Phi$  is sampling point phase. Obviously,  $A$  contains precise and accurate results under-sampling frequency, and  $\mathcal{C}$  is the coarse frequency measurement results. After a simple proof can be obtained the following conclusions:

$$R_2 R_1 A = A\Phi \quad (4)$$

$$\text{In the formula, } R_1 = E\{X_1 X_1^H\} - \sigma^2 I = AR_s A^H,$$

$$R_2 = E\{X_2 X_1^H\} = A\Phi R_s A^H.$$

Prove:  $R_1 = AR_s A^H$ :  $A$ , the equation on both sides of left multiplication  $(A^H A)^{-1} A^H$  can get

$$(A^H A)^{-1} A^H R_1 = (A^H A)^{-1} A^H AR_s A^H = R_s A^H \quad (5)$$

The above equation into the R2, :

$$\mathbf{R}_2 = \mathbf{A}\Phi\mathbf{R}_s\mathbf{A}^H = \mathbf{A}\Phi(\mathbf{A}^H\mathbf{A})^{-1}\mathbf{A}^H\mathbf{R}_1 \quad (6)$$

$$\mathbf{R}_2\mathbf{R}_1^{-1} = \mathbf{A}\Phi(\mathbf{A}^H\mathbf{A})^{-1}\mathbf{A}^H \quad (7)$$

Both sides of the right multiplication A, we can get :

$$\mathbf{R}_2\mathbf{R}_1^{-1}\mathbf{A} = \mathbf{A}\Phi \quad (8)$$

This shows that the eigenvalue and eigenvectors of  $\mathbf{r}_1, \mathbf{r}_2$  respectively for  $\mathbf{A}$  and  $\Phi$ . Correspond to eigenvalue and eigenvectors, thus avoiding the complexity of the matching process. By the analysis of the previous section shows that the traditional measurement of frequency is not only volume calculate, and higher signal to noise ratio is required, the measured frequency of a single channel has high real-time, high accuracy of frequency measurement, but it requires a multi-channel parallel processing. And due to the broadband characteristics of the array, Angle measurements related to the signal carrier frequency and angle of arrival of the solution of the coupled problem. The entire bandwidth is divided into dozens of narrow-band band, angle estimation is an effective method in the narrow-band band, but the required hardware and software cost much. In this regard, the proposed program is estimated based on the frequency and angle of cascade, its essence lies in the design of filters based on frequency measurements, without considering the other band signal that does not exist, so you can save on hardware and software cost of effectively.

This method focuses on the design of the filter. In the estimate of the different frequency signals can be seen as the process of joint estimation of signal angle and frequency. However, joint estimation have excellent performance, but the required computation, can not achieve real-time. Thus many of its alternative dimensionality reduction methods is put forward. The conditions of this method is the use of the signal frequency is known (estimated to be), convert the jointly estimated for the cascade [4,5], effectively reducing the amount of computation. Through theoretical analysis[1], the frequency angle cascade is estimated to be equivalent to joint estimation of two-dimensional parameter filtering, filter conversion for joint noise subspace weighted. Specific process are shown in Figure 4.

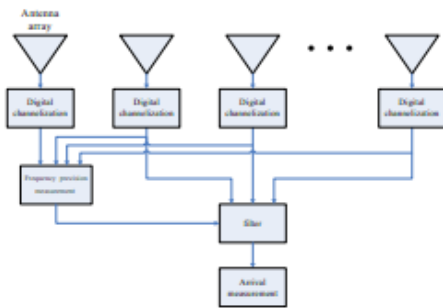


figure 4 frequency and arrival are estimated to cascade

Set the frequency parameter  $\omega$ , the angle parameter of  $\theta$ . two-dimensional MUSIC algorithm to two-dimensional search in the whole  $(\omega, \theta)$  parameter plane,  $f(\omega, \theta)$  is the largest  $(\omega_i, \theta_i)$ , which is the  $i$  th signal parameter.

$$f(\omega, \theta) = \frac{1}{\mathbf{a}(\omega, \theta)^H \mathbf{G}\mathbf{G}^H \mathbf{a}(\omega, \theta)} \quad (9)$$

Where  $\theta(\omega, \theta)$  for two-dimensional array in the space-time manifold.  $\mathbf{G}$  is a two-dimensional noise subspace.

Two-dimensional array in the space-time manifold with a  $(\omega, \theta)$  is written as:

$$\mathbf{a}(\omega, \theta) = \mathbf{A}_\omega(\omega)\mathbf{a}_\theta(\theta) \quad (10)$$

Among them,  $\mathbf{a}_\theta(\theta)$  is the airspace array manifold,  $\mathbf{A}_\omega(\omega)$  is the part of two-dimensional array manifold which does not contain space-time airspace information, for a diagonal matrix. Of course, the airspace array manifold still contains the frequency information, which is the coupling of frequency and angle information at the frequency measurement accuracy thus affecting the angle measurement accuracy.  $\mathbf{a}_\theta(\theta)$  and  $\mathbf{A}_\omega(\omega)$  according to the different space-time array structure decisions. Then:

$$\begin{aligned} f(\omega, \theta) &= \frac{1}{\mathbf{a}_\theta(\theta)^H \mathbf{A}_\omega(\omega)^H \mathbf{G}\mathbf{G}^H \mathbf{A}_\omega(\omega)\mathbf{a}_\theta(\theta)} \\ &= \frac{1}{\mathbf{a}_\theta(\theta)^H \mathbf{G}_\omega(\omega)\mathbf{G}_\omega(\omega)^H \mathbf{a}_\theta(\theta)} \end{aligned} \quad (11)$$

The denominator of  $f(\omega, \theta)$  can be written as  $\mathbf{a}_\theta(\theta)^H \mathbf{A}_\omega(\omega)^H \mathbf{G}\mathbf{G}^H \mathbf{A}_\omega(\omega)\mathbf{a}_\theta(\theta)$ . This shows that when the spectrum estimation of the angle parameter, frequency parameter can be seen as a weighting factor of the noise space. Therefore, the accurate time-domain parameters, substituted into the above equation, two-dimensional space-time noise subspace is weighted by one-dimensional parameter estimation of DOA parameters.

#### IV . Simulation Analysis

This section by computer simulation to verify the correctness of this paper, and assess the angular measurement error. Simulation 1: In order to investigate the ability to distinguish signals of different frequencies, simulation parameters set as follows: three narrow-band signal, which frequency is 122MHz, 206MHz and 250MHz, the angle parameter  $(0^\circ, 0^\circ)$ 、 $(25^\circ, 10^\circ)$  and  $(10^\circ, -5^\circ)$ . Sampling length of  $1\mu\text{s}$ , sampling frequency 1GHz, the number of domain array is 6 and 10 times under-sampling program of non-channelized. Only Select a signal. In comparison, the figure also draw the estimated results of the non-channel

program, in which bandwidth is divided into 64 channels. Angular measurement method using MUSIC method and the use of non-uniform linear array in the two programs.

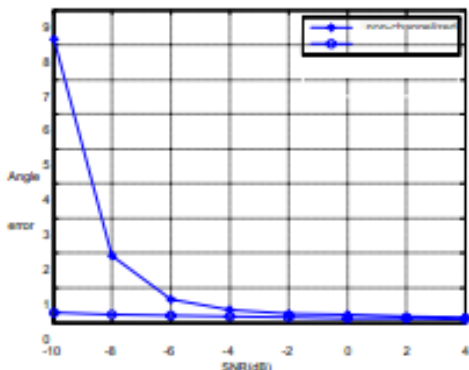
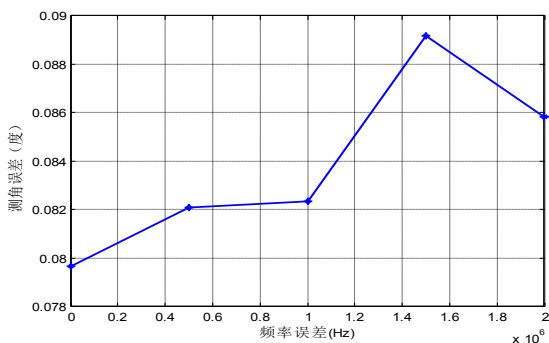


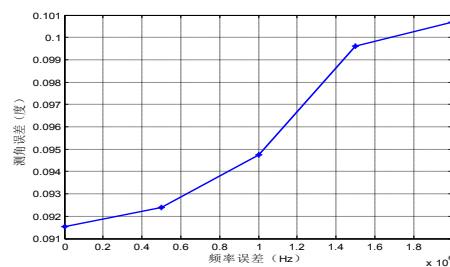
Figure 5 the Comparison of angle error

It can be seen from Figure 5, only when the signal to noise ratio greater than 2dB, the capability of non-channel and channel scheme are uniform, in low SNR, the program's parameters estimated performance is poor, far well short of targets, and therefore not considered in this project.

Simulation2: taking into account the precision of the filter and frequency measurement in this process, we investigated the impact of the frequency measurement accuracy for the angle measurement accuracy. Parameter settings as above. When the frequency measurement accuracy for Hz, the angle measurement accuracy shown in Figure 6.



(a) azimuth



(b) Pitch

Figure 6 Influence of frequency error on measuring errors

It can be seen from the figure, with the increase of the frequency error, the angle measurement error also showed an increasing trend, but the modest increase in angle measurement error: When the frequency of errors increased from 0 to 2MHz, the angular measurement error is only 0.01 degrees. This is because, although the frequency error in the 2MHz even greater, but still very small relative to the terms of the signal carrier frequency. From the previous simulation results shows that, even if it is a coarse frequency measurement results, the error is much smaller than 2MHz. Therefore, the frequency error of angle measurement accuracy can be neglected.

## V. Conclusion

In this paper, a digital channel and spatial spectrum analysis technology combined with the array signal processing method. The algorithm uses the technology of the digital channel data preprocessing, First digital channelized coarse frequency measurement, then the spatial spectrum algorithm for precision measurement frequency and DOA (azimuth) is estimated. Which is characterized by two-dimensional parameter measurement and estimation into cascade of one-dimensional parameter measurement and estimation, which greatly reduces the computation, the use of spatial spectrum estimation algorithm to improve the signal parameter estimation of signal to noise ratio. Simulation results demonstrate the effectiveness of the algorithm. The algorithm has low computational complexity, high precision, able to adapt to the harsh environment of low SNR, with a certain degree of practicality.

## References

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