**Final Report: Application Research on DOA Estimation Based on Software-Defined Radio Receiver**

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| **Introduction**  Direction of arrival estimation is an active area in array signal processing. It has a wide range of applications in the fields of communication, radar, detection and navigation. Many researchers have innovated based on the DOA estimation algorithm, and most of the algorithms have been verified on the simulation platform. As we all know, the experimental results of the simulation platform are different from those in engineering applications. The article we got focuses on the engineering application of DOA estimation. It uses KerberosSDR equipment and four omnidirectional antennas as signal receivers, and uses Raspberry Pi as a data processor to implement a system with a simple structure and reliable DOA estimation performance.  https://imgconvert.csdnimg.cn/aHR0cHM6Ly9pbWcyMDE4LmNuYmxvZ3MuY29tL2Jsb2cvNTMzOTMzLzIwMTgxMi81MzM5MzMtMjAxODEyMDYxNjI5NTI2NjEtOTg4ODYyMjA3LnBuZw?x-oss-process=image/format,png  Fig.SDR schematic  KerberosSDR is a new 4-input coherent RTL-SDR. RTL-SDR is a very cheap software-defined radio receiver. Each RTL-SDR consists of an RTL2832U chip and an R820T tuner. It can receive radio frequency signals from 25MHz to 1.75GHz and convert them to baseband. Finally, an 8-bit digital sampling signal is output from the USB port. There is a noise source module inside KerberosSDR, which can realize the sampling time synchronization and phase synchronization of the four signal receiving channels.    Fig. Raspberry Pi schematic  The four signal receiving channels share a clock source, and the four digital signals communicate with the Raspberry Pi through the USB HUB. Run the signal processing algorithm on the Raspberry Pi, and display the DOA estimation results and signal strength in real time through the web page.  Teacher Wu put forward higher requirements for the DOA algorithm in our paper, and a dynamic signal source was used in the paper. In the paper, the communication frequency between the drone and the remote controller is 2.400-2.4835 GHz, which is not within the RTL-SDR receiving frequency range. Therefore, the small FM transceiver is fixed on the drone as the signal source, and the transmission frequency is the FM transceiver. The frequency is 446.0063MHz. A UAV equipped with an FM transceiver is hovering in the air to ensure that the signal sent by the FM transceiver will not be blocked by obstacles. This signal source is used in the paper to verify the accuracy of the DOA estimation of the system.    Fig. Schematic diagram of the experiment in the paper  In this experiment, we have also made some attempts on the detection process of mobile signal sources, but the effect is not good. We will discuss the problems we encountered in detail in the article. In the experiment, we conducted two rounds of technical demonstration. The first round of technical demonstration was the part of the demonstration related to the selection of the signal source, and the second part was the technical demonstration of our choice of experimental measurement points. After two rounds of technical demonstrations, we finally chose the experimental method of detecting fixed signal sources, and we also had certain considerations for site selection.  **Theoretical knowledge:**  The formal proposal of the MUSIC algorithm in 1986 was a milestone event in the field of spectrum estimation. It changes the design idea, starts with the eigendecomposition of the signal correlation matrix, breaks through the theoretical Rayleigh limit of the traditional FFT-based spectrum estimation technology, and obtains high spectrum estimation accuracy. In this period and a later period of time, deterministic maximum likelihood method (DML), random maximum likelihood method (SML), weighted signal subspace fitting method (WSF), noise subspace fitting method (WNF), etc. were successively With mature development, super-resolution spatial spectrum estimation has thus formed a relatively independent technical research direction, which has attracted increasing attention and attention.  In this part, we first analyze and introduce the three spatial spectrum direction finding algorithms of MUSIC, DML, and WSF, focusing on the analysis and comparison of direction finding accuracy, calculation amount, implementation complexity, and coherent signal direction finding capabilities, which are closely related to engineering practice. Key indicators. Then confirm the advantages and disadvantages of each algorithm through simulation verification.  **1 Analysis and comparison of three spatial spectrum direction finding algorithms**  **1.1 Introduction to three types of spatial spectrum direction finding algorithms**  The MUSIC algorithm is the most frequently mentioned spatial spectrum direction finding algorithm. The algorithm has the same frequency multi-signal direction finding capability and high resolution direction finding capability. Compared with other spatial spectrum direction finding algorithms, the MUSIC algorithm is relatively simple in calculation and stable in performance, so it has become the most widely used spatial spectrum algorithm in engineering practice. Maximum Likelihood Estimation (MLE: Maximum Likelihood Estimation) is a widely used parameter estimation method, which is widely used in communication, radar, navigation and other fields. It has also been applied in the field of spatial spectrum direction finding and achieved good results. . The basic idea of ​​the ML algorithm is: the likelihood function of the received signal is defined as a conditional probability density function containing the parameter to be estimated, so that the likelihood function gets the maximum estimated parameter value, which is the best estimated parameter. In spatial spectrum estimation, this estimated parameter is the angle of incidence. According to the statistical characteristics of the incident signal, it is divided into two algorithms: deterministic maximum likelihood (DML) and random maximum likelihood. When the incident signal is assumed to obey the Gaussian random distribution model, the random maximum likelihood algorithm is obtained; when the incident signal model is an unknown deterministic model, the deterministic maximum likelihood algorithm is obtained.  Suppose the model of the signal is as follows:  Among them, X(t) is the received signal, A(θ) is the array manifold matrix, s(t) is the received signal, and N(t) is the noise signal. Represents the angle of multiple incident signals.  The deterministic maximum likelihood algorithm can be boiled down to finding the angle of incidence θ so that the following formula obtains the extreme value:  The random maximum likelihood algorithm can be boiled down to finding the angle of incidence, so that the following formula obtains the extreme value:  Among them, is the autocorrelation matrix of the received signal, M is the number of arrays, and N is the number of signals. is the orthogonal projection matrix of array manifold matrix A, is the projection matrix of matrix A, and is the generalized inverse matrix of matrix A. tr is the trace of the matrix. det is the determinant of the matrix to be solved.  The design idea of the signal subspace fitting algorithm is to find an incident angle estimation parameter so that the array manifold matrix closely related to the estimated parameter and the signal subspace of the received signal are as similar as possible (fitting). Using the Frobenius norm of the matrix and some of the unique properties of the orthogonal projection matrix, the weighted signal subspace fitting algorithm can be reduced to finding, so that the following formula finds the best value:  Where is a weighted value, . Where is a diagonal matrix composed of eigenvalues corresponding to the eigenvectors of the signal subspace.  **1.2 Comparison of three types of spatial spectrum direction finding algorithms**  The predecessors have carried out detailed analysis, comparison and summary on the three types of algorithms. The main conclusions are as follows:  (1) The parameter estimation variance of each algorithm and the Kramer-Labor Bound (CRB) have the following relationship, that is, the parameter estimation performance of the WSF and SML algorithms is the best, the DML algorithm is in the middle, and the MUSIC algorithm is the worst. In special cases where certain conditions are met, the estimated variances of the three can be equivalent.  (2) The MUSIC algorithm cannot solve the direction finding problem of co-frequency coherent signals, while DML, SML, and WSF can well solve the co-frequency coherent signal direction finding problem, so as to obtain better direction finding performance.  (3) The computational complexity of the MUSIC algorithm is relatively minimal. The main computational complexity is focused on the eigendecomposition of the signal correlation matrix and some matrix multiplication operations, and the matrix eigendecomposition only needs to be operated once. The solution of the maximum likelihood estimation algorithm and the subspace fitting algorithm is a non-linear optimization problem in a multi-dimensional space, which usually requires the iterative solution of the optimization algorithm. Iterative solution means multiple operations such as matrix inversion and multiplication, which increase the amount of computation.  **2 Simulation verification of three spatial spectrum direction finding algorithms**  The simulation mainly verifies several key characteristics of the three typical spatial spectrum direction finding algorithms of MUSIC, DML, and WSF, including direction finding accuracy, same frequency multi-signal direction finding ability, and coherent signal direction finding ability. In addition, the simulation verification also focuses on two issues that have a major impact on the direction-finding equipment and product development, namely the tolerance of phase errors between receiver channels and the influence of the aperture-wavelength ratio on the direction-finding results.  The simulation conditions are set as follows: 9 The wavelength ratio of the antenna circular array and the aperture is set as the ratio of the radius of the circular array to the wavelength of the incident signal. The angular interval between the three incident signals is set to 20 degrees. Investigate the changes in the direction finding accuracy of the three direction finding algorithms in two dimensions. The first dimension is the aperture-wavelength ratio, and the second dimension is the phase error between channels. At the same time, two scenarios of 3 signal fully coherent and 3 signal incoherent are investigated, so there are 6 simulation verification results for 3 direction finding algorithms.  For the first dimension: the phase error between the 9 channels is randomly selected, and the maximum phase error ranges from 0 to 20 degrees in 0.2 degree steps; for the second dimension: the aperture-wavelength ratio is in the range of 0.1 to 1, which is The value is 1/r, the value of r ranges from 10 to 1, and the decreasing step value is 0.2; the aperture-wavelength ratio is in the range of 1 to 11, and the increasing step value is 0.2. The simulation results are shown in Figure 1 to Figure 6, where the X and Y axes represent the phase difference between the channels and the aperture wavelength ratio, respectively, and the Z axis represents the final statistical error of direction finding. It should be noted that the numerical unit of the X axis in the figure is 0.2 degrees. The value on the Y axis represents the serial number of different aperture wavelength ratios. The value 50 corresponds to the radius of the circular array equal to the wavelength of the incident signal.    Fig. MUSIC algorithm simulation verification results (non-coherent scenario)  Fig.DML algorithm simulation verification results (non-coherent scenario)  Fig. SSF algorithm simulation verification results (non-coherent scenario)  Fig. MUSIC algorithm simulation verification results (coherent signal scenario)  Fig. DML algorithm simulation verification results (coherent signal scenario)  Fig. SSF algorithm simulation verification results (coherent signal scenario)  Through the analysis of the simulation results, we can know:  (1) The MUSIC algorithm can perform direction finding of the same frequency incoherent signal well. When the aperture-wavelength ratio is greater than or equal to 1, the MUSIC algorithm also has a strong tolerance for phase errors between channels. But without additional measures, the conventional MUSIC algorithm cannot cope with coherent signal scenarios.  (2) The DML and WSF algorithms can perform the direction finding ability well when the aperture-to-wavelength ratio is approximately equal to 1, but when the aperture-to-wavelength ratio is greater than 1, they cannot effectively find the direction.  (3) With DML and WSF algorithms, a wide-band direction finding can be achieved only when the phase consistency between channels is close to ideal. Since the usual direction finding equipment needs to have the direction finding ability of a wider frequency band, in order to realize the direction finding ability of the wide frequency band, these two types of algorithm schemes need to ensure that the phase error between the channels is small. This requires the direction-finding equipment to have sufficient ability to measure and correct phase errors between channels. The DML and SSF algorithms are very sensitive to changes in the aperture-wavelength ratio, which adds difficulty to the design of direction-finding antennas and direction-finding equipment.  **Simulation**  **DOA(Direction Of Arrival)**  **Introduction**  Suppose the system have M antennas, N signal packages, K targets.  Begin with time difference, if the signal arrives at ULA with angle , from the figure we can notice that there are different s, which causes phase difference , where c is the propagation speed of light, m is the number of arrays.  图表, 雷达图  描述已自动生成  We can induce the formula of arrived signal , assume there is only one signal package  We can simplify  When it comes to N packages, they come from N different directions:  **The simplest DOA estimation: spatial Fourier transform**  The form of the received signal  Although we don't know the angle of the signal, for a given array, the mathematical form of its steering vector is known. For example, for ULA, it must be of Vandermonde structure. Based on this, we have a method of DOA estimation.  Specifically, we can construct a steering vector, the angle of which may be given as α, then we can construct a steering vector with the incoming wave direction α as  Use our assumed steering vector a(α) and the received signal to do the vector inner product, that is    The result should be a scalar. A simple calculation can get    The equal sign is taken at α=θ.  From this inequality, we can see that if we are right, that is, α=θ, then the result obtained is a maximum value.  Therefore, we can guess all the angles again and find the one with the largest result. The corresponding angle is the result of our DOA estimation.  Here can lead to a method of DOA estimation, the pseudo code is presented as follows:    **Algorithm simulation example**  Example 1: Assuming that there is only one target at θ1=5°, the result is  https://img-blog.csdnimg.cn/20200511222035229.jpg?x-oss-process=image/watermark,type_ZmFuZ3poZW5naGVpdGk,shadow_10,text_aHR0cHM6Ly9ibG9nLmNzZG4ubmV0L3B3YW5nOTU=,size_16,color_FFFFFF,t_70#pic_center  Example 2: Assuming that the two targets are respectively located at θ 1 = 5 °, θ 2 = 10 °, the result obtained by the above method is  https://img-blog.csdnimg.cn/20200511222153895.jpg?x-oss-process=image/watermark,type_ZmFuZ3poZW5naGVpdGk,shadow_10,text_aHR0cHM6Ly9ibG9nLmNzZG4ubmV0L3B3YW5nOTU=,size_16,color_FFFFFF,t_70#pic_center  Example 3: Assuming that the two targets are located at θ 1 = 5 °, θ 2 = 30 °, the result obtained by the above method is  https://img-blog.csdnimg.cn/20200511222236394.jpg?x-oss-process=image/watermark,type_ZmFuZ3poZW5naGVpdGk,shadow_10,text_aHR0cHM6Ly9ibG9nLmNzZG4ubmV0L3B3YW5nOTU=,size_16,color_FFFFFF,t_70#pic_center  It can be seen from the three simulation examples that there is no problem with a single target, but when the two targets are too close, the DOA algorithm cannot distinguish between the two targets. This brings certain problems to our experiment:  1. The first is the actual effect of this algorithm. We can see that as the target approaches in the experiment, the effect of the DOA algorithm is relatively poor, and we cannot effectively distinguish the target. This requires us to introduce an effective distance threshold for the algorithm. When the distance is less than this threshold, we cannot use this algorithm to distinguish.  2. Combining the conclusions we got in the previous experiments and the knowledge that Mr. Wu told us in class, we can know that for an algorithm, there is always an extra cost. The additional cost of the algorithm is an important constraint that Mr. Wu repeatedly emphasizes throughout the communication principles and the entire content of the wireless communication course. This brings us to the question that needs to be considered in our experiments: Is there a higher resolution algorithm? And is there any additional overhead proposed by Mr. Wu for this algorithm?  **Traditional: MVDR(Minimum Variance Distortionless Response) Method etc**  First introducing weight vector , this vector helps us coordinate a specific direction to receive signals. It also make a great contribution in constraining the variance.  图示  描述已自动生成  The beam formed signal can be written as:  From the formula above we can calculate the beam formed signal power  If we take out the original signal :  Obviously, we want to minimize the noise and makes the signal go through the gateway completely, so we have our mathematic expression:  **MVDR beamforming calculation steps**  Step1: Estimate the autocorrelation matrix R from the received snapshot signal x (n );  Step2: Calculate the inverse matrix R^-1 of the autocorrelation matrix R;  Step3: According to the geometry of the array, construct the corresponding steering vector a(θ);  Step4: Make θ follow a certain step, scan at the angle you want to observe, and calculate Pθ successively;  Step5: Perform spectral peak search on Pθ to find the θ corresponding to the peak point;  **Conclusions and reflections**  1. The MVDR beamforming method can only process incoherent signals.  In solving the equation (8), the inverse operation of the autocorrelation matrix R is carried out. This requires R to be full rank, that is, the signals are irrelevant. If there is a coherent signal, then the above derivation cannot continue until equation (8). So, what if the signals are coherent?  2. MVDR beamforming is versatile, not limited to linear arrays.  It can be seen from the derivation throughout the text that there is no specific structure applied to a (θ ). For other forms of arrays, just modify the form of a (θ );  Use the MVDR beamforming method for DOA estimation without knowing the number of sources. MUSIC, ESPRIT algorithms, etc. all need to estimate the number of sources;  Using the MVDR beamforming method for DOA estimation, the resolution is much higher than that of the spatial FFT, which can be seen from the following simulation.  **Simulation results**  Suppose a uniform linear array has 16 elements, λ / 2 array; take 1024 snapshots to estimate the autocorrelation matrix R, two signals enter the large array from 10° and 20° directions respectively, and the signal-to-noise ratio is 10dB. Taking the signal coherent and incoherent conditions, using the MVDR beamforming method described in this article and spatial FFT and DOA estimation, the results are as follows.  5.1 DOA estimation with MVDR beamforming method  https://img-blog.csdnimg.cn/20200527130251494.jpg?x-oss-process=image/watermark,type_ZmFuZ3poZW5naGVpdGk,shadow_10,text_aHR0cHM6Ly9ibG9nLmNzZG4ubmV0L3B3YW5nOTU=,size_16,color_FFFFFF,t_70#pic_centerhttps://img-blog.csdnimg.cn/20200527130334772.jpg?x-oss-process=image/watermark,type_ZmFuZ3poZW5naGVpdGk,shadow_10,text_aHR0cHM6Ly9ibG9nLmNzZG4ubmV0L3B3YW5nOTU=,size_16,color_FFFFFF,t_70#pic_center  It can be seen from the simulation results that when the signal is incoherent, this method has a higher resolution; but when the signal is coherent, although there are still two peaks in the 10° and 20° directions, the corresponding ordinate is smaller. , And there are peaks in other places, which brings difficulty to the subsequent detection algorithm.  As a comparison, the results of the spatial FFT are also placed here. It can be seen that the resolution of the MVDR beamforming method is much higher.  https://img-blog.csdnimg.cn/20200527130752526.jpg?x-oss-process=image/watermark,type_ZmFuZ3poZW5naGVpdGk,shadow_10,text_aHR0cHM6Ly9ibG9nLmNzZG4ubmV0L3B3YW5nOTU=,size_16,color_FFFFFF,t_70#pic_center  **Conventional Subspace-Based: MUSIC, ESPRIT**  **MUSIC(Multiple Signal Classification)**  The MUSIC algorithm is also called the decomposition subspace algorithm. The MUSIC algorithm has good angle measurement performance when performing DOA estimation on non-coherent signal sources. Since the MUSIC algorithm breaks through the performance bottleneck of the linear prediction algorithm, it can distinguish multiple target signal sources existing in a beam.  The mathematical model of the target signal source is:  Assuming that the noise is spatially ideal white noise and the noise power is , the received data covariance matrix of the antenna array can be obtained from above:  Eigenvalue decomposition of **:**  Where is a subspace formed by eigenvector corresponding to large eigenvalues, which also becomes a signal subspace, and is a subspace formed by eigenvector corresponding to small eigenvalues, and also becomes a noise subspace. Under ideal conditions, the steering vector in the signal subspace is orthogonal to the noise subspace:  Considering that the actual received data matrix is limited, the maximum likelihood estimate of the covariance matrix is:  The MUSIC algorithm is implemented with minimum optimized search:  The spatial spectral of MUSIC algorithm is:  This is the matlab simulation process carried out in our root data  clc; clear all; close all;  %% -------------------------initialization-------------------------  f = 500; % frequency  c = 1500; % speed sound  lambda = c/f; % wavelength  d = lambda/2; % array element spacing  M = 10; % number of array elements  N = 100; % number of snapshot  K = 6; % number of sources  doa\_phi = [-30, 0, 20, 40, 60, 75]; % direction of arrivals  %% generate signal  dd = (0:M-1)'\*d; % distance between array elements and reference element  A = exp(-1i\*2\*pi\*dd\*sind(doa\_phi)/lambda); % manifold array, M\*K  S = sqrt(2)\(randn(K,N)+1i\*randn(K,N)); % array of random signal, K\*N  X = A\*S; % received data without noise, M\*N  X = awgn(X,10,'measured'); % received data with SNR 10dB  %% calculate the covariance matrix of received data and do eigenvalue decomposition  Rxx = X\*X'/N; % covariance matrix  [U,V] = eig(Rxx); % eigenvalue decomposition  V = diag(V); % vectorize eigenvalue matrix  [V,idx] = sort(V,'descend'); % sort the eigenvalues in descending order  U = U(:,idx); % reset the eigenvector  P = sum(V); % power of received data  P\_cum = cumsum(V); % cumsum of V  %% define the noise space  J = find(P\_cum/P>=0.95); % or the coefficient is 0.9  J = J(1); % number of principal component  Un = U(:,J+1:end);  %% music for doa; seek the peek  theta = -90:0.1:90; % steer theta  doa\_a = exp(-1i\*2\*pi\*dd\*sind(theta)/lambda); % manifold array for seeking peak  music = abs(diag(1./(doa\_a'\*(Un\*Un')\*doa\_a))); % the result of each theta  music = 10\*log10(music/max(music)); % normalize the result and convert it to dB  %% plot  figure;  plot(theta, music, 'linewidth', 2);  title('Music Algorithm For Doa', 'fontsize', 16);  xlabel('Theta(°)', 'fontsize', 16);  ylabel('Spatial Spectrum(dB)', 'fontsize', 16);  grid on;    It can be seen that when the incident signals are not correlated with each other, the traditional MUSIC algorithm can detect the approximate direction of arrival of six sources with high resolution, which are -29.7°, 0°, 19.8°, 39.8°, 60.4°, 74.7° , But there is still the problem of estimation accuracy, and there are many improved MUSIC algorithms that can be improved.  It should be noted that the degree of freedom of a half-wavelength uniform linear array with the number of elements M is M-1, which means that the maximum number of sources that can be resolved by the linear array is M-1. At the same time, if there is a coherent source, the effect of the MUSIC algorithm will be unsatisfactory  **Spatial smoothing MUSIC algorithm**  According to the information we consulted, we found that when multiple incident signals are coherent, the traditional MUSIC algorithm is not ideal. This is because when the multiple incident signals we use are coherent, part of the energy will be dissipated into the noise subspace, making the MUSIC algorithm unable to effectively estimate it.  In order to solve this situation, we found out the relevant methods through research and investigation. We have mainly learned by looking up information  Decoherence through dimensionality reduction processing is called dimensionality reduction processing because this method splits the original array into many sub-arrays, and reconstructs the received data covariance matrix through the covariance matrix of the sub-arrays. The DOF of the array will vary with If it is reduced, the number of coherent signals that can be resolved is reduced.  Let's first look at the effect of traditional MUSIC algorithm for DOA estimation of coherent signals.  This is the matlab simulation process carried out in our root data  clc; clear all; close all;  %% -------------------------initialization-------------------------  f = 500; % frequency  c = 1500; % speed sound  lambda = c/f; % wavelength  d = lambda/2; % array element spacing  M = 20; % number of array elements  N = 100; % number of snapshot  K = 6; % number of sources  coef = [1; exp(1i\*pi/6);...  exp(1i\*pi/3); exp(1i\*pi/2);...  exp(2i\*pi/3); exp(1i\*2\*pi)]; % coherence coefficient, K\*1  doa\_phi = [-30, 0, 20, 40, 60, 75]; % direction of arrivals  %% generate signal  dd = (0:M-1)'\*d; % distance between array elements and reference element  A = exp(-1i\*2\*pi\*dd\*sind(doa\_phi)/lambda); % manifold array, M\*K  S = sqrt(2)\(randn(1,N)+1i\*randn(1,N)); % vector of random signal, 1\*N  X = A\*(coef\*S); % received data without noise, M\*N  X = awgn(X,10,'measured'); % received data with SNR 10dB  %% calculate the covariance matrix of received data and do eigenvalue decomposition  Rxx = X\*X'/N; % covariance matrix  [U,V] = eig(Rxx); % eigenvalue decomposition  V = diag(V); % vectorize eigenvalue matrix  [V,idx] = sort(V,'descend'); % sort the eigenvalues in descending order  U = U(:,idx); % reset the eigenvector  P = sum(V); % power of received data  P\_cum = cumsum(V); % cumsum of V  %% define the noise space  J = find(P\_cum/P>=0.95); % or the coefficient is 0.9  J = J(1); % number of principal component  Un = U(:,J+1:end);  %% music for doa; seek the peek  theta = -90:0.1:90; % steer theta  doa\_a = exp(-1i\*2\*pi\*dd\*sind(theta)/lambda); % manifold array for seeking peak  music = abs(diag(1./(doa\_a'\*(Un\*Un')\*doa\_a))); % the result of each theta  music = 10\*log10(music/max(music)); % normalize the result and convert it to dB  %% plot  figure;  plot(theta, music, 'linewidth', 2);  title('Music Algorithm For Doa', 'fontsize', 16);  xlabel('Theta(°)', 'fontsize', 16);  ylabel('Spatial Spectrum(dB)', 'fontsize', 16);  grid on;   This is the result of our algorithm simulation. It can be seen that for coherent signals, the traditional MUSIC algorithm DOA estimation effect is very poor. **Spatial smoothing algorithm**  The dimensionality reduction processing and decoherence methods mainly include spatial smoothing processing algorithms, and the spatial smoothing processing algorithms can be divided into forward spatial smoothing algorithm (FSS), backward smoothing algorithm (BSS), forward and backward smoothing algorithm (FBSS), as described above Said that the estimation effect of these algorithms is very good, but the aperture of the array is lost, resulting in a decrease in the number of resolvable coherent signals.  **Linear array signal model**  https://img-blog.csdnimg.cn/20201028152823249.png?x-oss-process=image/watermark,type_ZmFuZ3poZW5naGVpdGk,shadow_10,text_aHR0cHM6Ly9ibG9nLmNzZG4ubmV0L3FxXzM2NTgzMzcz,size_16,color_FFFFFF,t_70#pic_center  **Forward spatial smoothing algorithm**  The forward spatial smoothing algorithm divides the array into multiple overlapping sub-arrays, and then averages the covariance matrix of the data received by the sub-arrays. When the number of sub-array elements is greater than or equal to the number of coherent signals, the coherence can be effectively decohered.https://img-blog.csdnimg.cn/20201030120335719.png?#pic_center  As shown in the figure above, we evenly divide the M-element array into L sub-arrays, and each sub-array has N=M-L+1 array elements. Taking the leftmost sub-array as the reference array, define the received data of the J-th sub-array as:    Then the covariance matrix (also called the spatial smoothing matrix) of the received data of the J-th subarray can be expressed as    among them,    A1 is the flow matrix of the first sub-array, that is, the reference array.  Therefore, the covariance matrix after forward space smoothing can be obtained by averaging the covariance matrix of each sub-matrix.    Using forward spatial smoothing covariance matrix and MUSIC algorithm, the orientation of multiple coherent signals can be distinguished. It can be proved that this method can detect up to M/2 coherent signals.  This is the matlab simulation process carried out in our root data  clc; clear all; close all;  %% -------------------------initialization-------------------------  f = 500; % frequency  c = 1500; % speed sound  lambda = c/f; % wavelength  d = lambda/2; % array element spacing  M = 20; % number of array elements  N = 100; % number of snapshot  K = 6; % number of sources  L = 10; % number of subarray  L\_N = M-L+1; % number of array elements in each subarray  coef = [1; exp(1i\*pi/6);...  exp(1i\*pi/3); exp(1i\*pi/2);...  exp(2i\*pi/3); exp(1i\*2\*pi)]; % coherence coefficient, K\*1  doa\_phi = [-30, 0, 20, 40, 60, 75]; % direction of arrivals  %% generate signal  dd = (0:M-1)'\*d; % distance between array elements and reference element  A = exp(-1i\*2\*pi\*dd\*sind(doa\_phi)/lambda); % manifold array, M\*K  S = sqrt(2)\(randn(1,N)+1i\*randn(1,N)); % vector of random signal, 1\*N  X = A\*(coef\*S); % received data without noise, M\*N  X = awgn(X,10,'measured'); % received data with SNR 10dB  %% reconstruct convariance matrix  %% calculate the covariance matrix of received data and do eigenvalue decomposition  Rxx = X\*X'/N; % origin covariance matrix  Rf = zeros(L\_N, L\_N); % reconstructed covariance matrix  for i = 1:L  Rf = Rf+Rxx(i:i+L\_N-1,i:i+L\_N-1);  end  Rf = Rf/L;  [U,V] = eig(Rf); % eigenvalue decomposition  V = diag(V); % vectorize eigenvalue matrix  [V,idx] = sort(V,'descend'); % sort the eigenvalues in descending order  U = U(:,idx); % reset the eigenvector  P = sum(V); % power of received data  P\_cum = cumsum(V); % cumsum of V  %% define the noise space  J = find(P\_cum/P>=0.95); % or the coefficient is 0.9  J = J(1); % number of principal component  Un = U(:,J+1:end);  %% music for doa; seek the peek  dd1 = (0:L\_N-1)'\*d;  theta = -90:0.1:90; % steer theta  doa\_a = exp(-1i\*2\*pi\*dd1\*sind(theta)/lambda); % manifold array for seeking peak  music = abs(diag(1./(doa\_a'\*(Un\*Un')\*doa\_a))); % the result of each theta  music = 10\*log10(music/max(music)); % normalize the result and convert it to dB  %% plot  figure;  plot(theta, music, 'linewidth', 2);  title('Music Algorithm For Doa', 'fontsize', 16);  xlabel('Theta(°)', 'fontsize', 16);  ylabel('Spatial Spectrum(dB)', 'fontsize', 16);  grid on;   It can be seen that when the 6 incident signals are uniformly coherent, the MUSIC algorithm based on forward smoothing can better estimate the DOA, but there are still estimation accuracy problems, such as the signal with a true incident angle of 75° The bearing is estimated to be 74.2°. **Backward spatial smoothing algorithm**  https://img-blog.csdnimg.cn/20201030120657822.png?#pic_center  Backward spatial smoothing is more accurately conjugate backward spatial smoothing, which is to smooth the covariance matrix of the conjugate received data of the backward sub-array. Define the first conjugate backward subarray {M,M−1,...,M−p+1} to be composed, and the second subarray to be composed of {M−1,M−2,...,M−p}, in turn The number of sub-arrays is L=M−p+1.  It is easy to know the relationship between the conjugate backward spatial smoothing covariance matrix and the forward spatial smoothing covariance matrix :    Using backward spatial smoothing covariance matrix and MUSIC algorithm can also distinguish the orientation of multiple coherent signals. It can be proved that the method can detect M/2 coherent signals at most.  This is the matlab simulation process carried out in our root data  clc; clear all; close all;  %% -------------------------initialization-------------------------  f = 500; % frequency  c = 1500; % speed sound  lambda = c/f; % wavelength  d = lambda/2; % array element spacing  M = 20; % number of array elements  N = 100; % number of snapshot  K = 6; % number of sources  L = 10; % number of subarray  L\_N = M-L+1; % number of array elements in each subarray  coef = [1; exp(1i\*pi/6);...  exp(1i\*pi/3); exp(1i\*pi/2);...  exp(2i\*pi/3); exp(1i\*2\*pi)]; % coherence coefficient, K\*1  doa\_phi = [-30, 0, 20, 40, 60, 75]; % direction of arrivals  %% generate signal  dd = (0:M-1)'\*d; % distance between array elements and reference element  A = exp(-1i\*2\*pi\*dd\*sind(doa\_phi)/lambda); % manifold array, M\*K  S = sqrt(2)\(randn(1,N)+1i\*randn(1,N)); % vector of random signal, 1\*N  X = A\*(coef\*S); % received data without noise, M\*N  X = awgn(X,10,'measured'); % received data with SNR 10dB  %% reconstruct convariance matrix  %% calculate the covariance matrix of received data and do eigenvalue decomposition  Rxx = X\*X'/N; % origin covariance matrix  H = fliplr(eye(M)); % transpose matrix  Rxxb = H\*(conj(Rxx))\*H;  Rf = zeros(L\_N, L\_N); % reconstructed covariance matrix  for i = 1:L  Rf = Rf+Rxxb(i:i+L\_N-1,i:i+L\_N-1);  end  Rf = Rf/L;  [U,V] = eig(Rf); % eigenvalue decomposition  V = diag(V); % vectorize eigenvalue matrix  [V,idx] = sort(V,'descend'); % sort the eigenvalues in descending order  U = U(:,idx); % reset the eigenvector  P = sum(V); % power of received data  P\_cum = cumsum(V); % cumsum of V  %% define the noise space  J = find(P\_cum/P>=0.95); % or the coefficient is 0.9  J = J(1); % number of principal component  Un = U(:,J+1:end);  %% music for doa; seek the peek  dd1 = (0:L\_N-1)'\*d;  theta = -90:0.1:90; % steer theta  doa\_a = exp(-1i\*2\*pi\*dd1\*sind(theta)/lambda); % manifold array for seeking peak  music = abs(diag(1./(doa\_a'\*(Un\*Un')\*doa\_a))); % the result of each theta  music = 10\*log10(music/max(music)); % normalize the result and convert it to dB  %% plot  figure;  plot(theta, music, 'linewidth', 2);  title('Music Algorithm For Doa', 'fontsize', 16);  xlabel('Theta(°)', 'fontsize', 16);  ylabel('Spatial Spectrum(dB)', 'fontsize', 16);  grid on;    It can be seen that when the six incident signals are uniformly coherent, the MUSIC algorithm based on backward spatial smoothing can better estimate its DOA, and the estimation accuracy is higher.  **Forward/backward spatial smoothing algorithm**  The forward and conjugate backward spatial smoothing covariance matrix are defined as the average of the forward spatial smoothing covariance matrix and the conjugate backward spatial smoothing covariance matrix, namely:    So as long as the number of spatial smoothing is greater than or equal to the number of coherent signal sources, the forward and conjugate backward spatial smoothing covariance matrices are generally full-rank. The maximum number of coherent signal sources that can be detected using the forward/backward spatial smoothing method is 2M/3. You may be curious how this maximum number of coherent signal source detections is obtained?  Assuming: The number of array elements of the array antenna is M, and the number of forward/backward spatial smoothing is L times respectively. Then the number of elements of each subarray is N=M−L+1. At the same time, it can be known that the maximum resolution is The number of signals is M−L, that is, the number of elements of the subarray minus 1; the number of signals that can be resolved by smoothing N times in the forward and backward directions is 2L. In the maximum case, the two are equal, so M−L= 2L, that is, L=M/3; Therefore, 2L=2M/3, so the maximum number of signals that can be resolved in the forward/backward spatial smoothing is 2M/3. Therefore, the forward/backward spatial smoothing improvement technology can greatly increase the array aperture.  This is the matlab simulation process carried out in our root data  clc; clear all; close all;  %% -------------------------initialization-------------------------  f = 500; % frequency  c = 1500; % speed sound  lambda = c/f; % wavelength  d = lambda/2; % array element spacing  M = 20; % number of array elements  N = 100; % number of snapshot  K = 6; % number of sources  L = 10; % number of subarray  L\_N = M-L+1; % number of array elements in each subarray  coef = [1; exp(1i\*pi/6);...  exp(1i\*pi/3); exp(1i\*pi/2);...  exp(2i\*pi/3); exp(1i\*2\*pi)]; % coherence coefficient, K\*1  doa\_phi = [-30, 0, 20, 40, 60, 75]; % direction of arrivals  %% generate signal  dd = (0:M-1)'\*d; % distance between array elements and reference element  A = exp(-1i\*2\*pi\*dd\*sind(doa\_phi)/lambda); % manifold array, M\*K  S = sqrt(2)\(randn(1,N)+1i\*randn(1,N)); % vector of random signal, 1\*N  X = A\*(coef\*S); % received data without noise, M\*N  X = awgn(X,10,'measured'); % received data with SNR 10dB  %% reconstruct convariance matrix  %% calculate the covariance matrix of received data and do eigenvalue decomposition  Rxx = X\*X'/N; % origin covariance matrix  H = fliplr(eye(M)); % transpose matrix  Rxxb = H\*(conj(Rxx))\*H;  Rxxfb = (Rxx+Rxxb)/2;  Rf = zeros(L\_N, L\_N); % reconstructed covariance matrix  for i = 1:L  Rf = Rf+Rxxfb(i:i+L\_N-1,i:i+L\_N-1);  end  Rf = Rf/L;  [U,V] = eig(Rf); % eigenvalue decomposition  V = diag(V); % vectorize eigenvalue matrix  [V,idx] = sort(V,'descend'); % sort the eigenvalues in descending order  U = U(:,idx); % reset the eigenvector  P = sum(V); % power of received data  P\_cum = cumsum(V); % cumsum of V  %% define the noise space  J = find(P\_cum/P>=0.95); % or the coefficient is 0.9  J = J(1); % number of principal component  Un = U(:,J+1:end);  %% music for doa; seek the peek  dd1 = (0:L\_N-1)'\*d;  theta = -90:0.1:90; % steer theta  doa\_a = exp(-1i\*2\*pi\*dd1\*sind(theta)/lambda); % manifold array for seeking peak  music = abs(diag(1./(doa\_a'\*(Un\*Un')\*doa\_a))); % the result of each theta  music = 10\*log10(music/max(music)); % normalize the result and convert it to dB  %% plot  figure;  plot(theta, music, 'linewidth', 2);  title('Music Algorithm For Doa', 'fontsize', 16);  xlabel('Theta(°)', 'fontsize', 16);  ylabel('Spatial Spectrum(dB)', 'fontsize', 16);  grid on;    Because the improved technology of forward/backward spatial smoothing greatly increases the array aperture, it can be seen from the above DOA results that the resolution has been improved.  **ESPRIT(Estimating Signal Parameters Via Rotational Invariance Techniques)**  The received signal is subjected to spatial Fourier transform (the difference between spatial Fourier transform and discrete-time Fourier transform is that the sum of the spatial Fourier transform is the space position m of the array element, while the time-domain Fourier transform is calculated The sum variable is discrete time n), and then the square of the modulus is taken to obtain the spatial spectrum, and the arrival direction of the signal is estimated (the phase φ corresponding to the maximum value of the spatial spectrum, and then according to the definition φ=2πdsinθ/λ, calculate θ).  Step 1  Calculate autocorrelation , apply eigenvalues decomposition to obtain eigenvectors  ***[V,D] = eig(A) produces a diagonal matrix D of eigenvalues and a full matrix V whose columns are the corresponding eigenvectors so that A\*V = V\*D.***  Step 2  Construct matrix and , they are the first M-1 columns and last M-1 columns of respectively.  Step 3  Calculate the eigenvalues of  Step 4  Calculate the  ***angle(H) returns the phase angles, in radians, of a matrix with complex elements.***  **Three algorithms compare the simulation process**  This is the matlab simulation process carried out in our root data  clc,clear all,close all  %% 产生信号样本  N=100;M=10;%信号样本数目和阵元个数  K=2;%信源个数  theta=[-10;40]\*pi/180;  SNR=[10;20];sigma=1;  Am=sqrt(2\*sigma^2\*10.^(SNR/10));  % Am=[sqrt(10.^(SNR/10))];  S=Am\*ones(1,N);  S(2,:)=S(2,:).\*exp(1i\*2\*pi\*rand(1,N));  for a=1:M  for b=1:K  A(a,b)=exp(-1i\*(a-1)\*pi\*sin(theta(b)));%第 b 列对应的都是 theta(b)  end  end  V=zeros(M,N);  for m=1:M  v=wgn(1,N,0,'complex');  v=v-mean(v);  v=v/std(v);  V(m,:)=v;  end  X=A\*S+V;  %% 利用接受数据估计信号的空间相关矩阵 R  R=zeros(M,M);  for i=1:N  R=R+X(:,i)\*X(:,i)';  end  R=R/N;%是一个统计平均  %MUSIC 算法  [VR,D]=eig(R);  D=real(D);  [B,IX]=sort(diag(D));  G=VR(:,IX(M-K:-1:1));  MUSICP=[];  for n=-pi/2:pi/180:pi/2  a=exp(-1i\*[0:M-1]'\*pi\*sin(n));  MUSICP=[MUSICP,1/(a'\*G\*G'\*a)];  MUSICP=real(MUSICP);end  n=length(MUSICP);  maxx=max(MUSICP);  figure,plot(-90:1:90,10\*log10((MUSICP+eps)/maxx)+3.5),axis([-90,90,-  60,inf]),title('MUSIC 算法')  %RootMUSIC 算法  syms z  pz=z.^([0:M-1]');  pz1=(z^(-1)).^([0:M-1]);  fz=z^(M-1)\*pz1\*G\*G'\*pz;  a=sym2poly(fz);  r=roots(a);  r1=abs(r);  for i=1:2\*K %每个信号源有 K 个  [Y,I(i)]=min(abs(r1-1));  r1(I(i))=inf;  end  for i=1:2\*K  theta\_esti(i)=asin(-angle(r(I(i)))/pi)\*180/pi;  end  %ESPRIT 算法  S=VR(:,IX(M:-1:M-K+1));  S1=S(1:M-1,:);  S2=S(2:M,:);  fai=S1\S2;  [U\_fai,V\_fai]=eig(fai);  for i=1:K  ESPRITtheta\_esti(i)=asin(-angle(V\_fai(i,i))/pi)\*180/pi;  end  %MVDR 算法  MVDRP=[];  for n=-pi/2:pi/180:pi/2  a=exp(-1i\*[0:M-1]'\*pi\*sin(n));  MVDRP=[MVDRP,1/(a'\*inv(R)\*a)];  end  n=length(MVDRP);  maxx=max(MVDRP);  figure,plot(-90:1:90,10\*log10((MVDRP+eps)/maxx)+3.5),axis([-90,90,-  35,inf]),title('MVDR')  %F-SAPES 算法  P=6;%子阵数目L=M+1-P;%子阵阵元数目，书上是 M-1  Rf=zeros(L,L);  for i=1:P  Rf=Rf+X(i:i+L-1)\*X(i:i+L-1)'/N;  end  Rf=Rf/P; %子阵平滑后的空间相关矩阵  n1=0:P-1;  n2=0:L-1;  cc=[1 zeros(1,L-1)];  for n3=-90:.5:90  fy=exp(1i\*pi\*sin(n3/180\*pi));  tt=[(fy.^(n1')).' zeros(1,M-P)];  Tfy=toeplitz(cc,tt);  GfTheta=1./(P^2)\*Tfy\*R\*Tfy';  Qf=Rf-GfTheta;  aTheta=fy.^(-n2');  Wof=(Qf\aTheta)./(aTheta'\*(Qf\aTheta));  sigma2sTheta(((n3+90)/.5+1))=Wof'\*GfTheta\*Wof;  end  maxx=max(sigma2sTheta);  figure,plot(-90:.5:90,10\*log10((sigma2sTheta+eps)/maxx)+3.5),axis([-90,90,-  35,inf]),title('F-SAPES')      The three pictures from top to bottom are simulation images of the MUSIC algorithm, MVDR algorithm, and F-SAPES algorithm. Because it is a preliminary exploration of the algorithm, we have a certain understanding of the principles and operation process of the three algorithms, but there is no complete system for the analysis process of the effect of the three algorithms. We have simulated the results of the three algorithms. With a certain understanding, a certain analysis was carried out. However, our overall grasp of the three algorithms is not yet in place, there are still certain deficiencies in the construction of the knowledge system, and there may still be certain imperfections in the principle analysis. Therefore, our analysis of the three algorithms will not be presented in the report. We will focus on this aspect and comprehensively improve it in subsequent experiments and reports.  **experiment procedure**  Our experiment is mainly divided into two parts, one part is field experiment, and the other part is data processing experiment. During the field experiment, we used the formula principle in the SDR official website to conduct experiments to determine the position of the antenna.  We all know that the antenna, which is an important part of the receiver, plays an important role in the signal reception and demodulation process. We must consider the distance between the antennas in the experiment. This is because the antenna must be within the distance designed by the hardware system to receive the transmitted signal normally, and then the signal position can be positioned normally.  Let us first describe in detail the two argumentation processes that we explained in the introduction part. The first is the problem of the signal source. In the experiment, Professor Wu gave us a signal source that can move the position, and we can detect the position of the signal source in this way. At the same time, we can also select a fixed signal source in the school, that is, a signal base station for detection. These are the two technical routes we can choose during the detection process.  Before the formal test, we conducted a field inspection of the experimental site. We finally chose Songhe Stadium as the experimental site. This is because Songhe Stadium is relatively open and is the flattest and least sheltered field in the school. Under this premise, we can ensure that the interference obtained in the experiment is minimal. Most of the base stations in our school are built on the mountains. Before the experiment, we first conducted research on the base stations in the school. The reason why we finally chose Songhe Stadium was also because the distance between Songhe Stadium and the base station was relatively moderate.  In the course of wireless communication, we learned the interference process of multipath effect on signal transmission, and also tried to calculate the influence of multipath effect according to the formula, but the whole calculation process is always based on theory. If we want to better understand the impact of multipath effects, we must do it through specific experiments. Fortunately, in this experiment, Mr. Wu provided us with specific experimental equipment to conduct physical experiments.  Let's talk about our first demonstration process. The signal source is very important in our experiment. The main process of our experiment is to detect the signal source by writing an algorithm. Therefore, we take the process of demonstrating the signal source as the first step of our specific experiment. This is also a very critical step.  **Calculation of antenna spacing**  We can find the calculation process of antenna spacing by consulting the official website information and asking students for advice. In the experiment, we explored two signal detection methods: circular antenna array and linear antenna array.    Figure Schematic diagram of linear antenna array antenna spacing  In the experiment, we can confirm that the frequency of the detection signal is f=850MHz and s=1/3 by consulting the data.  So we can get the following formula:  According to formula (33), we can get the calculation result of the distance between the linear array antennas. We follow this calculation result in the experiment.    Figure The distance calculation method of circular array antenna  For the circular antenna array structure, the same is true for f=850MHz, s’=1/3:  So we can get the following formula:  According to the above formula, we can perform calculations to get the arrangement of the antenna.  **Pre- experiment**  Before the formal experiment, we first conducted a preliminary experiment. Through the pre-experiment process, we can make a certain plan for the direction of the steps in the experiment. In the pre-experiment preparation process, we have made some predictions about the difficulties and problems that may be encountered in the experiment, and made targeted preparations for the problems. Before the experiment, we expected that the wind at the experimental site might be relatively high. The antenna we used is mobile. If there is wind disturbance, the antenna may fall down. In the theoretical preparation stage of the experiment, we found that in order to make the SDR work properly, we must determine the antenna spacing. Therefore, we need to prepare cardboard and sponge glue for fixing before the experiment.  D:\南科大文件\MobileFile\IMG_20210531_143854.jpgD:\南科大文件\MobileFile\IMG_20210523_135020.jpg  Fig. Foam glue and fixed cardboard used in the experiment.  In the pre-experiment process, we carried out an all-round detection process for the test site, and determined several sites that can be tested. Since the mobile signal source must require the input of electrical energy, we need to find a location with a power interface to conduct experiments. This requires us to conduct experiments indoors. In the process of indoor experiment, because there are more indoor objects and the structure is more complicated, multipath effect will be generated. In the preliminary experiment, we did not make a very detailed record, so there is no picture of us searching for mobile signal sources.  In the process of searching for the mobile signal source, the actual experimental effect is not very good. After completing the experiment, we conclude and analyze that this is because the signal strength of the mobile signal source is not high on the one hand, and on the other hand, we will use the signal source The distance between the antenna and the antenna is slightly farther. This is because it is difficult for us to detect the accurate signal position if the distance is too close. As the distance becomes longer, the signal transmission process has also changed, and the signal transmission is more affected. At the same time, outdoor signal sources will also cause certain interference to indoor mobile signal sources. Therefore, the experimental effect of the indoor signal source detection process is not very ideal.  Therefore, the result of our first demonstration is that we should conduct outdoor experiments to detect fixed signal sources outdoors. This is because we believe that the outdoor signal source is more stable on the one hand and the detection effect is better on the other hand. The subsequent experimental process also proved our guess.  Professor Wang also described a similar measurement experiment process in our theoretical course after the experiment. Professor Wang’s experimental model took the Runyang Stadium platform as the experimental location and measured the signal base station facing the Songhe Stadium. In addition to the main peak of the signal source, Professor Wang’s experimental results also have two peaks caused by reflections, which also proves that we chose the experimental location more correctly.  **Formal experiment process**  After completing the pre-experiment, we discussed and gained a further understanding of the details and process planning in the experiment. After that, we officially started the formal test part.  In the formal experiment, the first relatively big problem we faced was the problem of site selection.  In terms of site selection, first of all, we have selected Songhe Stadium as our experimental site in terms of address selection, but in specific experiments we need to actually demonstrate the location of the antenna. This is our second demonstration process. Before the start of the experiment, we thought that a more suitable site was the geometric center of the stadium football field. We pointed out in the previous report that we need to find a relatively open area to carry out the experiment, so as to ensure the least interference in the experiment. In fact, the most open area of ​​the entire Songhe Stadium must be the geometric center of the football field. Therefore, in the experiment, we first choose the geometric center of the football field to carry out the experiment.  In the figure below, we placed the site of our first experiment. We can see that the experimental location is very open, with few obstructions, and the distance to the tree-like signal tower is relatively moderate, so we theoretically choose this location. It is possible to get better experimental results.  D:\南科大文件\MobileFile\IMG_20210525_165416.jpg  Figure The location where we performed the experiment is located at the geometric center of the football field  D:\南科大文件\MobileFile\IMG_20210525_165413.jpg  Figure We can see that the experiment location is very open around    Figure The image we draw based on the data obtained in the experiment  As shown in the figure above, we can roughly see that there are three peaks, but in fact, the effect of this experiment is not very satisfactory. We will analyze it in detail in the following report.  But in fact, the results obtained in our experiments are not very satisfactory. Since we did not have standard experimental results as a control during the experiment, we did not have a proportional control after the experiment was done at that time, and we did not find an experiment in this position. The result is not ideal. We ended the experiment after we finished collecting the data. The photos of the experimental location we took in the report were taken afterwards, so there was no antenna and experimental equipment on the cardboard. This is also a regret in our experiment. During the experiment, we paid one-sided attention to the record of the experimental results, and did not record the experimental process in detail. The pictures in the report are the photos we took by rechecking the scene before the report, so it may be different from our actual experimental site. There is a certain difference. We must remember this lesson in the subsequent experiments and pay attention to the record of the experimental process.  **Double verification experiment**  After completing the first experiment, we received the data recorded by Professor Wang. This is an experiment conducted by Professor Wang's research group. The experimental process is that Professor Wang uses Range-Doppler Calculation to detect the signal source. Let's first briefly introduce the process of the experiment:    Figure Schematic diagram of Professor Wang's experimental geography  During the experiment, Professor Wang chose the platform on the second floor of Runyang Gymnasium as the measurement point. We can get a general understanding of the experimental site from the satellite map. We can see that the position of the receiver in the experiment is facing the approximate position of the base station. The arrows in the schematic diagram indicate the main reflection buildings in the area of the base station and the receiver. In the experiment, the expert apartment and the student apartment have two main signal reflection areas, which also explains why the image obtained in our experiment has three peaks instead of one.    Figure Experimental data recording equipment  In the experiment, Professor Wang detected the signal source of the base station. This is the device used by Professor Wang to detect the signal. We can see that Professor Wang used a device with four antennas for detection.    Figure The weather and surrounding environment of Professor Wang's experiment  The temperature during our experiment was relatively high and it was sunny, while the time of Professor Wang's experiment was cloudy, and the humidity and temperature of the air were also somewhat different from when we experimented. This reflects the problem of control variables. In theory, we need to strictly control the variables in the experiment, and we also need to consider weather and environmental factors more in the process of signal source detection experiments. This is because the transmission of electromagnetic waves is related to the temperature and humidity of the environment, so there should be a slight gap between the transmission results obtained in our experiment and the results of Professor Wang's experiment. If we want to simulate the effect of the actual experiment as much as possible, we need to fully consider the environmental issues during the experiment. That is to say, if we want to study the detection process of the signal source more comprehensively, we need to conduct multiple experiments in different environments. In our experiments, due to the limitations of equipment and time, we have no way to conduct a full range of experiments, but in the subsequent experiments, we need to consider the needs of the experiment as much as possible in all directions, and then complete the experiment.    Figure Range-Doppler Calculation experiment conducted by Professor Wang  Professor Wang conducted the Range-Doppler Calculation experiment. In the experiment, Professor Wang used eight linear array antennas to attach an iron plate to an experimenter. The experimenter with the iron plate attached to his body ran back and forth during the experiment to record the data. After completing the data recording, we can use the experimental data to complete the experimental process of Range-Doppler Calculation.  When we just came to the conclusion, we thought that the experimental location of Runyang Stadium was not the best, so the experimental results obtained may not be as good as those obtained in the center of the stadium. But in fact, we can see that the experimental data we got is not as good as the experimental data obtained by Professor Wang. We believe that on the one hand, our experimental equipment is not as good as that used by Professor Wang, and on the other hand, because the temperature of our experimental environment is relatively high. We chose the experiment time to start at 11 am. At this time, the radiation of the sun is more serious. We found in the later experiments that the instrument will generate some heat during use, and at the same time, the instrument will also generate some heat due to the light radiation. In the final experiment, the temperature was relatively high, so the performance of the instrument dropped drastically, so the effect of the experiment was not satisfactory. We also need to consider these issues in the subsequent experiments, that is, the temperature of the experimental instrument. In fact, it is not only the temperature, the experimental environment, the wind direction, and the level of the site need to be fully considered.  **MATLAB experimental part DOA estimate**  In the process of DOA estimation, we used the MUSIC algorithm to conduct experiments. In this experiment, our algorithm and programming part is mainly divided into two parts: self-developed part and open source part. The more important music algorithms have more mature open source algorithms on the website. We only need to make appropriate modifications to apply them to our experiments. The self-developed part of our algorithm is mainly focused on the data processing stage. In the experiment, we also have finished programs to complete the experiment, but we found that the data we got is generally of low quality, and the experimental conclusions may not be accurate enough. Therefore, we chose to use data for processing to complete the experiment.  During the operation of the MUSIC algorithm, we first calculate the autocorrelation matrix of the signal, which is , and then perform eigenvalue decomposition on the autocorrelation matrix. After the eigenvalue decomposition is completed, the algorithm will reorder the eigenvalues, that is, arrange the eigenvalues ​​in descending order. Through this arrangement process, we can get the corresponding eigenvalues ​​and eigenvectors. This is also the process of decomposing the matrix. Here we review the knowledge of linear algebra, which is also one of the gains from our experiment.  After we complete the eigenvalue rearrangement and get the corresponding eigenvalues ​​and eigenvectors, we need to set a threshold of 0-1 to limit the size of the noise space. This question seems very abstract, let's take an example to illustrate. For example, the threshold is 0.8. Before that, we define a variable P whose meaning is the sum of the total eigenvalues, that is, the trace of the matrix. We then define a value , which means an accumulated array of eigenvalues. When , the remaining eigenvalues ​​are the eigenvalues ​​corresponding to the noise space. This is a piece of core code in our experiment. The core code part rewrites the process we describe into an algorithm. We present the code below:  [U,V] = eig(Rf); % eigenvalue decomposition  V = diag(V); % vectorize eigenvalue matrix  [V,idx] = sort(V,'descend'); % sort the eigenvalues in descending order  U = U(:,idx); % reset the eigenvector  P = sum(V); % power of received data  P\_cum = cumsum(V); % cumsum of V    %% define the noise space  J = find(P\_cum/P>=noiseThreshold); % or the coefficient is 0.9  J = J(1); % number of principal component  Un = U(:,J+1:end);  After we have completed the above part, we need to bring the calculated noise space into the formula. We can see that when the corresponding theta value represents the direction of arrival, due to the existence of orthogonality, the denominator is small, resulting in a large value of P. When the P value is relatively large, a spike will occur. After a complete beam steering and a complete search from 0 to 180 degrees, the corresponding wave direction of arrival can be found.  图表, 雷达图  描述已自动生成  Figure Reference signal  图表, 雷达图  描述已自动生成  Figure Passive signal    Figure DOA estimation result obtained by MUSIC algorithm. The geometric layout of the reference signal is shown in the figure.  Above we show the image estimated by DOA, we can see that the image we recovered from the data is roughly the same as the result of geographic location analysis. We analyze that the peak generated by 300 degrees is the signal reflected by the expert apartment, while the peak generated by 30 degrees is the signal reflected by the freshman dormitory, and the signal reflected by the Huiyuan from 90 degrees to 60 degrees. We can know from the analysis of the image that the signal intensity reflected by the expert apartment is relatively large, the signal intensity reflected by the freshman dormitory is slightly smaller, and the signal intensity reflected by the Huiyuan is the smallest and occupies a relatively large range. The reason for this problem is that on the one hand, the distance and angle of the reflecting object are different, and on the other hand, it is because of the area and height of the building. Here we focus on the problem of the small reflection intensity of Huiyuan. We believe that the signal intensity reflected by Huiyuan is relatively small and the range is relatively large. The main reason is that the structure of the buildings in Huiyuan area is relatively complex, the area of ​​the building is relatively large, and the height is relatively low. There is no obvious reflection peak. On the other hand, the angle formed by the direction line between Huiyuan and the receiver is relatively large, so signal detection may not be effective. This also shows that we should pay attention to the angle when detecting the signal source.  We submitted the complete code along with the report.  **Range-Doppler estimate**  The core formula for calculating the distance Doppler map is to calculate the ambiguity value, which is given by the following formula：  The τ and in the formula are the variables we scan. By changing different values of τ and , the corresponding target can be found in the range Doppler diagram.  The core code of concrete realization is:  col = 1;  for fd = -rangeFd:2:rangeFd  row = 1;  for tau = 0:rangeTau      k = 1:shape(1,2);  k = exp(1i\*2\*pi\*fd\*k/25000000);    RD(row,col) = abs(sum(redirectedX(1,10-tau:10-tau+length(Y)-1).\*k(10:end).\*conj(redirectedY(1,:))));  row = row + 1;  bar = [row,col]  end    col = col + 1;    end    Figure passive signal's original range-doppler spectrum  The above figure shows the original range-doppler spectrum of the passive signal, without post-processing the obtained results. It can be seen that the vertical axis is distance and the horizontal axis is doppler. In the area of 20hz and 30m, there is an ambiguity peak. After the corresponding calculation, the speed is about 1.4m/s. There is a large peak in the area where doppler fre is 0 in the spectrum, because most objects in reality are still.   |  |  | | --- | --- | | Deg=-20 | Deg=30 | |  |  |   Table range-doppler spectrum with Deg=-20 and Deg=30   |  |  | | --- | --- | | Deg=29 | Deg=30 | |  |  | | Deg=31 | | |  | |   Table range-doppler spectrum with Deg=29 and Deg=30 and Deg=31    Figure The spectrogram we obtained based on the signal in the experiment    Figure Speed angle    Figure range angle  This is the image of the speed angle and the range angle. It is not very relevant to our experiment. We will not explain it here. We will show all the result images in the video.    Figure The technical route followed in the experiment  This is the technical route we followed in our experiment and the general flow chart we followed during the experiment.  **Experimental details**    Figure Receiver control interface  In the Recelver Configuration interface, we can control the details of the receiver by adjusting the center frequency, sampling frequency and gain of the four antennas. We can see that the gains of the four antennas must be consistent.  In the IQ Preprocessing interface, we can adjust the bandwidth of the filter. So as to further complete our experiment.    Figure Signal display interface  Let's analyze the signal display interface again, we can see that the Four-channel phase synchronization image is not as effective as in the paper. We analyze this is the reason why our experimental equipment is not as good as the paper and our experimental environment is not professional in the paper.  In fact, the image we have selected here is the image obtained by our best experimental results. Even in this case, our experimental effect is still far from the standard effect, but according to the experimental results, we can conclude that we get The result is correct. We can generally reproduce the results of the paper.    Figure Signal waveform display module control panel    Figure Signal waveform display module    Figure The image drawn by the music algorithm based on the collected data  We can see that the image effect we get is still relatively good. The image obtained by the system using the music algorithm has three peaks, the middle peak is the highest, which is obtained by our real signal source. The peaks on both sides correspond to the reflection of the building. We can see that overall the experimental data we got is also good, and can generally detect the location of the signal source.  **Site selection in the experiment**  After we completed the first experiment, we conducted a controlled experiment and found that our experimental results were not very good, so we went to the original site for the second experiment. When conducting the second experiment, we were under a lot of pressure. This is because we have Professor Wang’s data, and we have a relatively standard control group in our experiment. We need to benchmark this control group, and we can’t get it. Bad results.  On this basis, we have the second argumentation process, which is the site selection problem in the experiment.   |  | | --- | | D:\南科大文件\MobileFile\IMG_20210525_165115.jpg | |  |   This is the signal detection process carried out on the structure of the grandstand. Because the topography of the grandstand stairs is more complicated, we predicted that the results of the experiment may be poor before the experiment, but we still carried out a field experiment, this is because we Hope to accumulate more error samples. The process of accumulating error samples can help us better understand the causes of errors, and it is also a good exercise for our level of analysis. Since this location is located on a ladder, the overall structure is more complicated, and there is no open test site, so we spent a lot of time and energy when doing this experiment.   |  | | --- | | D:\南科大文件\MobileFile\IMG_20210525_165249.jpg | |  |   This is the result of our experiment in a relatively flat position on the stand. We can see that although the result is better than the result of the above experiment, the overall interference is still relatively large and the effect is not very good.   |  | | --- | | D:\南科大文件\MobileFile\IMG_20210525_164657.jpg | |  |   This is the signal detection process carried out on the platform of the gymnasium. Because there is a stepped structure in the back and a grid of iron material in the front, the transmission effect is not very ideal. But the effect of this position compared to the stand has been greatly improved. This is because there is a large interference structure on the connection between the stands and the signal source.   |  | | --- | | D:\南科大文件\MobileFile\IMG_20210525_164553.jpg | |  |   This is the result of the experiment we conducted on the stand of Runyang Stadium. We can see that the signal peak can be seen roughly, but other directions have different levels of signal source strength. We think it may be because there is a certain error in the data we recorded in this part. But we are not sure of this reason. This is because due to the limited time, we can not complete the experiment for each location, so the experimental samples obtained are not too enough, so we generally have not too much experience in discussing the experimental results. According to the experimental results we have completed before, the experimental effect of this location should be relatively good, and the height is relatively moderate, but it is close to the edge of the step that will cause interference, so we believe that there are signal waveforms in all directions, which may be the data recorded by the experiment. some problems.   |  | | --- | | D:\南科大文件\MobileFile\IMG_20210525_165413.jpg | |  |   This is the data obtained from the experiment we conducted in the center of the football field. We have analyzed it in the previous article, and we will not discuss it in detail here. We can see that the signal waveform is generally not ideal, but we analyzed that the reason for this phenomenon is that in addition to the high temperature effect, it may also be because the height of our experiment is relatively low, and there is a stand facing the signal source. Obstruction, so the signal will be affected to a certain extent. We analyze that this is also the reason why other angles in the direction diagram also have numerical values.  **This is the location and result of our final experiment after demonstration**   |  | | --- | | **D:\南科大文件\MobileFile\IMG_20210523_134650.jpg** | | **D:\南科大文件\MobileFile\IMG_20210525_164637.jpg** | |  |   **The result of the experiment is relatively good**  **Some pictures in the experiment**    Figure Antenna array in the experiment      Picture Wei Xinyuan is conducting an experiment      Figure Wei Xinyuan and Zhang Zhentao communicate in the experiment according to the technical route of the experiment | |
|  | |
| **Experience**  This is the content of all our final papers. We hope to use a longer and higher quality final paper to bid farewell to our original study. In the process of completing the interim report, we mainly carried out the preliminary preparation and research work of the project. In the process of completing these tasks, we mainly conduct all preparations by consulting data.  In order to make our interim report more organized and better guide our follow-up project work, we divided the preliminary preparations into two parts. The first part is the process of paper review. We have reviewed more than five general and informative related papers, and first have a comprehensive understanding of the entire project process. In the experiment, Mr. Wu repeatedly emphasized that the follow-up research should be guided by systematically learning the introduction of experimental knowledge. This is also what we need to follow in each experiment.  The second part is the process of systematic learning. In this process, we systematically compared the content of this article, and learned the coding process of the algorithm with the help of CSDN tools, and carried out a systematic comparison and comprehensive analysis of these three algorithms. In this process, we have a certain understanding of the calculation process of the algorithm through the simulation algorithm in the data. The simulation process of the algorithm and the optimization of the algorithm have been studied to a certain extent. Analyze the algorithm through the results.  The third part is our process of internalizing knowledge and relearning. We internalize knowledge through systematic learning, and then transform it into our own learning results. During this process, most of the knowledge content in our interim report was edited and compiled by ourselves.  After completing the interim report, we fully launched the final project. In fact, we internally gave a high evaluation of our interim report. This is because we have also carried out part of the final project work during the preparation of the interim report. This is mainly a simulation of the final project.  In fact, this part of the simulation work is not directly required by the experiment. However, we have added some algorithm simulation work to the interim report. Most of these algorithms are open source, we have made certain modifications to them, and have a certain understanding of the advantages of various algorithms. In the process of understanding the algorithm, we have a certain understanding of the underlying logic of the algorithm, which will better guide our subsequent experimental work.  In the experiment, we pre-formed the simulation work. According to our plan, the simulation work should be carried out after the completion of the interim report until May 1st Labor Day. But in actual implementation, we completed most of the simulation work before writing the interim report. The reason why we pre-process the simulation work is because on the one hand we hope to complete the entire experimental work faster, on the other hand we also set aside more time for basic experiments. So as to ensure the high-quality completion of our basic experiments.  After completing the interim report, we mainly carried out preliminary preparations for experimental equipment and experiments. The specific experimentation starts in May. From the completion of the interim report to the beginning of the test, we have made technical improvements. This mainly includes our preparation for the algorithm and coordination of the equipment.  Although we have the open source music algorithm code, we need to make certain modifications to the music algorithm. On the other hand, our equipment is not very complete. We purchased six additional antennas in the experiment and prepared data cables for data transmission. In general, the preparatory work is relatively sufficient, which has laid the foundation for our subsequent experiments to be successful.  In the course of the experiment, we conducted two rounds of technical demonstrations, and these two rounds of technical demonstrations are what we have gained a lot from this experiment. We hope to talk about it in detail.  **Experience in the experiment**   1. The temperature during our experiment was relatively high and it was sunny, while the time of Professor Wang's experiment was cloudy, and the humidity and temperature of the air were also somewhat different from when we experimented. This reflects the problem of control variables. In theory, we need to strictly control the variables in the experiment, and we also need to consider weather and environmental factors more in the process of signal source detection experiments. This is because the transmission of electromagnetic waves is related to the temperature and humidity of the environment, so there should be a slight gap between the transmission results obtained in our experiment and the results of Professor Wang's experiment. If we want to simulate the effect of the actual experiment as much as possible, we need to fully consider the environmental issues during the experiment. That is to say, if we want to study the detection process of the signal source more comprehensively, we need to conduct multiple experiments in different environments. In our experiments, due to the limitations of equipment and time, we have no way to conduct a full range of experiments, but in the subsequent experiments, we need to consider the needs of the experiment as much as possible in all directions, and then complete the experiment. 2. In the experiment, we also overcome many difficulties. One of the main difficulties is the location problem. We chose the playground center as the experimental site at the beginning, but the signal detection effect is not very good. We think it may be a high temperature environment. This caused a certain impact on the performance of the machine. Afterwards, we chose the stadium stand as the experimental site, but the detection effect was relatively poor. Later, we will analyze that it may be due to the larger interference caused by the ladder structure of the stand. In general, we have done a lot of related site selection work, made some screenings, and finally selected the measurement sites in our experiment. 3. The first thing we gained in the experiment was programming ability. Programming ability is essential in our professional courses. In this experiment, our algorithm and programming part is mainly divided into two parts: self-developed part and open source part. The more important music algorithms have more mature open source algorithms on the website. We only need to make appropriate modifications to apply them to our experiments. The self-developed part of our algorithm is mainly focused on the data processing stage. In the experiment, we also have finished programs that can complete the experiment, but we found that the data we got is generally of low quality, and the experimental conclusions may not be accurate enough. Therefore, we chose to use data for processing to complete the experiment. 4. The first thing we gained in the experiment was programming ability. Programming ability is essential in our professional courses. In this experiment, our algorithm and programming part is mainly divided into two parts: self-developed part and open source part. The more important music algorithms have more mature open source algorithms on the website. We only need to make appropriate modifications to apply them to our experiments. The self-developed part of our algorithm is mainly focused on the data processing stage. In the experiment, we also have finished programs that can complete the experiment, but we found that the data we got is generally of low quality, and the experimental conclusions may not be accurate enough. Therefore, we chose to use data for processing to complete the experiment. 5. Then there is the team cooperation ability. The smooth progress and successful completion of the experiment cannot be separated from the cooperation between the team and the team. In the experiment, we have both consensus and disagreement on the experimental direction and technical route. When we have differences in the experiment At that time, we resolved the differences through negotiation and finally pushed the experiment to be completed correctly. When conducting signal detection experiments, we ensured the quality of the experiment and increased the speed of the experiment through cooperation between groups. Although we cooperated between the two groups, the data measurement and data processing processes were completed independently, and the experimental technical routes between the two groups were also different. 6. In the end, we use tools and the ability to read literature. During the entire experiment, we read a certain amount of literature and have a certain grasp of theoretical knowledge related to the experiment. Before conducting specific experiments, a certain simulation was carried out for different doa estimation algorithms. In the experiment, we simplified the experiment process with the help of tools, improved the efficiency of the experiment, and made the quality of our experiments higher. 7. In addition to the gains in the experiment, we still have certain shortcomings in the process of completing the specific experiment. In the experiment, we use the kerborsSDR system to roughly reproduce the location of the base station. However, due to the equipment may have certain problems, the internal noise source cannot be turned on, which makes the equipment unable to synchronize. The experimental site is not an ideal site, so it is subject to some interference and some deviations may occur. The temperature of the experiment place is not very ideal, resulting in high temperature of the experiment equipment, which will affect part of the experiment performance. In general, the quality of the experiment is not low, and the influence of part of the error is kept within a controllable range. 8. This is the signal detection process carried out on the structure of the grandstand. Because the topography of the grandstand stairs is more complicated, we predicted that the results of the experiment may be poor before the experiment, but we still carried out a field experiment, this is because we Hope to accumulate more error samples. The process of accumulating error samples can help us better understand the causes of errors, and it is also a good exercise for our level of analysis. Since this location is located on a ladder, the overall structure is more complicated, and there is no open test site, so we spent a lot of time and energy when doing this experiment.  |  |  | | --- | --- | | D:\南科大文件\MobileFile\IMG_20210525_165115.jpg |  | | |
| **Score** |  |

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