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Design of UAV Detection Scheme Based on Passive Acoustic Detection

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Abstract. With the rapid development of UAV technology, the "black flight" of UAV is an urgent problem to be solved in daily work and life. In this paper, the current research status of unmanned aerial vehicle (UAV) detection system at home and abroad is described, and a design idea of UAV detection system based on sound signal is proposed for radio spectrum monitoring, radar detection, light detection and exploratory problems. Combined with the needs of the problem, aiming at the three key problem links of the system, the paper puts forward specific solutions to provide the foundation for the system implementation and application in the next step.

1. Introduction

In the process of daily work and life, UAV has become an important factor that interferes with navigation safety. Especially in the vicinity of the airport, the aircraft landing process, often encountered near the navigation target of personal drones into the no-fly zone. Flight attendants at airports have relatively limited means to deal with drones flying illegally. In particular, the sudden take off of the UAV, it is difficult to timely and effective disposal of the situation. If management is carried out in the way of expanding the scope of management, it will cause greater waste of manpower and material resources, resulting in low management efficiency. The control of low-altitude aircrafts is an urgent problem in the management process.

At present, domestic and foreign UAV detection technologies can be roughly summarized into radio spectrum monitoring, radar detection, acoustic detection and other categories, among which radio spectrum signal monitoring and radar detection methods have been relatively mature and some market-level products have appeared. However, there are relatively few researches on UAV detection system based on acoustic monitoring, namely microphone array. The sound source detection UAV is an effective supplement to the current UAV detection system.

2. Analysis of UAV Detection System Based on Acoustic Signal

The sound waves of the drone are derived from the air vibration caused by the motor work and the rotation of the rotor. Since the generated flight noise is significantly different from the audio characteristics of other objects in the airspace, the UAV is used for identification and positioning. The system is based on the microphone array system for research and design, mainly focusing on the research of high-speed microphone array acquisition system with small size and convenient arrangement, feature analysis and recognition of unmanned aerial sound signals and sound target positioning based on microphone array.

2.1. Modular microphone array high-speed synchronous acquisition platform
The data acquisition system mainly performs signal conditioning, sampling processing, and storage or transmission. At present, the commercial data acquisition system mainly appears in the form of a board or an industrial control host. Although the performance is relatively strong, the entire collection system has a large volume and is not suitable for flexible arrangement in the application environment in the form of nodes. Therefore, a portable, compact acquisition platform suitable for flexible deployment is the key to the realization of the UAV detection system.

The acquisition platform adopts a modular design idea, and the data sampling board and the data acquisition and processing board are separately designed to change the number of data acquisition channels, so that the system can collect the system. Specifically, the acquisition system is mainly composed of a pre-set acquisition module, central processing control, and software design.

2.2. Sound signal recognition
The sound signal recognition mainly includes sound feature extraction, sound feature database construction and feature classification judgment.

Since the composition of the sound signal is relatively complicated, it is difficult to directly use the time domain signal for classification and recognition. Generally, a feature vector representing the original signal is extracted from the signal to replace the signal for recognition. There is a big difference between the generation mechanism of the sound signal of the drone and the generation mechanism of the voice signal. For example, the sound signal of the rotor type drone is mainly generated by the friction between the rotor of the drone and the atmosphere, but from the two sound signals. Based on the basic characteristics, it is verified that the two sound signals belong to the short-term generalized stationary random signal. Therefore, the common sound signal feature extraction method can be used as the basis, and the sound signal characteristics of the drone can be extracted appropriately. By studying several kinds of sound features such as linear predictive cepstral coefficient (LPCC), short-time Fourier transform (STFT) and spectroscopic saliency, the sound characteristics of unmanned aerial vehicles are studied.

The more mature classification methods used in voice recognition are K-NN, support vector machine (SVM), hidden Markov model (HMM), Gaussian mixture model (GMM), neural network and sparse representation. In order to further improve the accuracy of recognition, we have used several methods of combining several classification methods, such as combining HMM with SVM, combining SVM with GMM, combining SVM with K-NN, and combining HMM with GMM. In recent years, with the wide application of deep learning in various fields, it has also been applied to voice recognition. In the actual recognition, different feature vectors are extracted, and different recognition methods are generally used to obtain different recognition accuracy rates. Under the premise of computational complexity, the eigenvectors and recognition methods are combined reasonably, and the algorithm suitable for UAV target recognition is selected and transplanted to the embedded system.

2.3. UAV positioning technology based on acoustic signal
The acoustic signal-based localization technique uses the microphone array to estimate the DOA of the sound source, also known as spatial spectrum estimation. At present, the mature broadband signal DOA estimation technology mainly includes the non-coherent signal subspace (ISM) method, the coherent signal subspace (CSM) method and the maximum controllable response power beamforming algorithm (SRP-PHAT) based on the MUSIC algorithm. Compressed sensing based DOA estimation method.

3. System design
The system mainly focuses on the passive identification and direction finding of unmanned aerial vehicles based on acoustic signals, and builds a modular reconfigurable microphone array acquisition platform, UAV sound feature extraction and recognition, and a microphone array based UAV positioning technology.

3.1. System architecture

The composition of the Zynq-based microphone array synchronous high-speed acquisition system is mainly shown in Figure 1. The system is mainly composed of MIC array module group and A/D sampling module group. (A/D processing chip selects TI's PCM4204 and its sampling rate is up to 216KHz and the sampling digit is 24 bits.)

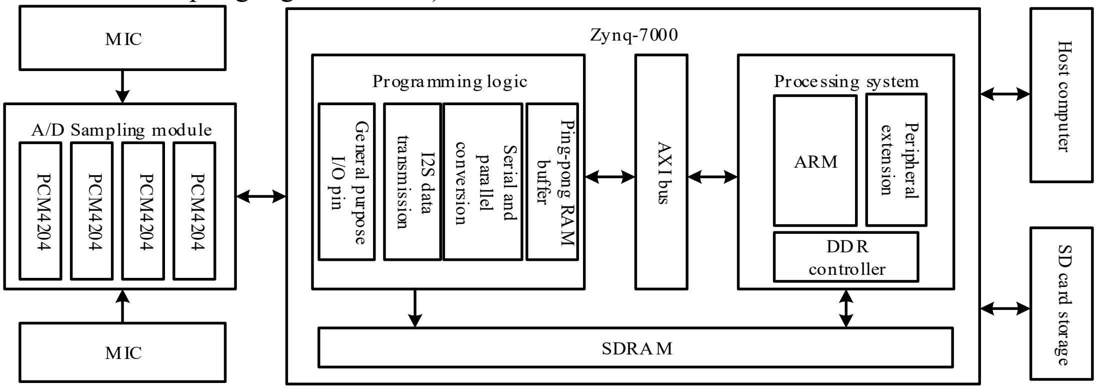

Figure.1 Comparison of frequency domain comparison of different drone acoustic signals

The ambient sound is collected by the MIC array module group. The analog signal is amplified by the op amp and sent to the A/D sampling module to be converted into a digital signal, which is sent to the PL end of the Zynq-7000 through a general-purpose I/O pin. Since the Philips I2S audio data format is serial transmission, serial-to-parallel conversion is performed on the PL side, and buffering is performed in the BRAM block by the Ping-Pang operation. The PS side and the PL side communicate and communicate via the AXI bus. The operating system is ported in ARM, and the data cached by the PL side is read through the BRAM controller. Two data storage modes are realized through the button control of the PL end, which are respectively transmitted to the upper computer through the network port for storage; or stored in the SD card by the Micro SD peripheral. The final digital signal is saved in a binary media file (.dat) format.

In ARM port Linux operating system, the process includes: compile u-boot, compile Linux kernel, generate device tree, generate file system. In the complex electromagnetic environment and the "quiet" state of the UAV They all work normally and can effectively overcome the limitations of radar detection "blind zones" and optical detection of small objects.The system opens up two processes, which are responsible for writing the data transmitted by the PL end into the system cache DDR3 and judging that the operation mode is forwarded by the network port or the SD card. Read data according to the transfer mechanism, store it in DDR3, and clear the flag bit. The base address of the BRAM block is from 0x4000_1FFF to 0x5400_3FFF, corresponding to the transmission channels PORT1 to PORT15, respectively.

3.2. Feature extraction algorithm

The sound signal of the drone can be approximated as a generalized stationary random signal, so the algorithm for processing the generalized stationary random signal can be used to analyze the sound signal of the drone. This paper intends to extract the sound characteristics of the drone using the Mel frequency cepstral coefficient (MFCC).

The Mel frequency cepstrum coefficient is an audio feature based on the analysis of the filter bank. It is a feature extraction method based on the nonlinear characteristics of human hearing. It has good recognition performance and anti-noise ability.

The extraction process of the Mel frequency cepstral coefficient has many similarities with the linear prediction cepstral coefficient (LPCC) extraction process, but the MFCC process translates the linear frequency domain into the Mel frequency domain, which is actually a linear filter bank. Convert to a Mel filter bank.

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The Mel cepstrum coefficient is divided according to the Meyer scale in the frequency band division. It is more similar to the human auditory system than the linearly spaced frequency band used for the normal cepstrum spectrum, so that the sound characteristics of the drone can be effectively extracted. . Therefore, the scheme uses MFCC as the feature extraction algorithm of the system.

3.3. Location Algorithm

For the positioning problem in complex environment, it is more convenient to combine the actual military application with the maximum controllable response power beamforming algorithm to process the sound signal.

The SRP-PHAT algorithm is a combination of delay-accumulated controllable response power (SRP) and phase shift (PHAT) methods. The algorithm makes full use of the short-term analysis characteristics of the former and the latter's insensitivity to environmental changes. When using the algorithm to represent the received signals of the array elements of the microphone array, it is usually necessary to consider indoor reverberation. The output power of the algorithm can be expressed by:

$$p(\mathbf{q}) = \sum_{m=1}^{M}\sum_{n=1}^{M}\int_{-\infty}^{\infty} \psi_{mn}(\omega) X_m(\omega) X_n^*(\omega) * e^{j\omega(\tau_n - \tau_m)} d\omega$$

Wherein, $X_m(\omega)$ is the windowed Fourier transform of the $x_m(t)$ signal received by the first microphone array element; τ represent the time delay when the array elements of the microphone array receive signals; q is the spatial position vector of the sound source; For the microphone array, the expression for $\psi_{mn}(\omega)$ is:

$$\psi_{mn}(\omega) = \frac{1}{\left|X_M(\omega) X^*{}_N(\omega)\right|}$$

Definition $R_{mn}(\tau)$, a generalized cross-correlation (GCC) function representing the received signal of the first microphone element and the first microphone element, and the time domain power representation of the method is obtained by:

$$p(q) = 2\pi \sum_{m=1}^{M}\sum_{n=1}^{M} R_{mn}(\tau_n - \tau_m)$$

τ_n, τ_m represent the delay of the received signals of the n and m microphones relative to the received signals of the reference microphone array. For the above formula, the DOA estimation algorithm expands the search in space to find the position with the largest power, and the corresponding orientation is the spatial spectrum estimation angle.

The algorithm has no complicated operations and is suitable for implementation in hardware systems. It can realize the miniaturization, portability and real-time of the system. Especially in the complex environment of the battlefield, the system can better accomplish the task when there are certain requirements for carrying, laying and running. Therefore, the scheme selects the SRP-PHAT algorithm as the positioning algorithm.

4. Conclusion

This paper analyzes the current research status of the UAV detection system for the UAV "black fly" problem, and exploratively proposes an unmanned aerial vehicle detection system based on acoustic signals. The system consists of a hardware acquisition system and a software algorithm part. Compared with the existing UAV detection system, the UAV target detection system based on acoustic signal is a passive detection system that is not easily interfered by electronic countermeasure equipment. In the complex electromagnetic environment and the "silent" state of the UAV, it works normally and can effectively overcome the limitations of radar detection "blind area" and optical detection of small objects. The proposed acoustic detection technology is not to replace other UAV detection technologies and systems, but to provide a complementary detection method for the problem that existing UAV detection systems and means cannot achieve effective detection under the above circumstances. Thereby, the diversity of the detection means of the UAV is realized, and it is ensured that it meets the detection needs in a complex environment, and the requirements of all-weather and multi-conditions are achieved.

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