Acoustic Signature Analysis for Localization Estimation of Unmanned Aerial Vehicles Using Few Number of Microphones

Torea Blanchard^{1,*}, Jean-Hugh Thomas^{1,2,**}, and Kosai Raoof^{1,2,***}

¹LAUM - UMR 6613, Avenue Olivier Messiaen, 72085 Le Mans, France

²ENSIM, 1 rue Aristote 72000 Le Mans, France

Abstract. In recent years, the current technological improvements of Unmanned Aerial Vehicles (UAV) have allowed more and more efficient use for applications ranging from simple amateur shooting to more professional tasks. Being handy, drones can easily fly near sensitive sites, such as power plants, airports or ministries. It is therefore necessary to develop systems able to keep watch such sites. However, the size and the composition of these devices make optical or electromagnetic systems inefficient for their detection. This study proposes an alternative exploiting the sound wave emitted by their motorization and / or their aerodynamic whistling. For this, an acoustic antenna with few microphones was sized to be sensitive to frequencies emitted by a drone. Firstly, an experimental analysis on the noise made by a drone was carried out. The results allowed us to settle the geometry of the antenna in order to process localization. Two methods of location are used. The first is based on an energy approach providing the acoustic field reconstruction in all directions of space. The second estimates the position of the source by inverting a system exploiting the arrival time differences of the acoustic wave between different pairs of microphones. Numerical simulations, supported by a measurement campaign, make it possible to highlight the performance of the methods used.

1 Introduction

Nowadays the use of Unmanned Aerial Vehicles (UAV) is more and more handy for applications ranging from simple amateur shooting to professional tasks. However some uses of these devices may be illegal or even show threatening behaviors. Indeed drones can easily fly near sensitive sites such as power plants, airports or ministries since their size and the materials used to develop them make optical or electromagnetic systems inefficient for their detection. In order to develop systems able to keep watch such sites, new techniques have to be studied. Acoustic methods exploiting the sound wave emitted by their motorization and/or their aerodynamic whistling can be used as an alternative.

The aims of the systems exploiting the acoustic characteristics of the UAV are (i) to be able to detect their acoustic signature, (ii) to estimate their position and (iii) to track their movement along the time. This study focuses only on the localization task and is restricted by the use of a few number of microphones. Two different approaches are used to estimate the direction-of-arrival (DOA) of the sound emitted by the source. The first method, known as the so-called beamforming technique, rebuilds the acoustic field in all directions of space. In the case of a single source without any noise the source direction is the one providing the most energy. The second approach is

based on the so-called time-difference-of-arrival (TDOA) estimation. The source direction is inferred using only the TDOA, the antenna geometry and the speed of sound information.

An experimental analysis of the sound emitted by a small quadri-motors UAV was first carried out in order to design an acoustic antenna which is adapted to the frequencies identified in the signal. Recordings in anechoic and real acoustic environments were made to evaluate the localization performance of both approaches. The paper is organized as follows, first the two localization techniques are briefly described. The performance of each method is evaluated through simulations for a given antenna geometry. Then some results on measurements made under both ideal and real conditions are shown and discussed.

2 Localization techniques

2.1 Localization estimation using beamforming techniques

Beamforming techniques are generally done in three steps. The first step consists of time-shifting each sensor signal in order to form the beam in the desired direction. Signal processing techniques are then used to design the beam pattern depending on the constraints encountered. Finally every sensor signals are summed giving the output of the beamformer. When the sensor signals are sync or equivalently once the beam is steered in the source direction, the power of the output of the beamforming is maximal.

*e-mail: torea.blanchard@hotmail.fr

**e-mail: jean-hugh.thomas@univ-lemans.fr

***e-mail: kosai.raoof@univ-lemans.fr

Many filtering processing improving the beampattern are proposed in the litterature as the Minimum Variance Distortionless Response (MVDR), known as the Capon beamforming or the Maximum Signal-to-Noise Ratio Filter [1,2]. Under some conditions these filters are strictly equivalent in the way that they maximize the Signal-to-Noise Ratio (SNR) of the input signal. Differences with fixed beamforming techniques as the Delay & Sum (DS) process lie in the fact that adaptive techniques are preferred to deal with the noise environment while keeping maximum noise reduction [3].

2.2 Direction-of-Arrival estimation

In the case of a far-field condition only, the Direction-of-Arrival (DOA) estimation of the source signal can be directly inferred from the Time-Difference-of-Arrival (TDOA) estimation. For instance in the case of a signal impinging two microphones, spaced by d from an angle of θ , the TDOA τ is:

$$\tau = \frac{d}{c} \cos \theta, \quad (1)$$

where c is the speed of the sound in the air. In a three dimensional context, the DOA of a signal source is obtained by combining the TDOA estimation from multiple microphones. In that case, the problem takes the following form [4]:

$$\begin{aligned} \tau &= c^{-1} \mathbf{D} \mathbf{n}_s, \\ \tau &= [\dots \tau_n \dots]^T, \mathbf{D} = [\mathbf{x}_1 \dots \mathbf{x}_n \dots \mathbf{x}_{M-1}]^T. \end{aligned} \quad (2)$$

where τ_n is the TDOA between the n -th microphone and the reference one, $\mathbf{x}_n$ is the vectorial position of the n -th microphone and $\mathbf{n}_s$ the normalized vector pointing the source position. $[\cdot]^T$ indicates the transpose operator. To ensure a unique solution of Eq. 2, $\mathbf{D}$ must be invertible which is the case if the antenna geometry is not plane. TDOA are estimated using Generalized Cross-Correlation (GCC) methods [5]. This approach is also known as acoustic goniometry due to its capability to give an angle estimation of an acoustic source.

3 Simulation

Simulations were carried out in order to evaluate the localization performance of both approaches described in Sec. 2. The geometry of the antenna is given in figure 1. The array is composed of ten microphones put on three arms. The reference microphone is placed at the center. On each arm, the first microphone is 5 cm from the reference one, the second is 20 cm and the last, 110 cm. An acoustic source emitting at a single frequency f was placed at $r = 15$ m in the direction $(45^\circ, 35^\circ)$ so that the far-field condition is respected. The frequency was chosen such that the wavelength of the acoustic wave equals twice the smallest inter-microphone distance d_{min} in order to respect the Shannon's theorem ($d_{min} \leq \frac{c}{2f}$). In this case the frequency is 3400 Hz.

Figure 1: Scheme of the antenna geometry composed of ten microphones. This configuration was used for the simulation and the experiment.

The DS method was used for the beamforming process. TDOA are estimated using the classical cross-correlation function. The algorithm was written to give the angular errors both in azimuth and elevation and the input SNR defined as:

$$\text{input SNR} = 10 \log_{10} \left(\frac{p_s}{p_n} \right) \quad (3)$$

where p_s and p_n are respectively the power of the source and noise signals. Figures 2 and 3 give the angular errors estimated as a function of the input SNR. The noise is modeled by a zero-mean Gaussian process with variance σ^2 . A number of ten realizations was done for each SNR. The estimates are then averaged. These results show that beamforming is more robust against noise than goniometry. This is certainly due to the fact that the input SNR is improved, in the beamforming case, since redundant information recorded at each microphone allows to increase the output SNR by a factor M (where M is the number of sensors of the array) when the noise is spatially uncorrelated. Up to about -35 dB the beamforming algorithm is able to locate the source with errors less than 2.8° . Acoustic goniometry ensures localization errors less than 0.5° up to 20 dB. Beyond that, the errors increase in a logarithmic way.

4 Measurement of a small Unmanned Aerial Vehicle

4.1 Recording in an anechoic environment

Measurements were carried out using a small Unmanned Aerial Vehicle (UAV) from DJI company in the anechoic chamber of the laboratory. The drone is composed of four bipal propellers. First, only one microphone was used in order to identify frequencies emitted by the vehicle. Harmonics are present up to about 6 kHz. Using a high-speed camera, recording 4000 frames per second, it has been possible to identify the origin of the harmonics. One part of them (those with lower energies) are of mechanical origin. The others are of aerodynamical origin. The perceived sound is mainly coming from aerodynamical turbulences.

Figure 2: Azimuth and elevation angle error evaluation as a function of the input SNR using Delay & Sum technique. The two last plot give a zoom in the range -15 and -55 dB.

Figure 3: Azimuth and elevation angle error evaluation as a function of the input SNR using goniometry method. The two last plot give the errors in the range 0 and 40 dB.

Figure 4: Azimuth and elevation angle error estimates using DS beamforming and goniometry techniques on a small quadri-motors UAV.

The localization step was made using the antenna geometry described in Sec. 3. In this configuration the studied frequency band is between 309 Hz and 3400 Hz according to the following relationship ($c = 340 \text{ m.s}^{-1}$) $\frac{c}{L} < f < \frac{c}{2d_{min}}$, where L is the antenna size. The DS algorithm and the GCC method with a β – PHAT weight function ($\beta = 0.5$) were used for the localization process. The drone was first placed in the ground at 3 m from the center of the antenna [in the direction $(45^\circ, 5.7^\circ)$, the center of the drone was considered in the same plane as the propellers]. The drone was flying vertically from the ground at a height of about 3.5 m with a constant speed. The tracking of the engine was ensured with a 0.5 s refreshing of the algorithms. The input SNR measured was about 43.5 dB. Figure 4 shows the azimuth and elevation errors for each method. Errors are less than 12.5° excepted for one point.

4.2 Measurements in real conditions

Localization measurements were carried out in real conditions and with the same drone as the one used in the anechoic chamber. Recordings were made for several trajectories. Only results for one trajectory are presented. Both the whole path followed by the drone and the trajectory studied are shown in figure 5.

Global Positionning System (GPS) coordinates were recorded by the drone and converted into its polar positions with the center of the antenna as the reference. Positions are compared to those estimated by beamforming and goniometry. The recording lasted 12 s. The drone was flying at an average height of 7 m from the south to the north and the antenna was oriented northward. Positions estimated by beamforming and goniometry are given for each time-window of 0.5 s. The input SNR measured is approximately 25 dB.

The results given in figure 6 show that the methods are able to globally track the drone along its azimuthal angle. However tracking along the elevation angle is more subject to high errors. This can be due to many factors as the presence of multiple sources as birds, sounds of vehicles from the road, short-time winds, or the reflexions coming

Figure 5: Plot of the path followed by the UAV during the whole flight (left) and the portion of the trajectory presented here (right).

Figure 6: Azimuth and elevation angle estimates of the drone using GPS coordinates, DS beamforming and goniometry techniques.

from the ground. The Doppler effect has not been integrated into the signal model for the localization process also.

5 Conclusion

Based on both simulations and measurements, this study has shown that acoustic localization processing is able to locate moving sources more or less efficiently according to the constraints encountered. Simulations showed that beamforming is more robust to noise than goniometry. This is due to the use of redundant information collected from all microphones. Recordings in ideal conditions (anechoic chamber) were carried out and the DS technique and the GCC method were used to track the drone during its flight. Results are encouraging. The algorithms were able to track the drone during the whole flight with errors less than 12.5° . Results in real conditions are less relevant since the tracking of the device was globally achievable along its azimuth angle only. Many factors, that can explain these results, have not been yet into account as the number of sources present in the recordings (birds, other vehicles, wind), the Doppler effect or even the reflexion from the ground. This study is encouraging in the sense that acoustic methods show interesting properties to locate and track objects emitting sound.

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