Acoustic Localization For Mobile Robots

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***Resumo* – Existem diferentes técnicas usadas pelos humanos para estimarem a localização de fontes de som. Uma delas baseia-se na diferença de fase do sinal captado por cada um dos ouvidos devido ao percurso executado pela onda sonora ter comprimento diferente. Este algoritmo pode ser implementada num Robot equipado com dois microfones e com recurso a algum processamento de sinal. Neste artigo mostra-se que utilizando um sinal sinusoidal com duas frequências é possível estimar-se com alguma precisão o ângulo de incidência da onda sonora mesmo na presença de ruído. Esta medida pode ser utilizada para a estimativa da localização de múltiplos Robots.**

***Abstract* - There are many techniques used by a human to estimate the location of a sound source. One of then is based on the phase difference on each ear due to a longer path taken by the acoustic wave to reach the farthest ear. That technique can be reproduced on a mobile robot equipped with two microphones and modest processing power to apply the proposed algorithm. It is shown that with two frequencies the angle in the horizontal plane of the sound source can be estimated even under heavy noise levels. That can contribute to the relative localization of multiple mobile robots.**

I. INTRODUCTION

The cues used by a human to localize the source of a sound are many and some of them are not yet fully understood. There is a great body of research on that subject. Some are easier than others to model and reproduce on a mobile robot. The phase shift of the perceived sound on each ear is the one of the most powerful cues used by humans so it is widely used on positional sound synthesis. A mobile robot, to extract the angle of incidence (the azimuth) of an acoustic wave, can use a similar technique. The problem can be made simpler because there are extra design options: the distance between the sound transducers can be wider and the head does not necessarily block the sound path.

Assuming the we are using this approach to obtain the relative positions of multiple mobile robots, we have the extra advantage of being able to design the emitted sound so that some desired properties are met. In our case we will ensure that the sound signal contains essentially two selected frequencies that can be used to extract the angle of the incident acoustic wave. The performance of this approach can be accessed under several situations where the signal is corrupted with white noise.

II. DETECTING THE DIRECTION OF AN ACOUSTIC PLANAR WAVE

We are only interested in the azimuth of the sound source. To estimate the elevation a third microphone could be use but the principle behind that approach would remain the same.

The robot acoustic system is composed by two microphones connected to a sound card that can sample with a frequency of 44.1kHz and with 16 bit resolution. The embedded PC takes care of the audio processing. The heaviest operation is the 1024 bytes FFT that is needed by the algorithm. A possible speedup is to calculate only the coefficients of the desired frequencies but the price is a less general implementation of the algorithm

In figure 1 it is shown the relative position of the microphones and the incident acoustic wave. We consider that the acoustic wave maintains the propagation direction constant and the microphones rotate β radians. That angle can be estimated from the perceived phase on both microphones. The following equations describe the relation between these variables.

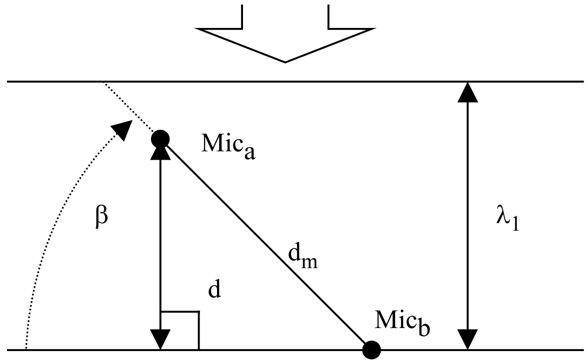

Fig. 1. Both microphones and the incident acoustic wave.

If at least one of the frequencies present on the generate sound has a wavelength (λ_1) greater than the distance between the two microphones (d_m), the phase difference (φ_1) is related with the wavelength and relative microphones position by (see figure 1):

$$\frac{d}{\lambda 1} = \frac{\varphi 1}{2\pi} \qquad (1)$$

The relation between the distance d measured along the wave propagation direction and the angle β is given by:

$$d = d_m \sin(\beta) \qquad (2)$$

replacing equation (2) in (1) results:

$$\beta = \arcsin\left(\frac{\varphi_1 \lambda_1}{2\pi\, d_m} \right) \qquad (3)$$

that is the value that we want to estimate.

These formulas assume that the angle β is between $\pi/2$ and $-\pi/2$. That knowledge must come from some prior expectation on the robots' positions. There is another planar wave that yields the same phases on the microphones and it is necessary some extra knowledge to choose the right direction. For lower values of β it is easier to choose the right direction because the other candidate is almost π radians apart.

Supposing that we have a second sinusoidal signal with a shorter wavelength λ_2 as shown in figure 2 it become possible to refine our measurement because with the lower wavelength there is more resolution on the result obtained from the phase difference.

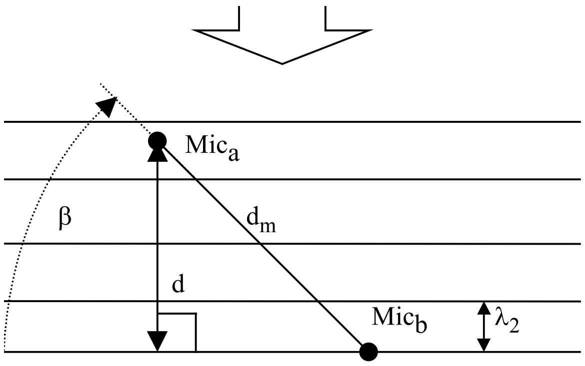

Fig. 2. The case when the incident acoustic wave has a shorter wavelength λ_2.

In this case, the phase difference is given by:

$$\varphi'_2 = \varphi_2 + k\, 2\pi \qquad (4)$$

with $0 < \varphi_2 < 2\pi$.

If we define d_2 as the fractional part of the distance d:

$$d = d_2 + k\, \lambda_2 \qquad (5)$$

then the relation between the phase difference φ_2 and the distance d_2 is similar to (1):

$$\frac{d_2}{\lambda_2} = \frac{\varphi_2}{2\pi} \qquad (6)$$

and we have:

$$d_2 = d_m \sin(\beta) - k\, \lambda_2 \qquad (7)$$

replacing equation (7) in (6) results:

$$\beta = \arcsin\left(\frac{(\varphi_2 + k\, 2\pi)\lambda_2}{2\pi\, d_m} \right) \qquad (8)$$

where k can be found by:

$$k = \mathrm{floor}\left(\frac{\varphi_1 \lambda_1}{2\pi\, \lambda_2}\right) \qquad (9)$$

III. RESULTS

In the presented results the noise was simulated and injected over the signal. With this approach it was easy to adjust the signal to noise ratio. In figure 3 it is shown the original signal and the corrupted one. To obtain the estimate distribution the experiment was repeated 500 times for each case.

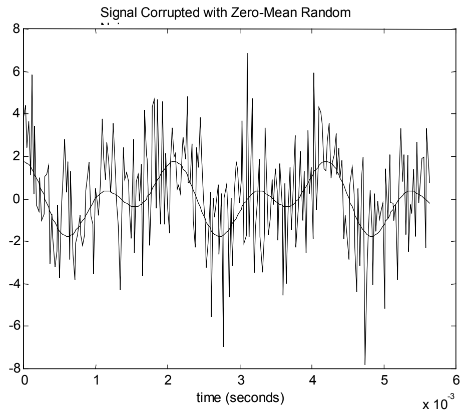

Fig.3. The audio signal corrupted by white noise.

Of course the S/N is a lot more favorable if we consider only the interesting frequencies. Anyhow, that is something that can generally be achieved if there is some prior knowledge of the typical noise power spectrum.

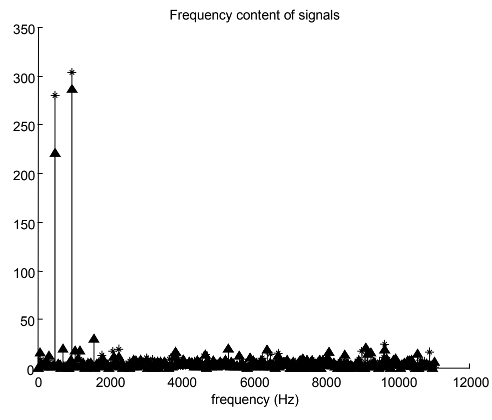

Fig. 4. The audio signal power spectrum on both microphones.

S/N	4		1		0.25	
β	**Mean**	**Cov**	**Mean**	**Cov**	**Mean**	**Cov**
15º	14.99	0.3586	15.05	1.509	14.95	5.627
30º	30.02	0.4576	29.95	1.866	30.12	7.279
45º	44.98	0.6369	44.99	2.710	44.98	12.24
60º	60.03	1.483	59.98	5.806	60.73	26.09
75º	75.08	5.837	75.94	26.54	76.26	74.62

Table 1 - Estimated azimuth (beta) for some signal to noise ratios with $\lambda = 0.724$

From Table 1 we can see that even under a very unfavorable S/N conditions the estimate appears to remain unbiased. Only the covariance reflects the noise interference. Of course, for a single estimate we can obtain an erroneous value.

In Table 2 we have a more favorable situation only by selecting a shorter wavelength. Both measures can be combined to lower the estimate covariance.

S/N	4		1		0.25	
β	**Mean**	**Cov**	**Mean**	**Cov**	**Mean**	**Cov**
15º	15.00	0.098	14.99	0.409	15.07	1.552
30º	30.01	0.118	30.07	0.439	30.19	2.072
45º	45.00	0.172	44.95	0.743	45.12	2.926
60º	60.01	0.351	59.98	1.537	60.15	5.271
75º	75.10	1.424	75.19	5.592	75.71	29.68

Table 2 - Estimated azimuth (beta) for some signal to noise ratios with $\lambda = 0.362$m

IV. CONCLUSION AND FUTURE WORK

A technique to extract the azimuth of the sound origin using two microphones was presented. The described algorithm is not computationally demanding and can be easily added to any mobile robot equipped with medium computational power.

The present technique can extract the horizontal direction of a generated sound even under extremely high levels of ambient noise. The quality of the direction estimate can be improved if a characterization of the ambient noise can be obtained and the used frequencies are placed where the signal to noise ratio is optimal. We hope to explore that possibility.

This work can be extended to ultrasonic frequencies. Of course, the higher sampling rate will place an extra burden on the acquisition system and the processing power must be increased because of the higher numbers of samples to process. Another problem with ultrasonic frequencies is the lower wavelengths involved. That will make impossible to have an unambiguous measure of the angle with only one frequency. A set of carefully chosen frequencies must be used to uniquely identify the correct angle.

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