MICROPHONE LOCATION BY TIME OF FLIGHT TRIANGULATION

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INTRODUCTION

Location of microphones is important for a number of reasons: (1) The field pattern of a microphone is a function of frequency; which determines the timbre (musical quality) of the recording, (2) The relative phase delays to multiple microphones effects the stereo image created, and (3) being able to recreate desired microphone placement is especially important for sound-on-sound recording, where multiple tracks are recorded at different times [1-2]. While some professional studios do have equipment that can determine the microphone location, this research is aimed to aid those studios that are lower budget, and for something that can be universally used. Previous literature reports has shown a couple of ways to determine the sound source location using microphones [3-4]. One method includes microphone arrays and energy-based position estimations [3]. Although this approach does not require accurate time synchronization, in a recording studio all equipment is usually run through one central digital audio station (DAW), so the ability to sync with in real time is important. Also, the resolution (program calculated accuracy) of 21 cm is quite poor for use in a recording studio since the precision of microphone placement really needs to be much better. Another method with much better resolution of 3 cm uses microphone arrays and time difference of arrival to determine the sound source 2-D location. With the use of microphones spaced out along an x-y plane at known distances and the time differences of the sound to the microphones, the x-y coordinates can be discovered [4]. With this information, there was still some educational gap between how to find the microphone location rather than source location. Time-of-flight triangulation measures the time it takes an acoustic wave to travel a distance through a medium, where the signals from a source arrives at different microphones at times proportional to their distances. The relative time delay between two microphones can be used to compute the microphone location.

OBJECTIVE

We are trying to verify a novel mathematical model to determine the 2-dimensional microphone location with the use of time-of-flight triangulation. This objective will be tested by comparison of the proposed mathematical model to the measured microphone location (cm).

SUCCESS CRITERIA

Based on previous literature results, it was determined that this novel microphone location model with be successful if the average difference between the measured and computed x-y location from all 5 trials is less than 3 cm [4]. The expected difference is approximately 7.7 mm, since the calculated model resolution is ~ 7.7 mm.

METHOD

All 5 experiment trials were performed in the Music Engineering Laboratory (MEL) at the Swanson School of Engineering, University of Pittsburgh. The facility itself incorporates soundproof walls and door, and acoustic ceiling tiles in order to create a quiet atmosphere within the room. The MEL already has existing equipment (microphones, piano, drums, computer with Digital Performer, cables, etc.) that is always ready to be used, so no extra equipment is necessary for the experiments.

Non-compressed sound files (.wav or .aiff) were used throughout the experiments because they fully preserve information. Lossy compression (.mp3) is when the compressed data is not the same as the original data, but rather a close approximation [1]. The compression is achieved by reducing the accuracy of certain parts of sound that are beyond the resolution of human hearing – *perceptual coding*. The information that is removed by the perceptual coder is called the *irrelevancy* of the signal. The *redundancy* and *irrelevancy* are removed in order to provide the lowest bit rate possible for a given audible quality [2]. Sound files (.wav or .aiff) were imported into Matlab using the *waveread* function.

To perform two-dimensional triangulation, the stereo sound files were created to be played through the 2 near-field monitors located at known distance *s* (Figure 1). These consisted of high frequency signal delayed between the left and right speakers long enough to be able to unambiguously determine the source in the MEL. The sound signal was created in Digital Performer (DAW Software) by recording a handclap. The recording was then played back through each speaker, while placing a reference microphone at a known distance *m* and a target microphone at previously measured distances. When the recording was played back, the microphone's detection of the handclap from the speaker was recorded into Digital Performer. Due to the high-frequency nature of the single-sample impulse sounds, it was assumed that virtually all of the energy emanated from the tweeters rather than the woofers. So, the microphones were placed in the same plane.

With Matlab, correlation was used to show how related the sound waves were at the different microphone distances and calculate the time delay between the sound files by simply subtracting the length of the sound files for each microphone and speaker combination (4 total) by the maximum lag index. We do not look at the distance between the speaker and the microphone because the correlation between the wave files is poor. We can find the distance using the rate-time-distance formula where the distance is d_i, the rate is the speed of sound is c, and the time delay which is the difference between first detection and the second detection is $t_2 - t_1$.

$$d_i = c(t_2 - t_1)$$

Now using length L (Figure 1) and d_1 and d_2 we can determine the radial distance from the speakers to the target microphone, R_i by:

$$R_i = L - d_i.$$

Then with the use of two quadratic equations and some simple algebra, the x-y location of the target microphone can be determined using the following equations:

$$x = \frac{{R_2}^2 - {R_1}^2}{4s} \qquad [1]$$

$$y = \sqrt{{R_1}^2 - (x - s)^2} \qquad [2]$$

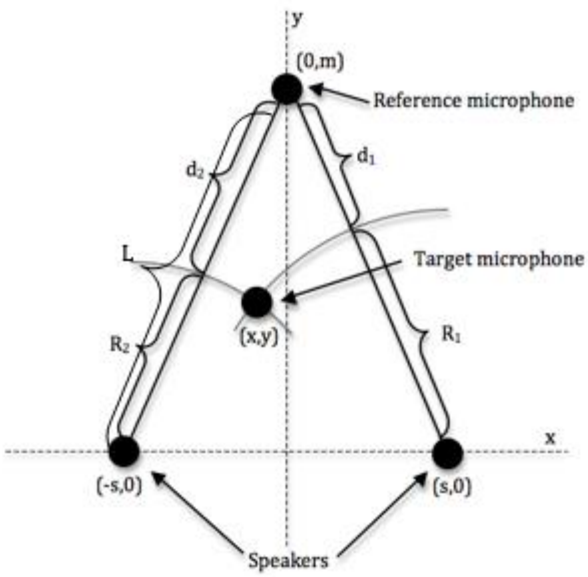

Figure 1. *Schematic of the experimental setup to determine the 2-D location of the target microphone.*

RESULTS

The results from the five trials can be seen in Table 1. The average difference between the measured and model x-y location was (1.34, 2.38) cm or for all measurements 1.86 cm.

Measured x-y location (feet)	Model x-y location (feet)	Absolute x-y difference (feet)	Absolute x-y difference (cm)
(0,5)	(0.03,4.91)	(0.03,0.09)	(0.91,2.74)
(-1,6)	(-1.06,5.93)	(0.06,0.07)	(1.83,2.13)
(1,6)	(0.96,5.94)	(0.04,0.06)	(1.22,1.83)
(-2,3)	(-1.99,2.89)	(0.01,0.11)	(0.30,3.35)
(2,3)	(1.92,2.94)	(0.08,0.06)	(2.44,1.83)

Table 1. *Compare x-y location of the measured and mathematical model to determine absolute difference.*

DISCUSSION

The results show that the success criteria was met for the proposed model by having an average difference of 1.86 cm (< 3 cm). The data suggests that this model is better than any other previously reported literature on 2-D microphone location. However, the results are higher than the expected difference of 1.0 cm. Possible sources of error were: (1) the measuring tape used did not give the exact distances, (2) the microphones were not placed perfectly in plane with the tweeters from the speakers, and (or) (3) the microphones were angled slightly from horizontal.

This system has the potential to be implemented into any music recording studio in order to determine an accurate and reliable microphone location. But, this is just a preliminary study, testing the basic methods of time delay to measure location where two speakers allow basically 2D operation. The major limitation of the study is that the speaker and reference microphone locations need to be known before running the program. One way to improve on the existing model would be to over constrain the system with an additional speaker to possibly eliminate the need to know these distances. A third speaker would also allow for full 3D location of the microphone. Another future extension would be to use the frequency dependent field pattern of the particular microphone along with a more complex transmitted pulse to determine the orientation of the microphone.

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REFERENCES

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