Acoustic Sensors for Biobotic Search and Rescue

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***Abstract*—Advances in neural engineering have enabled direct control of insect locomotion. Insect biobots, with a natural ability to crawl through small spaces, offer unique advantages over traditional synthetic robots. A cyberphysical network of such biobots could prove useful for search and rescue applications in uncertain disaster environments. Our previous work has demonstrated control of Madagascar hissing cockroaches using a Kinect-based computer vision platform. We now demonstrate low-power insect-mounted acoustic sensors for future use in both environmental mapping and localization of trapped survivors. Our experimentation has shown the capability of an insect mounted array of microphones to localize a sound source.**

Keywords—sound localization, microphone array, biobot, search and rescue, neural engineering, cyberphysical network

I. Introduction

Advances in neural engineering have enabled direct control of insect locomotion [1], [2]. Our previous work has demonstrated the ability to control Madagascar hissing cockroaches in both open and maze-like environments [3]–[7]. These remotely controlled insect biobots can be used to localize surviving victims after natural disasters. Towards this goal, directional and omnidirectional acoustic sensors could provide an effective tool to detect help calls from victims buried under rubble, locate these victims by tracking the source of the sound, and establish a communication channel with first responders. Moreover, the addition of a small buzzer would enable biobotic agents know when they approach another agent. It has been shown that this information can be used to construct a topological map of an unknown environment [8].

Traditionally, sound localization techniques rely on an array of omnidirectional microphones and use beamforming or cross-correlation-based time delay to determine the direction of arrival of the sound source. However, these techniques typically require larger microphone arrays in order to maximize the delay experienced by the different nodes [9]. Other work [10] has demonstrated the feasibility of amplitude-based sound localization. However, this focused on large, high quality microphones operating a fixed distance. We expand on this work by significantly reducing the size of the microphone array using low-cost alternatives and introducing a variable distance to the sound source.

In a low-power system designed to be used on an insect biobot, both size and processing speed are limiting factors. With microphones spaced 1 cm apart, sound at one microphone would reach the others just 30 microseconds later. To avoid these limitations, we propose the use of an array of three unidirectional microphones spaced 120° apart. By examining the relative amplitude of each signal, the direction of arrival can be estimated. This allows the array to be smaller and reduces the computational requirements, making online, real-time processing feasible. This array would enable an insect biobot to automatically locate and approach a sound source. This could be useful in searching for victims of a disaster calling for help, or for calling the biobots to a base station for solar-powered charging [5].

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In addition, we propose a separate, single-microphone system capable of recording and streaming short bursts of audio. Such a system could be used for listening to sounds in a disaster environment. We also explore the feasibility of transmitting audio data wirelessly over a distributed ZigBee sensor network.

II. Microphone Array for Sound Localization

A. Sensor Array Design

To achieve insect biobot-based sound source localization, a backpack with a microphone array consisting of three unidirectional electret condenser microphones [11] was developed. As shown in Fig. 1, the microphones are spaced 120° apart on a circle of radius 1 cm. Each microphone unit has two pads for data and ground connection at the back. Connections are made to a small printed circuit board through soldering two pins. A thin double layer of sticky foam is used to fill the gap between the microphone and printed circuit board, thereby holding the microphone in place. Microphone circuit boards can then be inserted to connectors on the backpack.

The output of each microphone is fed to a preamplifier [12], as shown in Fig. 2. This low-voltage amplifier suppresses signals below the noise floor and limits signals above the

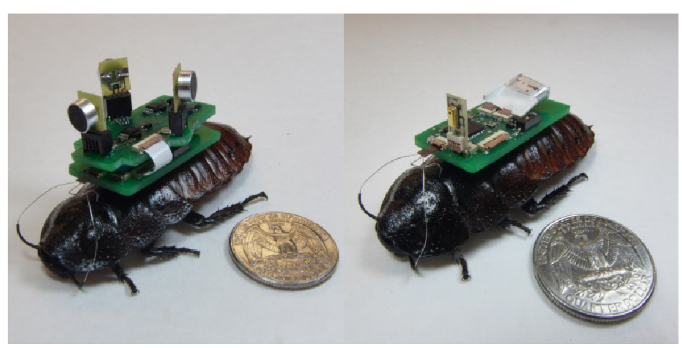

Fig. 1. Biobot backpacks with (left) omnidirectional microphone for audio recording and streaming, (right) Unidirectional microphone array for source localization

rotation point. For signals within its input range, it offers a variable compression ratio. For this application, a 1:1 compression ratio is used to maximize the dynamic range of the preamplifier output with respect to its input. The output of the preamplifier is then buffered and passed through an active second order Sallen-Key high pass filter with a cutoff frequency of 62 Hz, which removes the preamplifier bias and acts as a non-inverting half-wave rectifier. The audio is then amplified and passed through an active anti-aliasing filter with a cutoff frequency of 617 Hz before being sampled by the analog-to-digital converter (ADC) on a system-on-chip [13].

Data was sampled using a resolution of 14 bits at a rate of 1.25 kHz from each microphone. Data was transferred in 30 sample windows via direct memory access (DMA) to buffers where the sample with the maximum amplitude was extracted. This extracted maximum was transmitted wirelessly via ZigBee to a PC with an RF USB dongle [14] for further analysis. Data was transmitted in 90 byte chunks every 1.08 seconds.

B. *Microphone Characterization*

One challenge involved in localization using amplitude is the sensitivity to differences in the microphone response. For example, it was found that each microphone exhibited different polar directivity characteristics. Some had a high sensitivity while others had a wider, more omnidirectional response.

In order to account for these differences and to examine the effects of distance and incidence angle on the microphone response, the array was characterized in an anechoic chamber as shown in Fig. 3. The microphone array was mounted on a stepper motor with 1.8° steps. A speaker nearby played a 300 Hz sinusoidal calibration tone [15]. The stepper motor, controlled with a National Instruments DAQ device, was spun in 9° increments, pausing for 3 seconds each step to sample audio using the windowed, maximum amplitude approach described previously. Data was transmitted to a PC for further analysis. Representative results from a single trial are shown in Fig. 4. This procedure was repeated as the speaker distance was varied from 10 cm to 90 cm in increments of 10 cm. For each trial, the intensity of the sound at the array was measured using a portable sound meter.

C. *Sound Source Estimation*

For each microphone, angle, and intensity, a single value was obtained by taking the median value in the 3 second window. Local regression was used to fit a surface to the amplitude vs intensity and direction data for each of the three microphones. As expected, each surface had a peak at an increment of 120°. These calibration surfaces were precomputed for each microphone array. Surfaces from one array are shown in Fig. 5.

Once the surfaces were computed, they were evaluated on a mesh with a resolution of Δx=1° and Δy=0.32 dB. Given a new measurement from each microphone at an unknown direction and intensity, it can be compared with these precomputed values. For each of the three microphones (n=1, 2, 3), the error between the measured (M_{nxy}) and computed amplitude (C_{nxy}) at a particular location can be compared. The point on the mesh with the minimum error according to (1) is taken as the estimate for the direction and intensity.

$$\arg\min_{x,y} \sum_n \left(E_{nxy} - A_{nxy}\right)^2 \tag{1}$$

TABLE I. MICROPHONE ARRAY SPECIFICATIONS

	Microphone Array	**Single Microphone**
Power consumption (streaming)	63 mW	36 mW
Size (mm)	32.5 (l) × 25.0 (w) × 20 (h)	34.0 (l) × 22.5 (w) × 16.5 (h)
Weight (no battery)	3.4 g	1.6 g
Weight (with 20 mAh LiPo battery)	4.1 g	2.3 g
Sampling rate	1.25 kHz	4.00 kHz

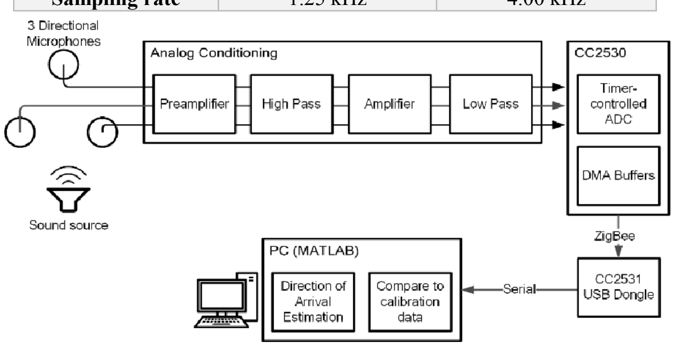

Fig. 2. Architecture of sound processing and localization pipeline

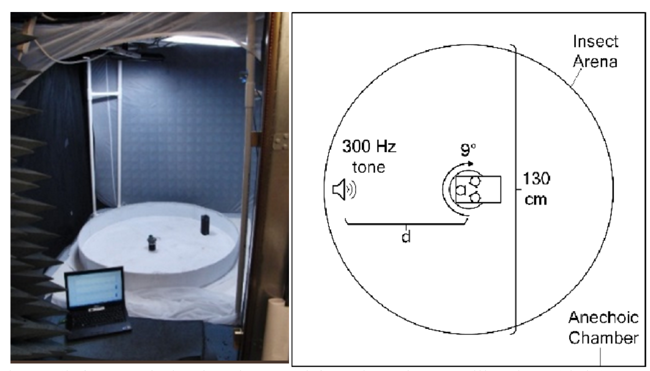

Fig. 3. (left) Anechoic chamber used for microphone calibration, (right) diagram of calibration setup

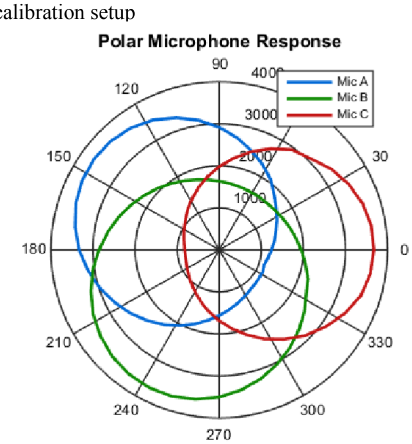

Fig. 4. Polar directivity plot for each microphone

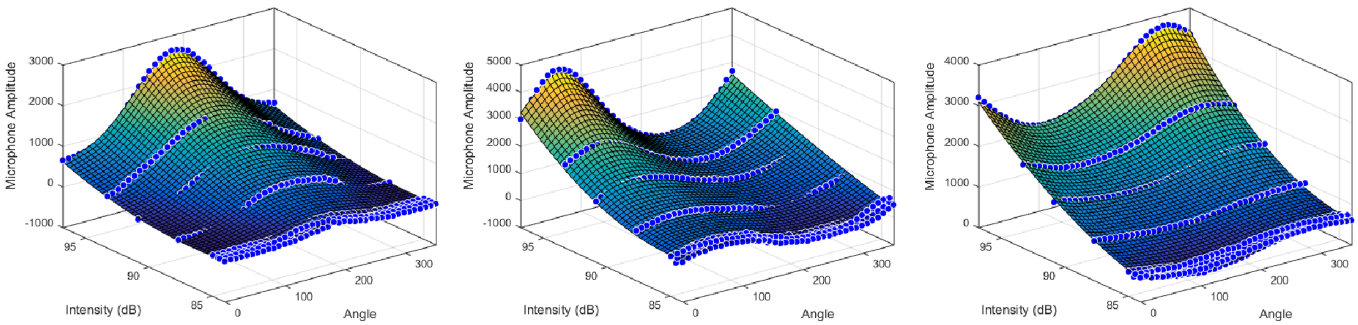

Fig. 5. Directional response of each microphone at varying sound intensity. Local regression was used to fit the surfaces to the collected data. The peak amplitude occurs when the source is directly in front of a particular microphone.

III. Steering Insects toward Sound

The source estimation technique can be applied in real time to data being streamed from the microphone array. MATLAB was used to compute the estimated direction of arrival every 720 ms. This estimated direction was transmitted over a local socket to the stimulation strategist software, RoachTrack [4], which was responsible for tracking, stimulus control, and data logging. When the insect's deviation from the estimated sound source exceeded 45°, a stimulus was issued to correct its path.

In order to evaluate the feasibility of this control scheme, the microphone array was tested on a low-cost robotic insect [16] controlled by an IR remote. These robots are imprecise and do not always respond in the same way to a command, making them a good simulation of an insect biobot. First, the robot was placed at a specified start location, 75 cm from the speaker. The initial orientation was varied from 0° to 315° in 45° increments and the robot was directed automatically toward the estimated sound source. The paths of these trials are shown in Fig. 6. Next, the robot was placed at various locations in the test arena, facing away, 75 cm from the speaker.

Finally, the microphone array was evaluated on a Madagascar hissing cockroach (*Gromphadorhina portentosa*) that had undergone surgical implantation of electrodes [3]. The insect biobot was placed in a start region, approximately 60 cm from the speaker in random orientations. A stimulus of five 30 ms pulses, with 50% duty cycle, was delivered at most every 400 ms in order to turn the insect. Sample paths of such a biobot is shown in Fig. 6. The average error between the true direction of arrival and the estimated direction was 27.3°, evaluated every 720 ms.

IV. Audio Transmission over Distributed Network

In addition to a three-microphone array for sound source localization, a separate backpack was constructed for use with a single, omnidirectional microphone. This was designed to record a single channel of audio and wirelessly transmit it to a base station over the ZigBee network either in bursts or real-time. The system-on-chip samples with a 14-bit resolution at 4 kHz and uses DMA to store the data in a 3.6 kB ring buffer. Transmitting the data at 9 kbps allows real-time streaming.

The transmission range of this circuit is determined by the transmit power of the CC2530 system-on-chip, which significantly influences power consumption. At its maximum transmission power, the range exceeds 75 m. However, by reducing the transmission power and introducing routing nodes within the network, data transmission can occur by hopping from one node to the next.

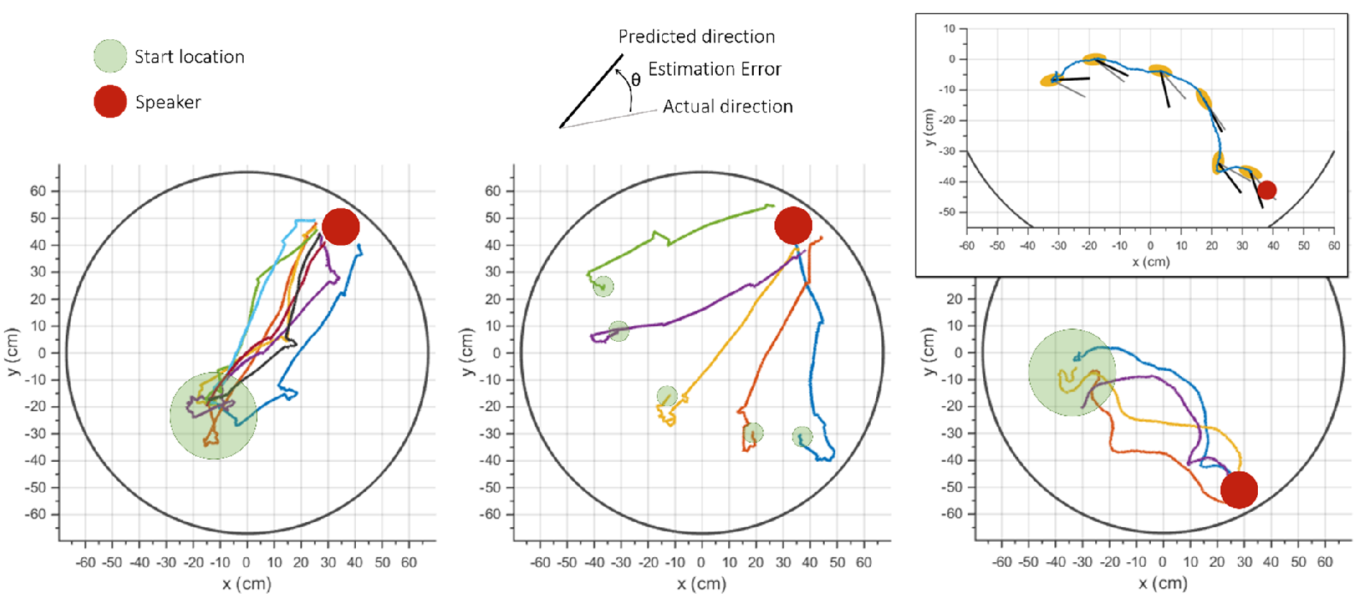

Fig. 6. Results of automated steering toward estimated sound source. (left) Robotic trial with varying orientations from same start location. (center) Robotic trial with varying start location and orientation away from speaker. (right) Insect biobot paths with random orientation and constant start location. (inset) analysis of predicted vs actual direction of arrival for a single insect trial. Videos of the trials can be found online [15].

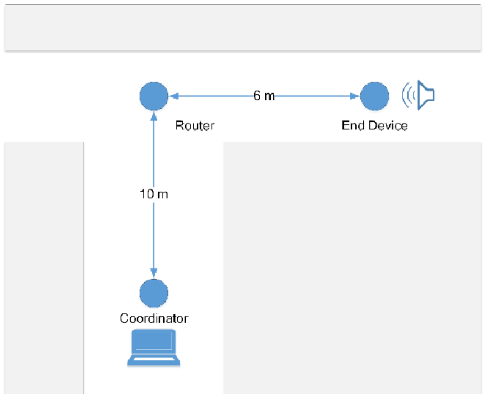

Fig. 7. Block diagram showing locations and distance of the end device from the coordinator separated by a wall with a router in between.

To demonstrate this, the sensor and coordinator device were separated by a wall as shown in Fig. 7. With a low transmit power, data communication would normally fail. The introduction of a router node enabled transmission in two hops, albeit with some packet loss. Samples of the recorded audio are available online [15]. Additional work remains to generalize this to a sensor network with more complicated network topologies.

V. Discussion

The current design of the microphone array requires remote processing. This aids in evaluation of the array and the control scheme, but requires significant wireless data transmission. An online algorithm would be more power efficient. The biggest challenge in realizing this goal is the calibration required for the microphones. The current method of source estimation requires an analysis of the pre-computed calibration surfaces. Improvements in microphone accuracy, or a generalized model of microphone response could enable online source estimation.

Another limitation of the current implantation of the microphone array is in the dynamic range of the audio signal. The systems can effectively localize sound within a 36 in radius. At further distances, additional amplification is required. A dynamic amplification may allow localization at further distances while preventing saturation at close range.

The degradation of audio quality in a multi-hop ZigBee network has been observed by others [17]. The use of compression schemes could improve quality, but this is difficult on such small sensors. Real-time streaming may be limited to only two hops before packet loss becomes unacceptable. For more than two hops, data could be transmitted in short bursts, with an acknowledgment scheme that ensures minimal packet loss.

VI. Conclusion

This work presents two low-power sensors for use on an insect biobot capable of localizing a sound source at close range and recording audio for transmission to a base station. The localization technique is shown to have an average accuracy of 27.3° while mounted on a moving biobot. The ability to steer agents toward a sound source was evaluated first on synthetic robots, then on insect biobots. Both were successfully navigated to the speaker when placed at various locations in a test arena, demonstrating the feasibility of such a localization scheme. We also demonstrate the ability of a second sensor to record and stream audio over a ZigBee sensor network. Although the addition of multiple hops degraded the audio signal, it increased transmission range and decreased power consumption. Such a sensor can be used to record bursts of audio and transmit them across a distributed network to aid first responders.

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