Microphone Arrays for Audio Enhancement

By James Hoder, ECE '19

Introduction

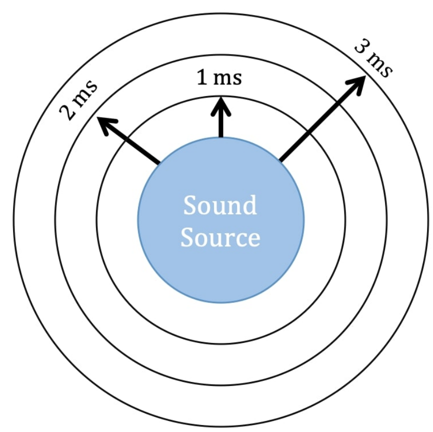
When capturing audio, nearby noise can often ruin a recording. The recording must be digitally altered to enhance the sound, remove noise, and save only the desired audio. In order to distinguish which parts of a recording are from the desired source, it is often necessary to take multiple recordings from different locations, and then compare the results. Afterward, the source of the desired audio and each source of noise can be identified so that only the unwanted sound is removed. A microphone array is a set of two or more microphones that are used in conjunction for this purpose of identifying each audio source.

Microphone Arrays

Human Ears: The First Microphone Array

The human auditory system allows a person to naturally discern the general direction of a noise source at any given time. Without thinking extensively, a person knows how to turn their head to listen to a voice, where to look when hearing something unexpected, and in which direction to move to avoid danger. The brain is capable of naturally identifying sound sources based on audio captured by the ears, because humans have one ear on each side of their head [1].

As shown in Figure 1, sound coming from a source arrives at different locations at different times, because it moves at a finite speed, 343 meters per second. Therefore, a sound arrives at a person's right ear either slightly before or after it arrives at a person's left ear, because they are slightly different distances from the source of the sound. This delay is known as time lag. The human brain calculates this time lag, and it is thereby able to determine the approximate direction of the source based on the orientation of the head.



.
Figure 1. An example of how distance causes a delay in time between different locations receiving the same source audio.

Microphone Technology

Microphones are used to simulate the sound recognition of an ear by digitally recording audio. Sound is a pressure wave that moves through the air. Just like a human ear has an eardrum that can sense the air pressure changing nearby, microphones have a thin diaphragm inside them that vibrates due to air pressure changes. As the diaphragm moves, a coil of wire, which is wrapped around a magnet, moves with it. The movement of the coil creates an electric signal identical to the audio it is recording. This electronic signal is then stored in a computer so that it can be analyzed. In a microphone array, the signal created by each microphone is stored so that they can all be compared.

Identifying Audio Sources

Microphone arrays function in nearly the exact same way as human ears. The multiple audio inputs to a system from the array give it multiple "ears" that allow the system to make informed conclusions about the origin of a sound. Typically, the microphones in a microphone array are in fixed positions, separated from one another on a single board by an exact, predetermined distance, much like a person's ears [2]. Each microphone is a different distance from the source and therefore experiences a different time lag, as shown in Figure 2. A system is thereby able to make precise calculations based on these time lag differences, and then determine the sound origin's location based upon those calculations. Just as the brain uses the ears to determine many qualities about a person's surroundings, a microphone array allows a system to identify unique qualities about its surroundings including sound-source location and distinct, noiseless sound qualities.

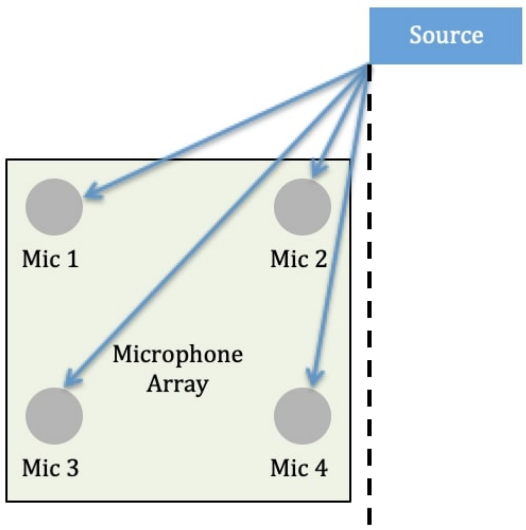


Figure 2. A depiction of the different distances of individual microphones in a microphone array to an audio source.

To calculate an audio source's location, firmware built into a microphone array first estimates the time lag between one primary microphone and each of the other microphones. For each calculation, the primary audio recording is held in a fixed position while the secondary recording is overlaid onto it and repeatedly shifted across it by a fraction of a second. As the secondary recording is shifted, the difference between the two signals is calculated at each point. When the difference between the two recordings is smallest, they line up most closely with one another in time. The time lag is determined as the amount that the secondary microphone's recording is shifted in order to line up both recordings.

After determining the time lag of each secondary microphone, the position of the audio source can be approximated using geometry. Using the speed of sound, the distance from each microphone to the source can be found from the time lag. As shown in Figure 2, there is only one possible source location for a set of distances that start at a single point and end at each of the microphones. Therefore, if the time lag estimates are correct, the source location can be very accurately determined. By repeating this process and searching for multiple different ways that the audio recordings line up, a microphone array can locate and identify multiple different sources and their locations. The audio from specific noise sources can therefore be identified and removed, if desired.

Applications

Source Localization

Microphone arrays are typically used for localizing the source of a sound. The differences in audio readings among the numerous microphones allows for the precise calculation of source direction and location, as stated above. By finding the direction of a source of a nearby sound, an electronic system can make informed decisions about how to interact with its surroundings, just like humans do.

Noise Reduction and Signal Enhancement

The differences in time lag between recordings from the audio samples can also be used to differentiate between different sounds being heard by the microphone array. Each microphone picks up sources of noise with varying delays and volumes. By comparing the differences in noise content among the microphone recordings, specific sounds can be isolated, then amplified or removed. Unwanted sounds can be strategically removed, almost entirely, and the enhanced audio to be processed can be more clearly and accurately be analyzed.

Drawbacks

While microphone arrays do allow for enhanced awareness of a system, they also create more room for error and over-complication. In theory, the process discussed above perfectly accomplishes the desired applications. However, this outcome is rarely the result, because sound often has so much inconsistent noise that the process of comparing signals can become very difficult and require estimation, which can cause error. Therefore, there are a number of drawbacks to consider and address when using microphone arrays.

Increased Processing Time

If trying to process audio in real time, the comparison and math necessary to effectively use a microphone array may be too slow to be useful. Shifting one recording across another by small time increments and calculating the difference at each point to determine time lag requires a very large number of small operations, so it can take a long time, especially if using a large number of microphones. While increasing the number of microphones can improve accuracy, it also increases computational time. The correct balance between the number of microphones and the increased computational time must be found, depending on the acceptable processing time for a given application.

Limitations in Distinguishing Differences

If the sound coming from a noise source is similar in pitch or tone to the sound coming from the desired source, it could make distinguishing the difference between the two sources difficult. In this case, it may be impossible to remove strongly interfering noise. In order to fix this problem, if certain pitches or tonal qualities are expected in the desired signal, those specific parts of the recording can be amplified prior to comparing the recordings from the multiple microphones. This process would create a greater difference between the desired signal and the noise, even if they share some attributes.

Array Placement

The placement of the array must be strongly considered for accurate results in a 2D plane. Just as the positioning of a person's head affects a person's perceptions, the position and orientation of a microphone array will strongly affect a system's ability to find the correct direction of the origin of the sound. If a microphone array is not sitting parallel to the plane in which it should read direction, then its chance of finding the proper distance to the source is reduced, because there is a less drastic difference in position between the two microphones in the relative plane. If a microphone array is not positioned properly, it may prove to be ineffective.

Conclusion

Microphone arrays are extremely useful sets of auditory sensors for use in complex systems. A sound source can be identified and located by calculating time lag amongst multiple audio signals, then using geometry to relate their time lags to distances from a single origin. This information can be used to reduce noise in a recording, so that specific sounds and sources can be isolated. Microphone arrays have many important considerations and varied applications, and therefore must be carefully designed and programmed for any system that uses them. Microphone arrays provide a system with a great deal of information, which can be used to enhance audio quality or to better understand a system's environment, but they present some considerable challenges that must be overcome to process that information.

References

[1]"Direction of sound - direction of sound waves - hear-it.org", *Hear-it.org*, 2018. [Online]. Available: https://www.hear-it.org/The-direction-of-sound.

[2]"ReSpeaker 4-Mic Array for Raspberry Pi", *Wiki.seeedstudio.com*, 2018. [Online]. Available: http://wiki.seeedstudio.com/ReSpeaker_4_Mic_Array_for_Raspberry_Pi/.

[3]"How do microphones work? | Types of microphone compared", *Explain that Stuff*, 2018. [Online]. Available: https://www.explainthatstuff.com/microphones.html.

[4]J. Valin, F. Michaud, D. Létourneau and J. Rouat, "Robust Sound Source Localization Using a Microphone Array on a Mobile Robot", *Jmvalin.ca*, 2018. [Online]. Available: https://jmvalin.ca/papers/iros.pdf.

[5]J. Fan, "Localization Estimation of Sound Source by Microphones Array", *2010 Symposium on Security Detection and Information Processing*, 2018. [Online]. Available: https://ac.els-cdn.com/.