

Quantifying acoustic array performance

More than counting the microphones

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Introduction

In the past decades, technology—ranging from computer performance [1] to the number of pixels in a digital camera [2]—has demonstrated an impressive growth. This is a consequence of Moore's law [3], which is both the observation and prediction that the number of transistors inside an integrated circuit increases exponentially over time. A similar trend can be observed for acoustic arrays: due to the advances in integrated circuits, digital microphones based on micro-electro-mechanical systems (MEMS) technology [4] have emerged as an economical, miniaturized alternative to analog condenser microphones for application in arrays.

This has enabled today's microphone arrays to have 32–128 microphones—or even more: the largest microphone array in the world, according to the Guinness world record [5], consists of an impressive 4096 microphones. However, it is important not to get dazzled by these numbers: similar to the Megahertz myth for processors or the megapixel myth for photography, one could formulate a microphone myth for acoustic arrays: contrary to what one might be tempted to believe, just the number of microphones does not tell the whole story of acoustic performance. In particular, the placement of the microphones is an equally important factor due to the laws of physics which ultimately limit the achievable array performance.

This white paper gives an overview of the effects of the number and placement of microphones on array performance and provides the basic tools for quantifying it.

Averaging reduces noise

Each microphone inevitably introduces some noise in its measurements of the sound pressure: due to production tolerances, the sensitivity varies slightly from microphone to microphone [6], and the electronics of the microphone also introduce self-noise. When a (quiet) sound is eclipsed by this noise, the corresponding sound source cannot be detected. Because sound decays as it travels away from the source, this can also limit the detection range of the acoustic array. Luckily, a well-known phenomenon in signal processing is that averaging measurements of multiple microphones tends to reduce the noise:

$$\text{Noise reduction (in dB)} = 20 \log_{10}(\sqrt{\text{Number of microphones}})$$

This formula means that each time the number of microphones is doubled, noise is reduced by 3 dB, which is barely noticeable to the human ear under normal circumstances [7]. Thus, while a modest acoustic array will greatly outperform a single microphone (e.g. an array of 64 microphones reduces noise by 18 dB), returns diminish as the number of microphones is further increased (e.g. the difference between arrays of 128 and 64 microphones is just 3 dB). Also, at some point, noise that is common to all microphones (e.g. from the power supply) will start to dominate, since it cannot be reduced through averaging. Finally, more microphones require more data processing which either forfeits battery life and portability, or requires compromises in terms of display frame rate or resolution.

To summarize: while increasing the number of microphones reduces noise, at some point the returns diminish and do not outweigh the disadvantages. With today's technology we believe the optimum is around 64 microphones.

Beamforming and wave propagation

Just averaging the microphone measurements does not enable visualization of sound projected over a camera image. Instead, a beamforming algorithm [8] must be used. Beamforming combines the signals of all microphones in the array so that contributions from sources at particular angles experience constructive interference while others experience destructive interference.

The underlying physics which enable beamforming are established by the wave equation.[9] In particular, sound propagates through the air with a fixed speed:

$$\text{Speed of sound} = 343 \text{ m/s.}$$

This means that sound does not only have a frequency but also a wavelength:

$$\text{Wavelength (in m)} = \frac{\text{Speed of sound (in m/s)}}{\text{Frequency (in Hz)}}$$

When one would freeze time, the wavelength is the physical length of the sound wave in the direction of its propagation.

Beamforming exploits the fact that the different microphones in an acoustic array measure different points of this wave. Therefore, a proper ratio between the distances between the microphones of the array and the wavelength is crucial for good acoustic performance. At low frequencies, the wavelength is large (e.g. 3.4 m at 100 Hz) and the array benefits from large distances between the microphones. At high and ultrasonic frequencies, the wavelength is small (e.g. 17 mm at 20 kHz) and the array benefits from small distances between the microphones.

Array diameter determines low frequency resolution

The largest inter-microphone distance corresponds to two microphones on opposite edges of the acoustic array. Thus, the diameter of the array (Fig. 1) is related to the resolution at low frequencies: when the array is too small, separate acoustic sources will blur together in the projected sound image. An analogy can be found in astronomy, where a telescope with a larger diameter (a.k.a. aperture) is able to resolve smaller details.

The Rayleigh criterion quantifies this effect by approximating the minimum angle between two acoustic sources below which they will blur together:

$$\text{Angle (in deg)} = 69.88 \times \frac{\text{Wavelength (in m)}}{\text{Array diameter (in m)}}$$

Of course, increasing the array diameter greatly reduces portability. Luckily, many interesting sound sources such as air leaks and electric discharges mainly emit sound at high frequencies. For these applications a compact acoustic array does not significantly impact performance.

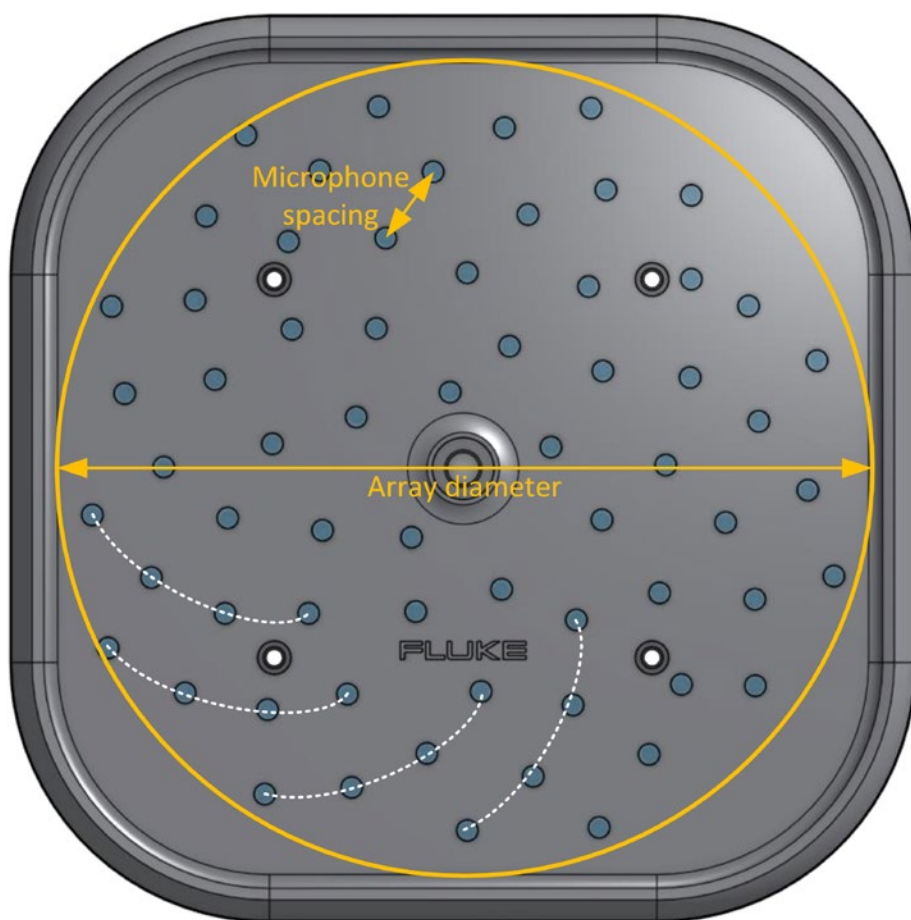


Fig 1: Array diameter and microphone spacing are two important parameters of an acoustic array. Large arrays have better resolution at low frequencies, and arrays with closer microphones ameliorate aliasing artifacts at high frequencies. Performance at high frequencies can be further improved by placing the microphones in a spiral (indicated by the white dashed lines).

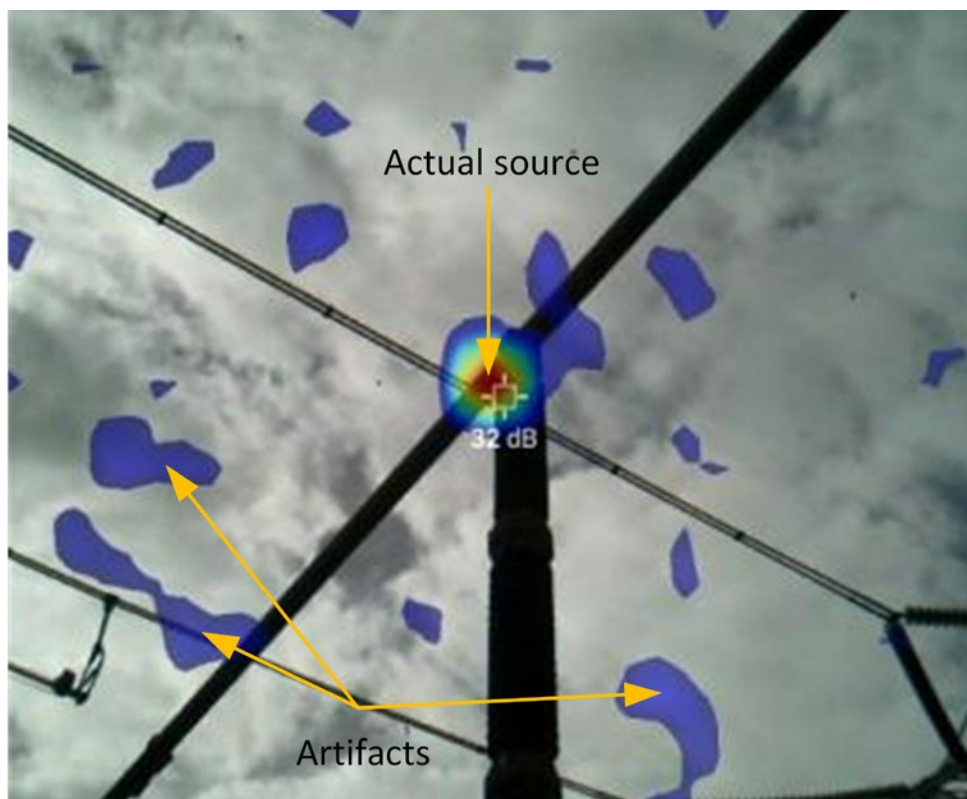


Fig 2: Example of artifacts in the projected sound image.

Microphone spacing determines high frequency artifacts

The smallest inter-microphone distance corresponds to two adjacent microphones (Fig. 1). Thus, microphone spacing is related to performance at high frequencies: when the spacing is too large, the sound waves cannot be uniquely resolved which results in aliasing (a.k.a. strong side-lobes). This manifests as artifacts or 'ghost' sources (which are not really there) in the projected sound image (Fig. 2). An analogy can be found in photography, where Moiré patterns can occur if the pixels of an image are too large. If the microphones are positioned on a regular grid, the frequency above which aliasing can occur is quantified using the Nyquist criterion:

$$\text{Aliasing frequency (in Hz)} = 0.5 \times \frac{\text{Speed of sound (in m/s)}}{\text{Microphone spacing (in m)}}$$

To reduce microphone spacing, either the diameter of the array must be reduced (which impacts lowfrequency resolution) or the number of microphones must be increased (which impacts battery life); both have undesirable effects.

Luckily, it turns out there is a smarter solution: Nyquist criterion can be surpassed by breaking the regularity of the microphone grid. In signal processing, this phenomenon is called sparse sampling (a.k.a. compressive sensing), because an irregular grid can be constructed by sampling points (and discarding others) from a much finer regular grid. Of course, at some frequency even sparse sampling will break down, but when implemented correctly this frequency can be pushed very far into the ultrasonic range.

There are many ways to choose a suitable irregular grid.[10] One interesting solution is inspired by nature: Fermat's spiral which describes the distribution of seeds in a sunflower head. This 'sunflower spiral' distributes the microphones efficiently and almost evenly over the array surface, but in such a way that the distances between pairs of microphones vary slightly (Fig. 1). This allows a sunflower array to greatly alleviate aliasing artifacts compared to a regular array with the same number of microphones, or conversely, provide the same acoustic performance with less microphones and hence longer battery life.

Conclusions

We have given an overview of the considerations related to the number and placement of microphones which impact array performance and provided the basic tools for quantifying it. A compact sunflower array with around 64 microphones such as the Fluke ii900 provides an excellent balance between acoustic performance—especially for sources caused by air leaks or electric power discharges—and usability considerations such as battery life and portability.

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05/2021 210512-en

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