soundskrit
Hear the impossible.

Introduction

Soundskrit microphones are a directional alternative to omnidirectional micro-electro-mechanical-system (MEMS) microphones for consumer electronics, enabling clear voice pickup by reducing background noise with an inherently directional design. While system designers are currently accustomed to using arrays of multiple omnidirectional microphones to mimic directionality, inherently directional microphones have several key attributes which differentiate them and require separate design considerations. When designing with directional microphones, it is necessary to understand how the behavior of these microphones deviates from traditional omnidirectional microphones and ensure that the associated performance metrics account for this accurately. The purpose of this document is to highlight unique characteristics of Soundskrit's directional microphones as well as describe appropriate methods for characterizing and evaluating their performance.

The Soundskrit Microphone

Both traditional MEMS omnidirectional microphones and Soundskrit directional microphones consist of a MEMS transducer and an application specific integrated circuit (ASIC) mounted on a printed circuit board (PCB) and are enclosed by a metal lid, as illustrated in Figure 1.

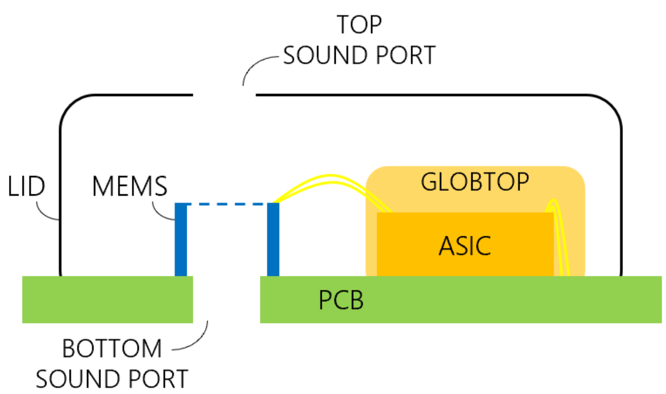

Figure 1: Cross-section schematic of Soundskrit MEMS microphone

Unlike traditional MEMS omnidirectional microphones, the Soundskrit microphone has several unique characteristics. Due to the directional nature of the Soundskrit microphone, the output of the microphone depends on the direction from which a captured sound wave originates. Additionally, the microphone has two sound ports, a top sound port in the metal lid and a bottom sound port in the PCB of the microphone. Depending on the distance of the sound source, and the nature of which the microphone is assessed, the sensitivity, frequency response and other performance characteristics may change.

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Average Directivity Index

The sensitivity of a directional microphone will change depending on the angle at which the sound wave originates from. Figure 2 illustrates an example of a sound wave arriving at a Soundskrit microphone with an angle of Θ.

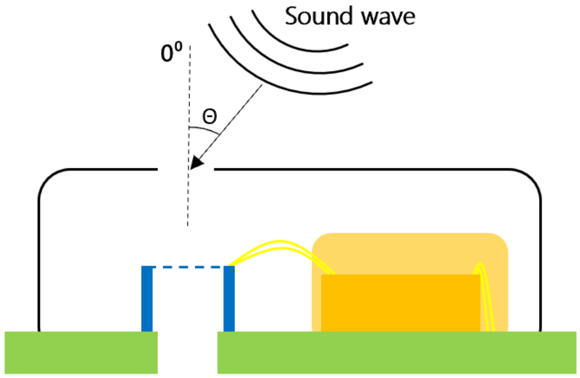

Figure 2: Sound wave approaching a Soundskrit microphone from angle Θ

The sensitivity of the Soundskrit microphone with respect to the incoming angle of a sound wave can be represented as a dipole pattern, or “figure-8”. In contrast, the typical omnidirectional microphone maintains an equal response regardless of the incoming direction of the sound. Representations of these polar patterns are in Figure 3 below.

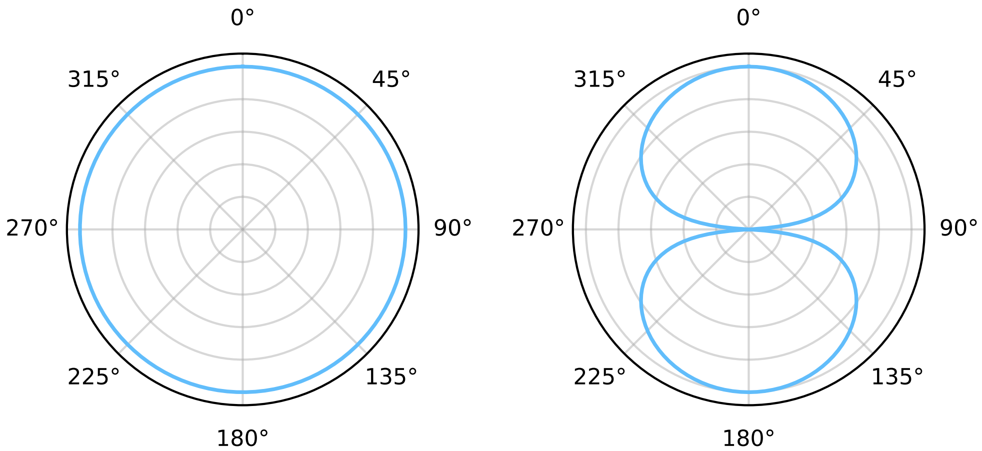

Figure 3: Polar pattern of an omnidirectional microphone (left) and Soundskrit microphone (right)

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When evaluating the performance of directional microphones, or microphone arrays, a metric known as the **directivity index** (DI) is used. The directivity index measures the ratio of the microphone output for a sound positioned directly in front of the microphone (θ = 0º) versus sound with the same amount of total acoustic power coming from all directions equally. The directivity index is calculated with the equation below:

$$DI = 4\frac{amplitude(\theta = 0)^2\left[\frac{V^2}{Pa^2}\right]}{\int_0^{2\pi} amplitude(\theta)^2\left[\frac{V^2}{Pa^2}\right]|sin\,\theta|d\theta}$$

Equation 1: Directivity index

Because omnidirectional microphones measure sound equally from all directions, they have a directivity index of 0 dB. In contrast, a microphone with an ideal dipole directionality pattern has a directivity index of 4.8 dB. Practically speaking this means that if a person is speaking in front of a dipole microphone in an environment with diffuse noise, or noise that is propagating equally in all directions such as babble noise or reverberation, the microphone will pick up the person's speech with 4.8 dB greater sensitivity than the diffuse background noise.

Engineers commonly attain directionality using arrays of multiple omnidirectional microphones. While these arrays can approximate a dipole polar pattern, their directionality changes depending on the frequency. However, the directivity index as described above is calculated only for a specific frequency. This single frequency DI does not accurately represent the directivity of an array across the frequency spectrum it is expected to record. To measure the directivity across the full audio spectrum (20 Hz to 20 kHz), the **average directivity index** ($\overline{DI}$) is a logarithmically weighted average of the directivity index at each frequency from 20 Hz to 20 kHz. If $DI[k]$ is an array with index k storing the DI for each frequency, the average directivity index can be calculated as shown below.

$$\overline{DI} = \frac{\sum_{k=1}^{N} W[k]\,DI[k]}{\sum_{k=1}^{N} W[k]}, \qquad W[k] = \log_{10}\frac{k+0.5}{k-0.5}, \qquad N = \text{length of } DI[k]$$

Equation 2: Average directivity index

The average directivity index provides a picture of a microphone's directionality across the audible spectrum. To demonstrate the difference across frequencies, Figure 4 shows the directivity index versus frequency for Soundskrit's SKR0400 and omnidirectional arrays of two microphones with different spacings. For the arrays, the outputs of the two omnidirectional microphones with the specified spacing were measured in a differential configuration, meaning one signal is subtracted from the other, to approximate a dipole directionality pattern to match the SKR0400.

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As shown in Figure 4, the SKR0400 exhibits a consistent directivity index across the frequency range. An ideal dipole microphone has a DI of 4.8 dB, which the SKR0400 matches across the audible frequency range. In contrast, typical omnidirectional microphone arrays with 40 mm spacing are not able to maintain directionality across the audible frequency range. This is reflected in a lower average directivity index of 2.2 dB, as opposed to the 4.8 dB of the SKR0400. As seen in Figure 4, the only way for the omnidirectional microphone array to improve its average directivity index is by reducing the spacing between the microphones to about 10 mm. However, as will be described later, this comes at the cost of a significant reduction in SNR. Note that the directivity index of the omnidirectional array may drop off at low frequency if the omnidirectional microphones are not perfectly matched, as seen with the measurement from the 40 mm array, which is not an uncommon phenomenon when dealing with typical microphone arrays using MEMS microphones with standard tolerances.

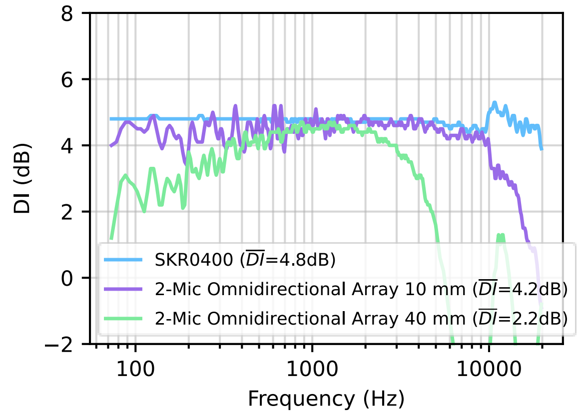

Figure 4: Directivity index vs. frequency of a Soundskrit microphone and an omnidirectional array

The Proximity Effect

When looking at the sensitivity of a directional microphone, it is important to consider the distance of the sound source *d*, as illustrated in Figure 5.

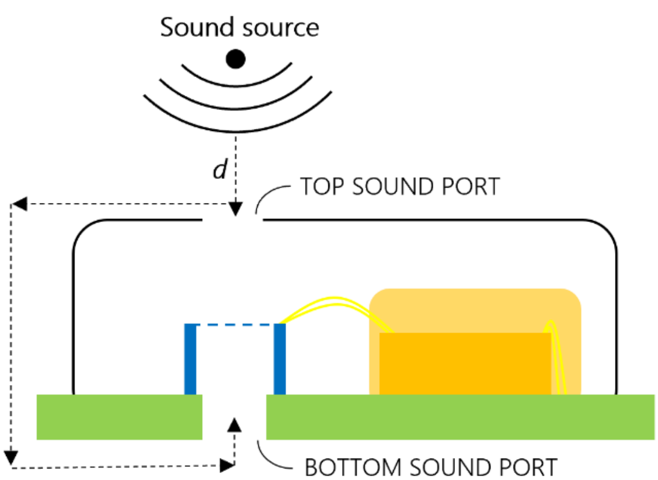

Figure 5: Sound source at a distance d from the microphone

When a sound source generates a sound wave that travels towards a directional microphone, it will first arrive at the microphone's top sound port with a certain acoustic pressure and then reach the microphone's bottom sound port with a different acoustic pressure. This difference in acoustic pressure causes fluctuations of the air inside the microphone package and excites the MEMS transducer. The distance the sound wave travels between the top sound port and bottom sound port is referred to as the **acoustic path length** and will be discussed further in the next section.

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Directional microphones exhibit a phenomenon known as the **proximity effect**. The proximity effect occurs when a sound source is close to a directional microphone, and manifests as an increase in low frequency sensitivity. This phenomenon is related both to the frequency of the sound and the microphone's distance from the sound source. Close to the source, the wave travels as a spherical wave while far away, it travels as a plane wave. The transition between a spherical wave and plane wave occurs when the distance from the source approaches the sound's wavelength, which is inversely proportional to the frequency. Because low frequency sounds have longer wavelengths, they travel as spherical waves for further distances.

A directional microphone responds to air motion due to the pressure difference across its top and bottom sound ports. In a spherical wave, the sound pressure decays faster with distance than it does with a plane wave. So, a directional microphone sees a larger pressure difference across its ports when exposed to a spherical wave. This leads to the sensitivity boost for low frequency sounds near the microphone. Figure 6 shows a graph of the boost in sensitivity a Soundskrit microphone sees as a function of distance *d*.

As shown in Figure 6, the proximity effect is most prominent when the microphone is close to the sound source. For example, in products such as boom microphones, earbuds, and smart glasses, where the microphone is about 10cm or less from the user's mouth, the microphone can see a noticeable boost in the low frequency sensitivity. This is desirable as it gives a full vocal sound that is reminiscent of radio voices. When the microphone is placed about 1 m or more away from the user's mouth, the proximity effect is mostly negligible as the only frequencies that still propagate as a spherical wave and create a boost in sensitivity are below 100 Hz, where there is no significant speech energy. It should be noted that Figure 6 shows the boost in sensitivity for a directional microphone (with a dipole beam pattern) with an acoustic path length of 10 mm. For different acoustic path lengths, the boost in sensitivity may vary from those values depicted in Figure 6.

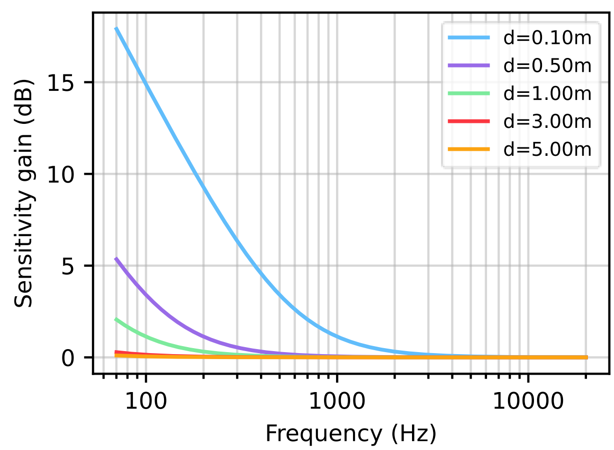

Figure 6: Soundskrit microphone sensitivity as a function of distance from the sound source

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Changing Acoustic Path Length

A directional microphone responds to air motion driven by the pressure difference across its inlet and outlet ports. When an acoustic wave travels across its ports, both the amplitude and phase of the pressure at each port determine the pressure difference. The proximity effect is a result of the pressure difference caused by the amplitude decay of the acoustic wave as it travels in a spherical manner (see previous section). In addition to the amplitude difference, the phase difference is also an important contributor to the pressure difference. In the far field (i.e., the microphone is subject to a plane wave), the amplitude difference is negligible (no proximity effect), and the phase difference of the acoustic wave is the dominant contributor to the pressure difference.

The phase difference is determined by the **acoustic path length**, as shown in Figure 7. The acoustic path length is the distance that an acoustic wave must travel from one sound port to the other. When the microphone is mounted on a PCB, the PCB will change the acoustic path length between the top and bottom sound port of the microphone, since a sound wave will have to wrap around the PCB to get from one sound port to the next. The acoustic path length depends on the width and thickness of the object (i.e., a PCB) the microphone is mounted on. When integrated inside of a product, the same principle applies.

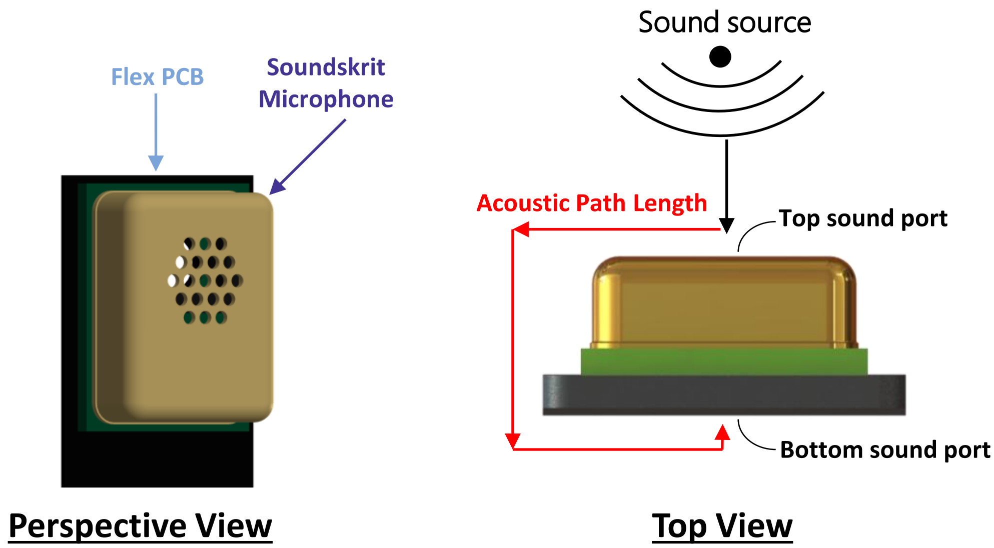

Figure 7: Perspective view (left) and top view (right) of a Soundskrit microphone mounted on a flex PCB for testing

As the acoustic path length increases, so does the phase difference and thus pressure difference seen by the two sound ports. This causes a sensitivity gain at all frequencies. Inversely, as the acoustic path length decreases, the sensitivity decreases accordingly. As the acoustic path length approaches the wavelength of the incoming sound, the response of the microphone may begin to degrade. A 20 kHz sound has a corresponding wavelength of 17 mm. Thus, to maintain ideal directionality up to 20 kHz, acoustic path lengths above 17mm should be avoided. However, many applications do not require ideal directionality all the way up to 20 kHz, and so longer acoustic path lengths can be used.

When a sound wave travels around the microphone and PCB, there are many paths it can follow. If the lengths of these paths are not equal (i.e., the PCB is not symmetric), then the acoustic path length is not necessarily defined as depicted in Figure 7. For example, if the acoustic path length around the

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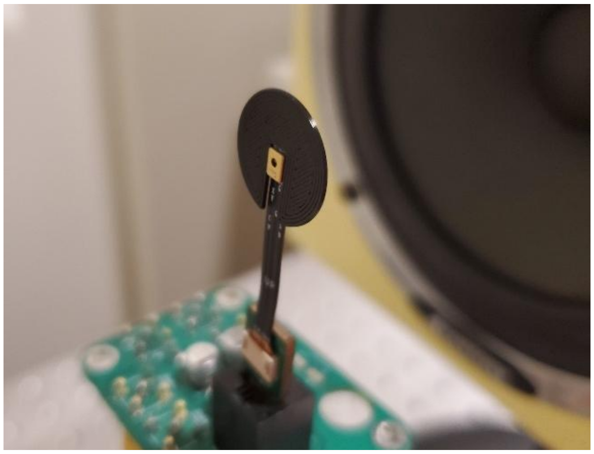
side of the microphone is different than around the top of the microphone, then the effective acoustic path length seen by the microphone will be in between the two lengths.

Figure 8 below shows the gain in the microphone sensitivity as a function of acoustic path length (when the microphone is subject to a plane wave). These measurements were taken with the microphone mounted on a thin circular disk of different sizes. Soundskrit microphones are characterized by an acoustic path length of 10 mm. The gain in Figure 8 is shown relative to 10 mm and follows the equation:

$$Gain = 20 * \log_{10} \frac{acoustic\ path\ length}{10mm}$$

Equation 3: Gain due to acoustic path length

The gain shown in Figure 8 is only applicable for scenarios where the acoustic path length is increased by using a wider PCB or supporting structure like the circular disk shown. It is assumed that the air volume in front of or behind the microphone is not increased. In other words, no long acoustic channels (or tubes) are coupled to the front or back sound ports of the microphone. Doing so may cause unwanted resonances at the microphone and the gain relationship between acoustic path length and sensitivity may deviate from that shown in Figure 8.



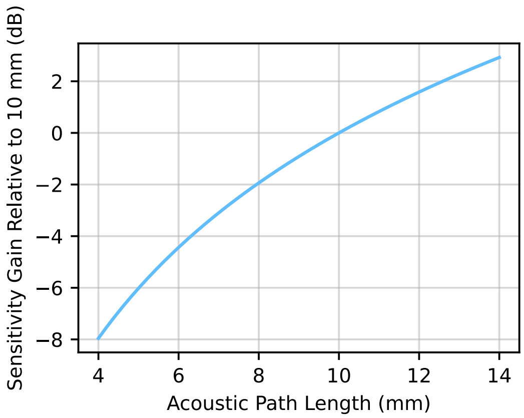


Figure 8: Soundskrit microphone sensitivity as a function of acoustic path length

A sensitivity gain has two important effects on the microphone's performance. First, as the self-noise of a microphone is constant, a gain in microphone sensitivity leads to an equivalent increase in SNR. Second, a gain in microphone sensitivity corresponds to a reduction in the acoustic overload point by an equivalent number of decibels. Thus, the microphone performance can be altered by choosing the appropriate acoustic path length when designing the microphones into an end-product. In practice, the acoustic path length seen by the microphone will rarely be less than 10 mm when accounting for the surrounding plastics of the enclosure.

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Calculating Signal-to-Noise Ratio

The **signal-to-noise ratio** (SNR) of an omnidirectional microphone is typically calculated by dividing the 1 kHz sensitivity of the microphone by the integrated noise across the audio spectrum. This works for omnidirectional microphones because most exhibit a flat frequency response across the range of audible frequencies. However, since the frequency response of a directional microphone is not flat, the SNR calculation must be modified accordingly to provide an accurate metric of performance. Instead of only using the 1 kHz sensitivity, the electrical noise of the microphone at each frequency (units of V^2/Hz) must be divided by the corresponding sensitivity squared at each frequency (units of V^2/Pa^2) to obtain the input referred acoustic noise at each frequency (units of Pa^2/Hz). Then, the A-weighted acoustic noise is integrated over the desired audio bandwidth, $f_1 - f_2$, and converted to an equivalent sound pressure level (dBA SPL). Finally, the SNR is calculated by subtracting the equivalent input referred noise from 94 dB SPL. In the case of the SKR0400, the equivalent input referred noise is 30.5 dBA, so the final SNR is 63.5 dBA. The equation for the calculation is shown below:

$$SNR = 94 - 20\,log_{10}\left(\frac{1}{P_{ref}^2[Pa^2]}\int_{f_1[Hz]}^{f_2[Hz]}\frac{noise\left[\frac{V^2}{Hz}\right]}{sensitivity\left[\frac{V^2}{Pa^2}\right]}A_w df[Hz]\right)$$

Equation 4: SNR calculation

To calculate the microphone SNR across the entire audible frequency range, f_1 = 20 Hz and f_2 = 20 kHz. In some applications, the full 20 kHz bandwidth may not be required. In this case, one simply integrates the input referred acoustic noise across the desired audio band.

To obtain a dipole directionality pattern with omnidirectional microphones, a differential array will subtract the output of two microphones with a given spacing. The spacing between the two omnidirectional microphones is the equivalent of a directional microphone's acoustic path length. To compare the performance of a directional microphone to an omnidirectional microphone array, the acoustic path lengths of both systems must be set to the same value. Figure 9 compares the SNR of the SKR0400 to an array using 70 dBA SNR omnidirectional microphones.

The sensitivity of the omnidirectional microphone array is low because of the small pressure difference seen between the two microphones. The MEMS transducers in typical omnidirectional microphones are not designed to react to such small pressure differences, resulting in a significant SNR drop, as shown in Figure 9. An array of two 70 dBA omnidirectional microphones has a 53 dBA SNR when spaced 10mm apart to create a dipole pickup pattern, 17 dBA loss in SNR. For the same acoustic path length (assuming it is not made excessively long), the SKR0400 always maintains 10.5 dBA higher SNR than the omnidirectional microphone array.

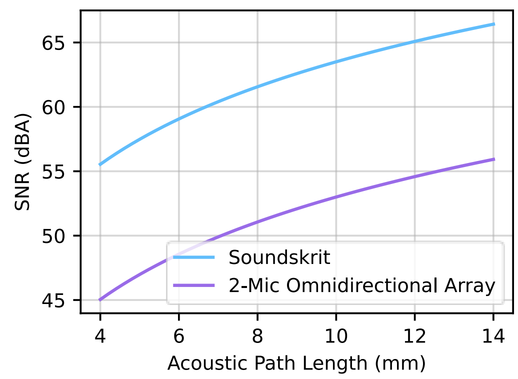

Figure 9: Soundskrit microphone sensitivity as a function of acoustic path length

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Calculating Total Harmonic Distortion

The **total harmonic distortion** (THD) of a microphone is a measurement of the harmonic distortion present in the signal captured by the microphone. It is calculated by playing an acoustic sine wave at a specific frequency and dividing the sum of powers of the harmonic components of the captured signal to the power of the fundamental frequency. The response of a microphone with a non-flat frequency response must first be equalized/flattened (see section Creating a Flat Frequency Response) before THD is calculated. This allows the harmonic components of the captured signal to be properly compared to the fundamental.

Creating a Flat Frequency Response

Designers often want their system to exhibit a flat frequency response to record a signal without introducing tonality from the microphone. Because the frequency response of a directional microphone is not flat, one can use a simple **equalization** (EQ) filter in software to flatten out the response. The sensitivity can be adjusted to match at all frequencies using common filters. An example of an un-equalized and equalized microphone signal is shown in Figure 10 below.

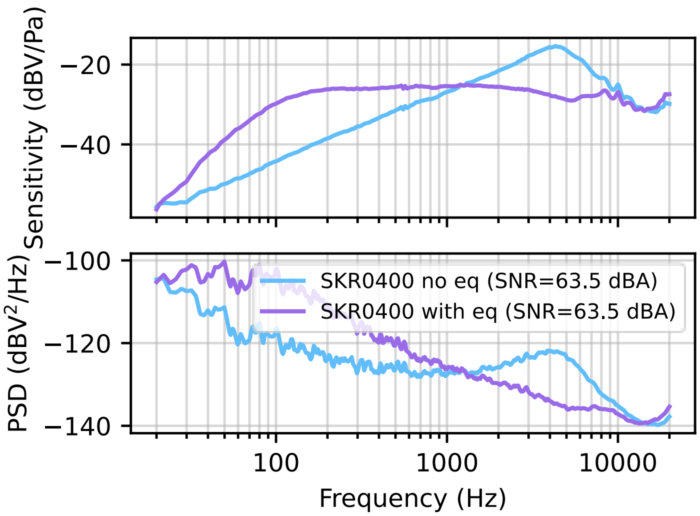

Figure 10: Un-equalized and equalized microphone sensitivity

As shown, the sensitivity response of the microphone can easily be equalized to a flat response or any other desired shape. In some applications, it may be desirable to implement a low frequency roll off below 100 Hz in the microphone response since the speech energy below 100 Hz is limited. Because equalization amplifies or attenuates both the microphone sensitivity and noise proportionally, it does not impact the SNR of the microphone. The input referred acoustic noise of the microphone never changes. For applications where the microphone is placed close to the user such as in the boom arm of a headset, the proximity effect provides a natural equalization and flattening of the response at low frequencies. So, software may only need to flatten out the high frequencies of the signal.

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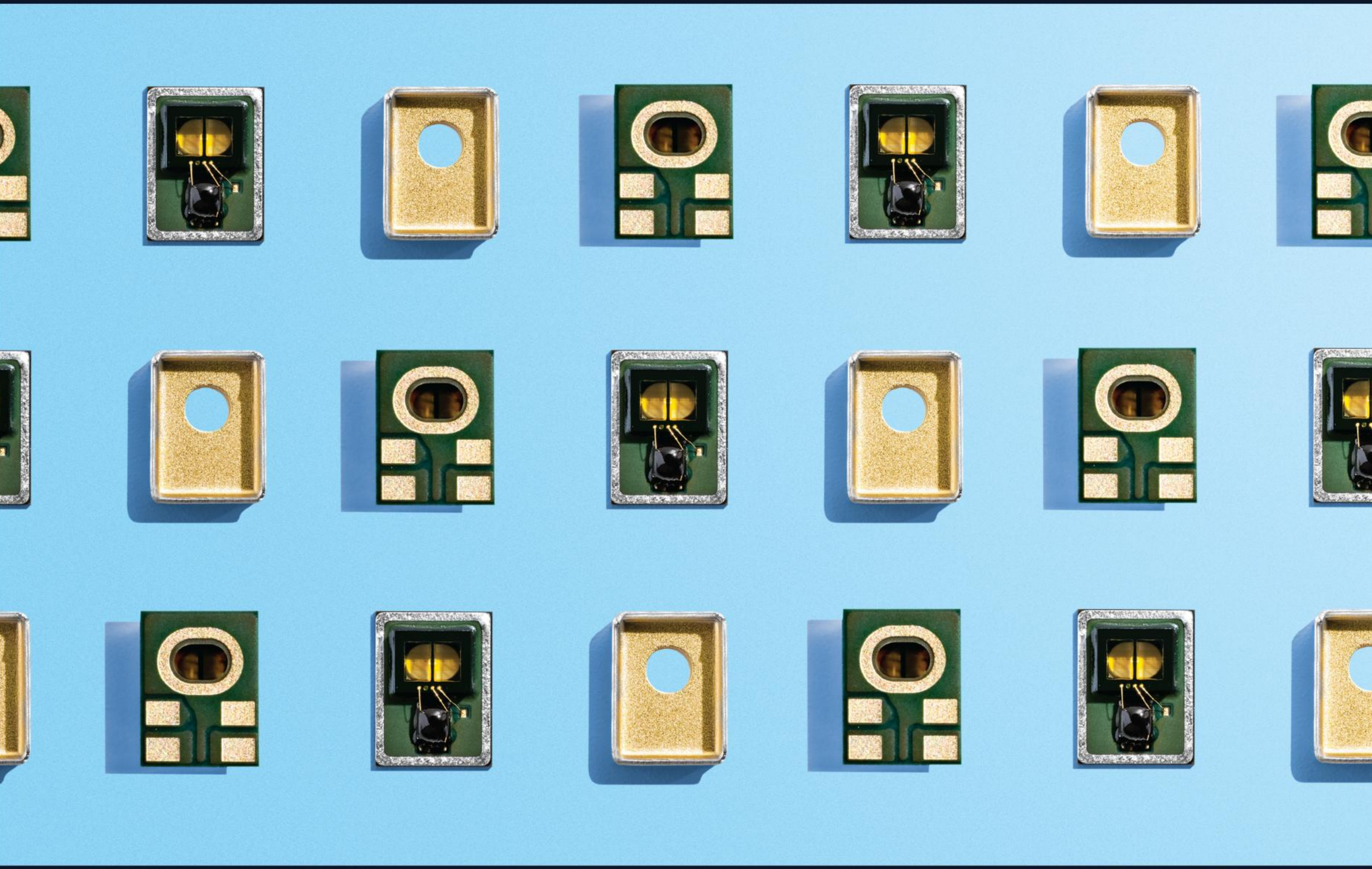
Soundskrit developed the first high-performance directional MEMS microphone on the market, leveraging years of research in bio-inspired MEMS based on how spiders and other insects in nature hear. In combination with Soundskrit's in-house audio processing algorithms, directional microphones can be used to capture and isolate any sound in an environment with a fraction of the size, power, and computation of traditional omnidirectional-based microphone arrays.

Soundskrit was founded in 2019 and is headquartered in Montreal, Quebec with an R&D facility in Ann Arbor, Michigan.



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