## Introduction to Measuring SNR on Directional Microphones

This application note provides guidance for engineers to measure **Signal to Noise Ratio (SNR)** on Soundskrit directional microphones. For reference, a possible test setup and data processing techniques are included.

SNR is a critical metric to quantify the performance of any microphone. This indicates the quietest sound a microphone can detect above the noise floor. This metric also reflects sound quality by indicating how audible the electrical noise floor is in a recording. A higher SNR indicates a microphone can pick up softer sounds or has a lower audible noise floor.

SNR is commonly simplified into a single value. The SNR of a microphone is typically calculated by dividing the 1 kHz sensitivity of the microphone by the A-weighted, integrated noise across the audio spectrum. This A-weighted value uses the sensitivity at 1 kHz to approximate the performance of the microphone across the entire frequency range. This is useful for quickly comparing the performance of different microphones. However, SNR varies with frequency so this can misrepresent the true performance of some microphones.

If a microphone has a 'flat' frequency response, the sensitivity does not change significantly with frequency. In this case, using only the 1 kHz sensitivity is representative of the microphone's performance across its operating range. While it is a safe assumption that omnidirectional microphones will have a flat frequency response, directional microphones inherently have a sloping frequency response. Because of this, relying on the SNR at 1 kHz will skew the performance of the microphone. Depending on the characteristics of the microphone, this can artificially improve or worsen the 1 kHz SNR. In Figure 1 below, you will see an omnidirectional microphone's frequency response alongside the sloped response of a directional microphone from Soundskrit.

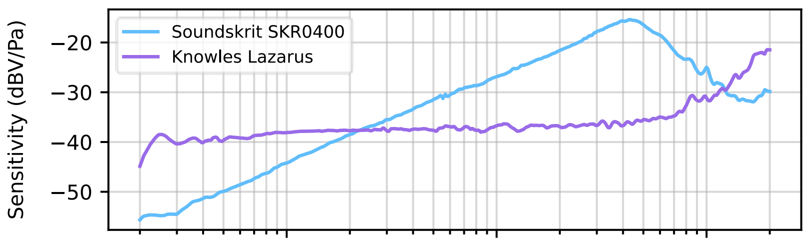



*Figure 1: Omnidirectional flat frequency response vs. Soundskrit directional microphone frequency response*

For directional microphones, we recommend using a process we refer to as **Full Band SNR**. This considers the full-band noise performance by assessing the sensitivity versus the input-referred noise. Essentially, the SNR is evaluated at all frequencies and integrated into the single A-weighted SNR value. This would provide a similar result as equalizing the microphone to a flat frequency response and then measuring the SNR using the 1 kHz value. Likewise, for an omnidirectional microphone, this method would produce the same value as the typical, single-frequency method. This single value enables a true representation of SNR for directional microphones and can be used to compare against traditional omnidirectional microphones.

## Directional Microphone Frequency Response

Soundskrit directional microphones create a dipole polar pattern by using two sound ports and a specialized transducer. These microphones sense the pressure delta between both ports which increases sensitivity to sounds coming from certain directions. Omnidirectional microphones sense pressure changes at a single port, making them insensitive to sound direction.

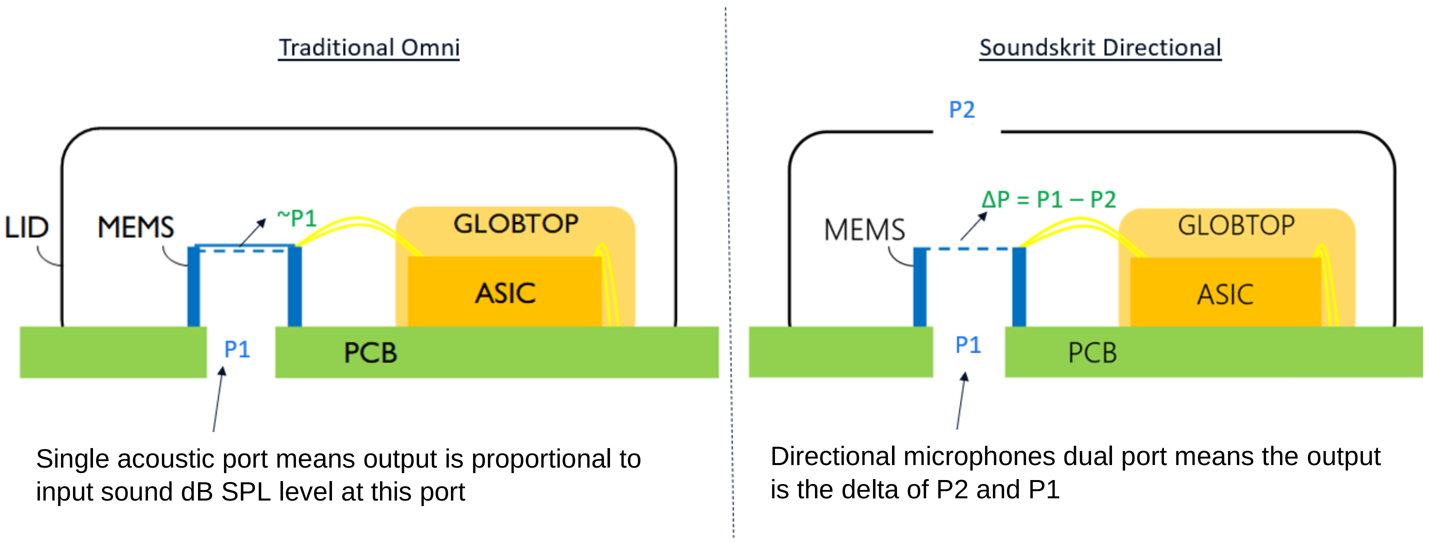



*Figure 2: Comparing omnidirectional microphones to a Soundskrit directional microphone*

Directional microphones measure the difference in pressure across ports P1 and P2. If the sound is coming from in front of the sensor, P1 will be closer to the sound source and the soundwave will have to travel further around the microphone to enter P2. Thus, the soundwave at P2 is slightly out of phase with P1 resulting in a pressure difference across the transducer. If the sound source is to the side of the microphone, the soundwave will hit P1 and P2 simultaneously resulting in equal pressure at both ports such that there is no delta across the transducer. To learn more about this effect and how Soundskrit microphones create a directional polar pattern, refer to this video which shows how our transducer responds to sound from different directions: How does Soundskrit's technology revolutionize MEMS microphones?

The pressure delta between P1 and P2 is likewise dependent on the wavelength of the soundwave. The smaller the wavelength, the greater the change in amplitude will be across the two points as the amplitude changes more quickly at higher frequencies. As the frequency of the sound decreases, the wavelength will increase resulting in a smaller pressure delta, thus the microphone is less sensitive at lower frequencies. This is why it is important to calculate the 'Full Band SNR' to account for the performance across all frequencies. Below is an illustration of two soundwaves of equal amplitude at 1 kHz and another 250 Hz demonstrating where in the wave each sound port is sampling to show how the pressure delta is affected.

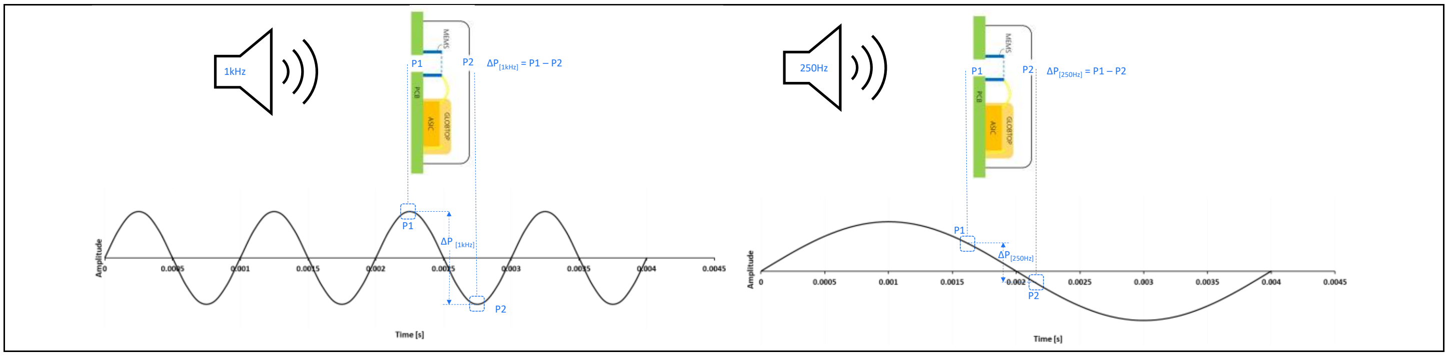



*Figure 3: Example of directional microphone sampling soundwaves of two frequencies*

These two plots demonstrate why lower frequency has less output for the same input dB SPL level.

$\Delta P_{[250Hz]} << \Delta P_{[1000Hz]}$ as the longer 250Hz wavelength results in a much smaller delta pressure across the two ports.

You may have noticed, however, that the frequency response of the SKR0400 in Figure 1 is not linear and peaks around 4 kHz. The acoustic resonance of the microphone also has an impact on the sensitivity. Our microphone transducer is designed to have a resonance at 4.3 kHz to maximize SNR for the human voice. After this resonance, the sensitivity begins to decrease.

## Calculating Full Band Signal-to-Noise Ratio

To calculate the Full Band SNR, 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 1: SNR calculation*

To calculate the SNR of the microphone across the audible frequency range, $f_1$ = 20Hz and $f_2$ = 20kHz. In some applications, the full 20kHz bandwidth may not be required. In this case, one integrates the input-referred acoustic noise across the desired audio band. This has the same effect as if you equalize the microphone to a flat frequency response and then measure the SNR at 1 kHz. This EQ-based method is the simplest way to calculate an accurate SNR for directional MEMS microphones and the method will follow in this guide.

## Measurement Procedure

Below is the recommended method to measure SNR. This method approaches full band SNR by equalizing the DUT and then measuring the 1 kHz SNR rather than fully measuring and calculating the full band SNR as detailed above since this will be the procedure most familiar to acoustic engineers. Below are the steps required to equalize the DUT and measure the SNR. There are several reference images, however these were not taken in an anechoic environment so these should be used as a guide rather than a specific benchmark. Measurements taken in an anechoic environment will be far more accurate and consistent.

Equipment required:

- Reference microphone – We use the EMX-7150 from iSEMcon
- Reference speaker – This speaker should act as a point source so the speaker should either be a single driver or have a coaxially mounted tweeter. We use the Avantone Active MixCube
- Audio analyzer
  - We use the APx500 from Audio Precision (AP), all instructions for configuring the audio analyzer in blue are specified only for this and similar models. Please refer to your equipment's manual if using another analyzer

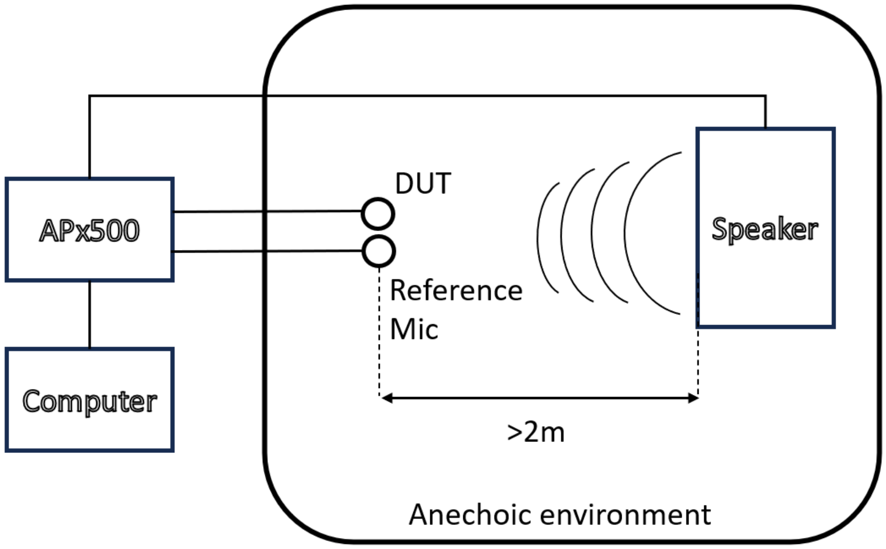



*Figure 4: Measurement setup block diagram*

1. **Setup:** In an anechoic environment, place a reference microphone and the Device Under Test (DUT) at least two meters away from the speaker. Connect the microphones and speaker to the audio analyzer. As analog MEMS microphones require a dedicated supply voltage, there are two methods to connect the microphone to the analyzer. First, you can use our SPIDAR board to provide power and connect directly to the XLR input or if you have a PDM module for your analyzer, this can be used to provide power.

    a. Our SPIDAR board includes an output labeled “AP Out” which directly connects to the output of the channel three input. To use this method:
        i. Connect the DUT to channel 3 on the SPIDAR board
        ii. Power the SPIDAR board by connecting it over USB to a computer
        iii. Use a 3.5mm to XLR cable (such as this one) to connect the output labeled “AP Out” to an analog input on your audio analyzer.

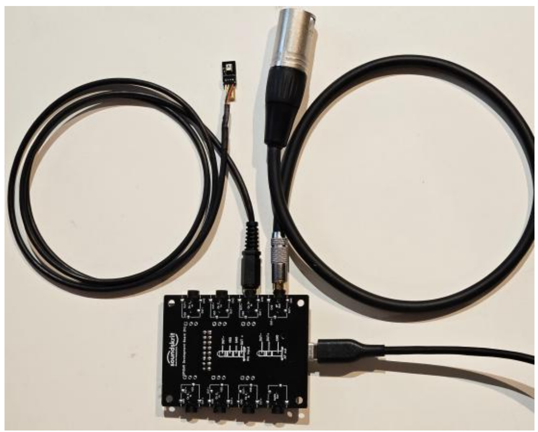


*Figure 5: SPIDAR Board AP Out Connection*

    b. If you have the PDM Module for your audio analyzer, this can also be used to power the microphone. To use this method:
        i. Connect VDD to VDD supply on the PDM Module
        ii. Connect OUT+ and OUT- to the analog + and – inputs, and ground to ground
        iii. In AP, under *Monitors/Meters,* set Vdd to 1.9 V and enable it


*Figure 6: PDM Module Vdd Supply Connection*

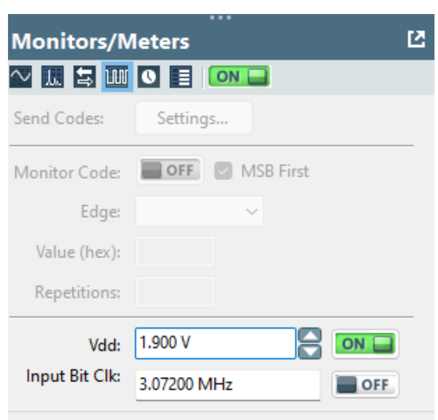



*Figure 7: Setting Vdd*

2. **Equalize the speaker:** EQ the speaker to output 94 dB SPL across all frequencies at the reference microphone location.

    a. Get the frequency response of the reference microphone output in dBV from the calibration file. Set reference microphone input in Signal Path Setup.

    b. Play a frequency sweep from 20 Hz to 20 kHz at a reference output level and measure the frequency response of the reference microphone in dBV. We use -33 dBV output, which gives us roughly 94dB SPL at 1 kHz, but this will depend on the speaker and system. In AP, select *Add Measurement → Frequency Response* to play a sweep from 20Hz to 20kHz at a level such that at the DUT location, you get around 94dB SPL.

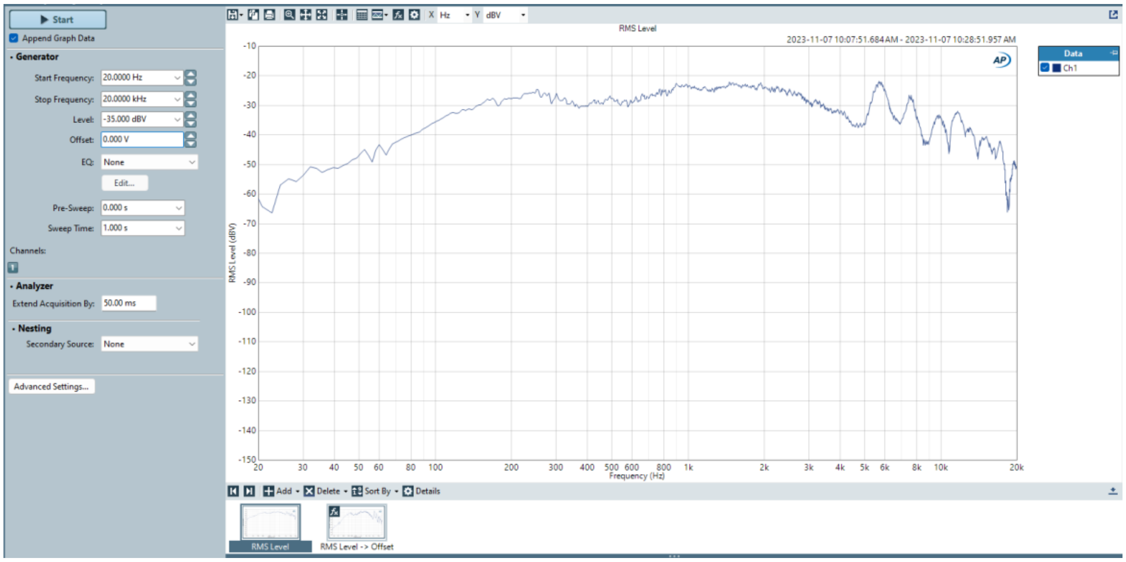



*Figure 8: Frequency Sweep Generation in AP*

c. To calculate the speaker EQ, subtract the measured reference microphone frequency response from the reference microphone sensitivity at 94 dB SPL: ($EQ_{speaker} = dBV_{ref\text{-}sen@94dB\ SPL} - dBV_{ref}$). The reference microphone sensitivity can also be a frequency-dependent function but for simplicity, we assume flat reference microphone response.

i. In AP, this can be done in the measurement plot window by clicking *Add → Derived Result → Derived: Offset* where *Source* is the reference microphone measurement. *Offset gain* is the 94 dB SPL sensitivity of the reference microphone: *Gain* = $-S_{ref\ measured@94dB}$.

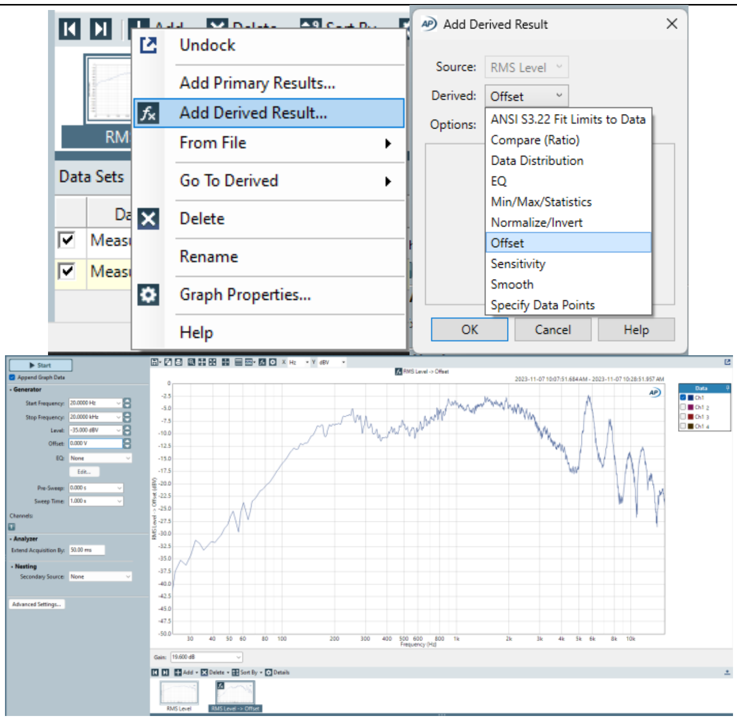



*Figure 9: Speaker EQ*

ii. Then, export this measurement. To import this as an EQ, the units must agree, so ensure the Y-axis label must be "dB" not "dBV" before exporting. Then import in *Signal Path Setup → Output Configuration → EQ → Create New...*

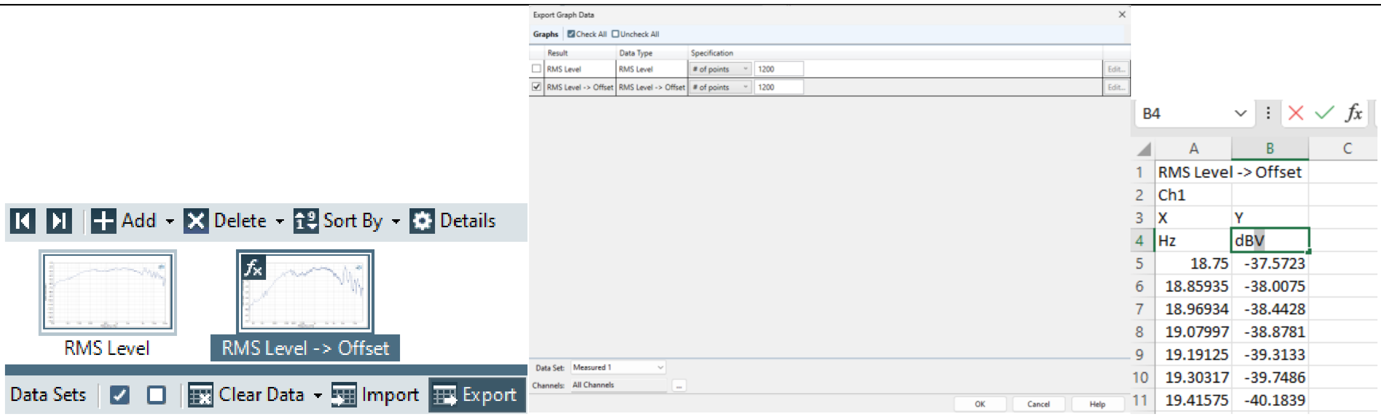



*Figure 10: Importing the EQ*

d. After applying EQ to the speaker, check that when you run the same Frequency Response measurement, you now get a flat line for the reference microphone response. (note: one may also use the reference dB SPL1 in AP, where you input 94dB SPL sensitivity level using known sensitivity and/or calibrator, then $EQ_{speaker}$ = 94dB SPL - dB $SPL_{ref}$).

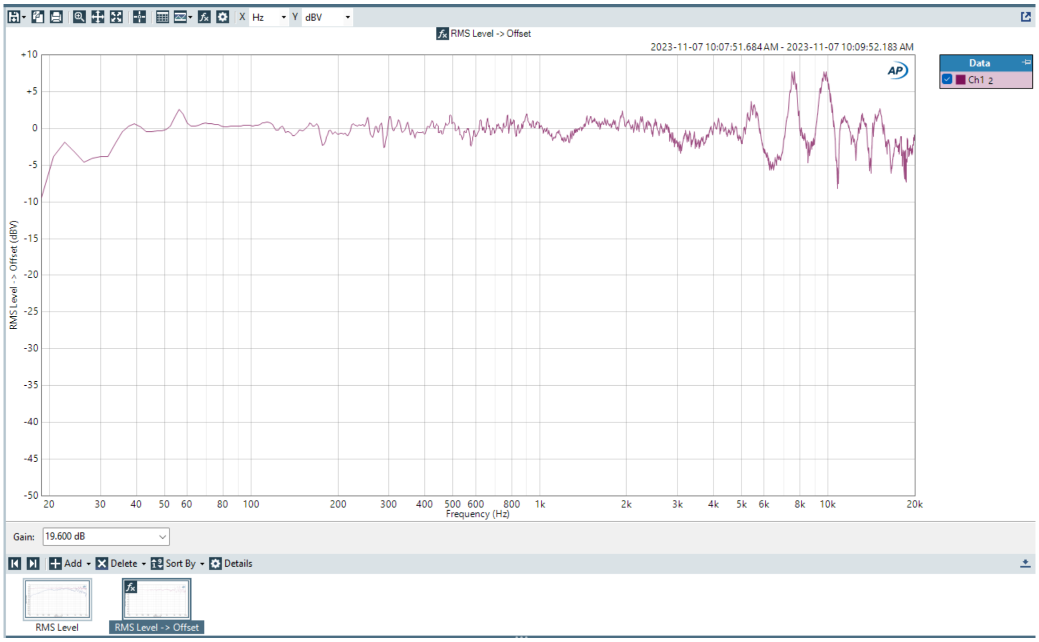



*Figure 11: Speaker with EQ applied*

3. **Equalize DUT:** Equalize the DUT output to be flat at all frequencies with 1 kHz sensitivity.
    a. In *Signal Path Setup*, change *input* to the DUT connector according to your configuration.
    b. In the Frequency Response measurement, play the same equalized speaker frequency sweep from above and get non-equalized DUT frequency response, if the above steps were done correctly, this frequency response should closely match the frequency response from the datasheet of the DUT. Note here that the datasheet measurement is represented relative to 1 kHz. To match this with your measurement, set the cursor to 1 kHz and adjust the offset so that the sensitivity at 1 kHz is 0 dBV.

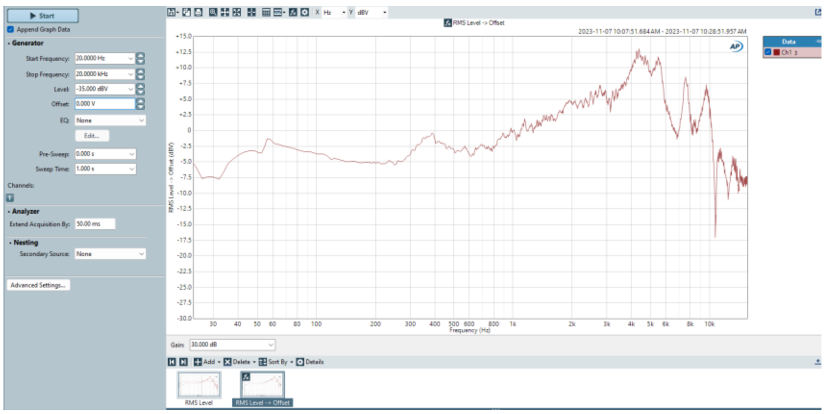


Figure 12: Frequency Response of Measured DUT (SKR0400)

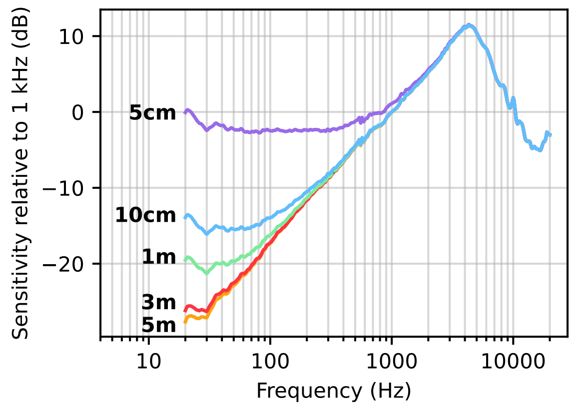



Figure 13: DUT (SKR0400) Datasheet Response

c. For the EQ, divide the DUT 1 kHz sensitivity by the output response at each frequency ($EQ_{DUT} = dBV_{sen1\ kHz} - dBV_{sen}$).
    i. To do this in AP go to *Add → Derived Result → Derived: Offset* where the source is the DUT mic response. Offset gain is the inverse of the 1 kHz DUT sensitivity, which can be found using the cursors on the measurement plot (*Offset Gain* = -$dBV_{sen1\ kHz}$). Export as a .csv.

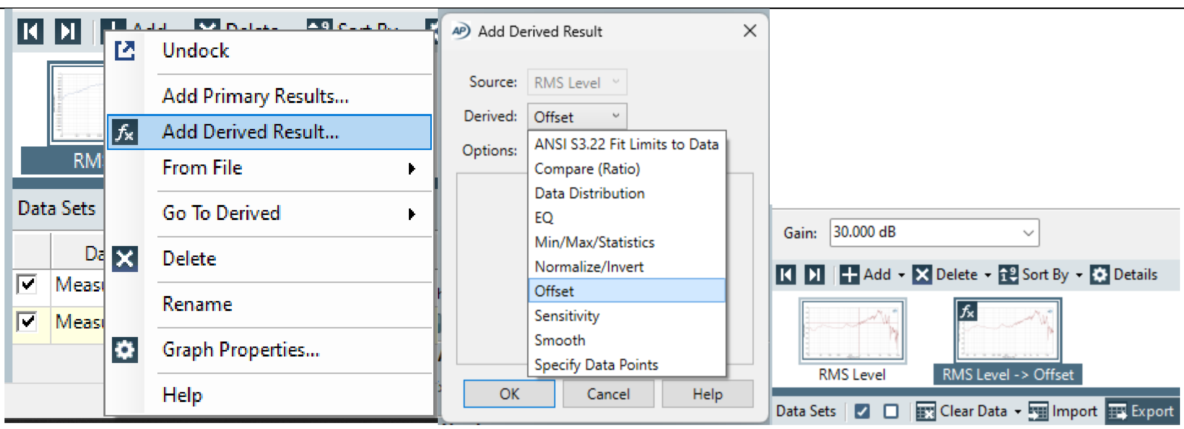



Figure 14: Generate the EQ

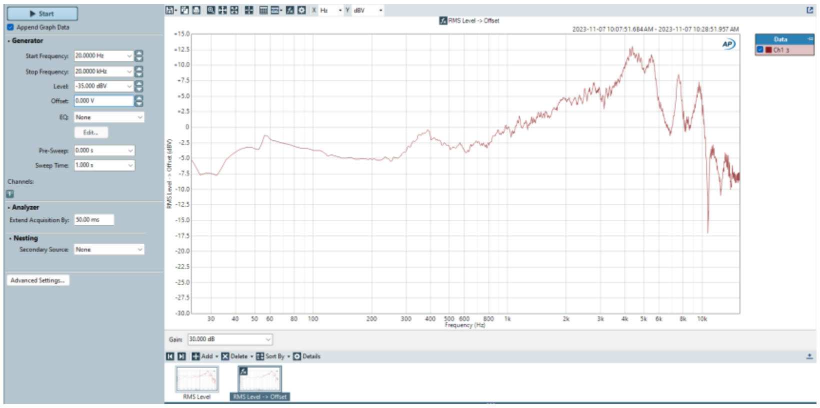


Figure 15: DUT Response

ii. Import the offset DUT frequency response file as an EQ in *Signal Path Setup* → *Input Configuration* → *EQ* → *Create New…* → *Import* and choose to "*Invert*" in the *Edit EQ* Table. The EQ level at 1 kHz should be equal to zero and the EQ curve should have the shape of an upside-down DUT sensitivity curve.

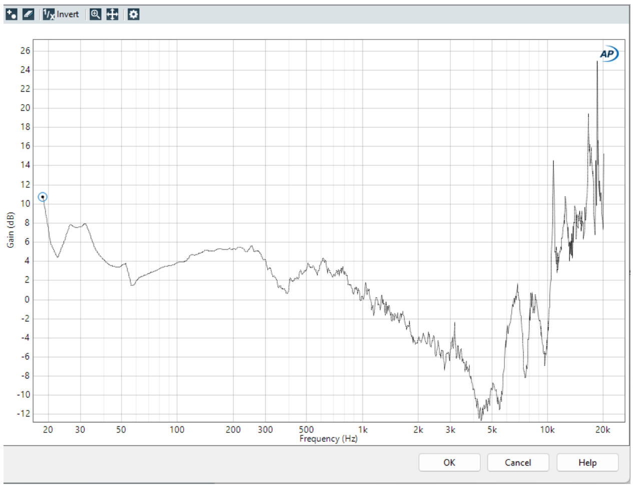



Figure 16: EQ curve for DUT after being imported and inverted

d. After applying the EQ to the DUT, play the frequency sweep in the Frequency Response measurement with both speaker and DUT EQ enabled. Check that the output frequency response is flat. Sensitivity at all frequencies should be equal to the DUT nominal 1 kHz sensitivity.

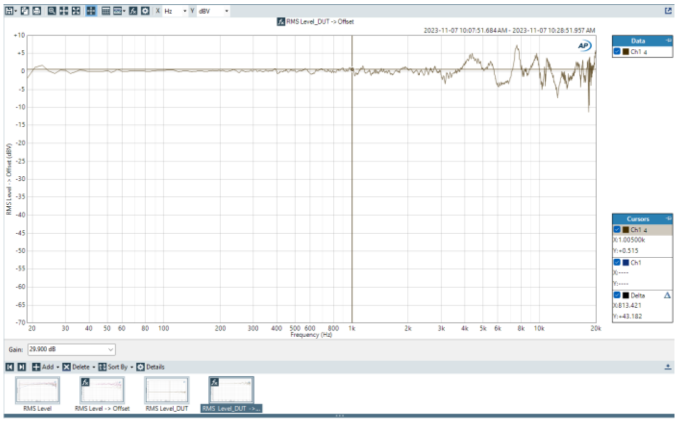


*Figure 17: DUT response with the speaker and microphone equalized*

4. **Calculate**: Calculate 1 kHz SNR using 1 kHz sensitivity at 94dB SPL / 1 Pa (found in previous step) and equalized integrated A-weighted noise level: SNR = $dBV_{sen1\ kHz} - dBVA_{EQ\text{-}noise}$.
   a. In AP Signal Path Setup, apply A-weighting to the input under the *Input Configuration → Filters → Weighting* drop-down menu. View RMS level in dBV to get dBVA noise level.

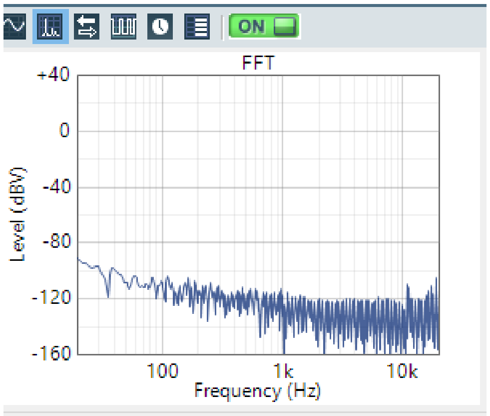



*Figure 18: Raw noise*

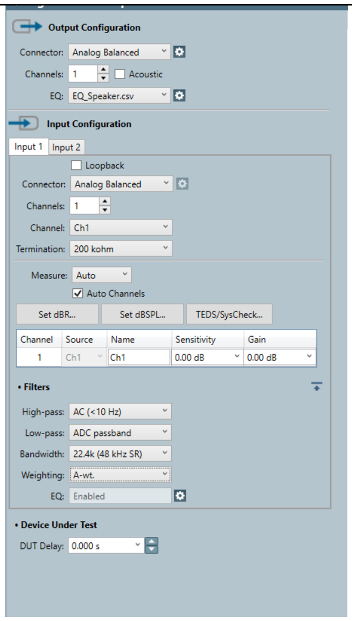



*Figure 19: A-weighting the noise*

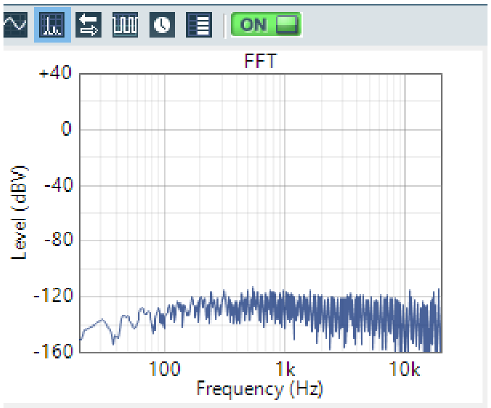



*Figure 20: A-weighted noise*

## Conclusion

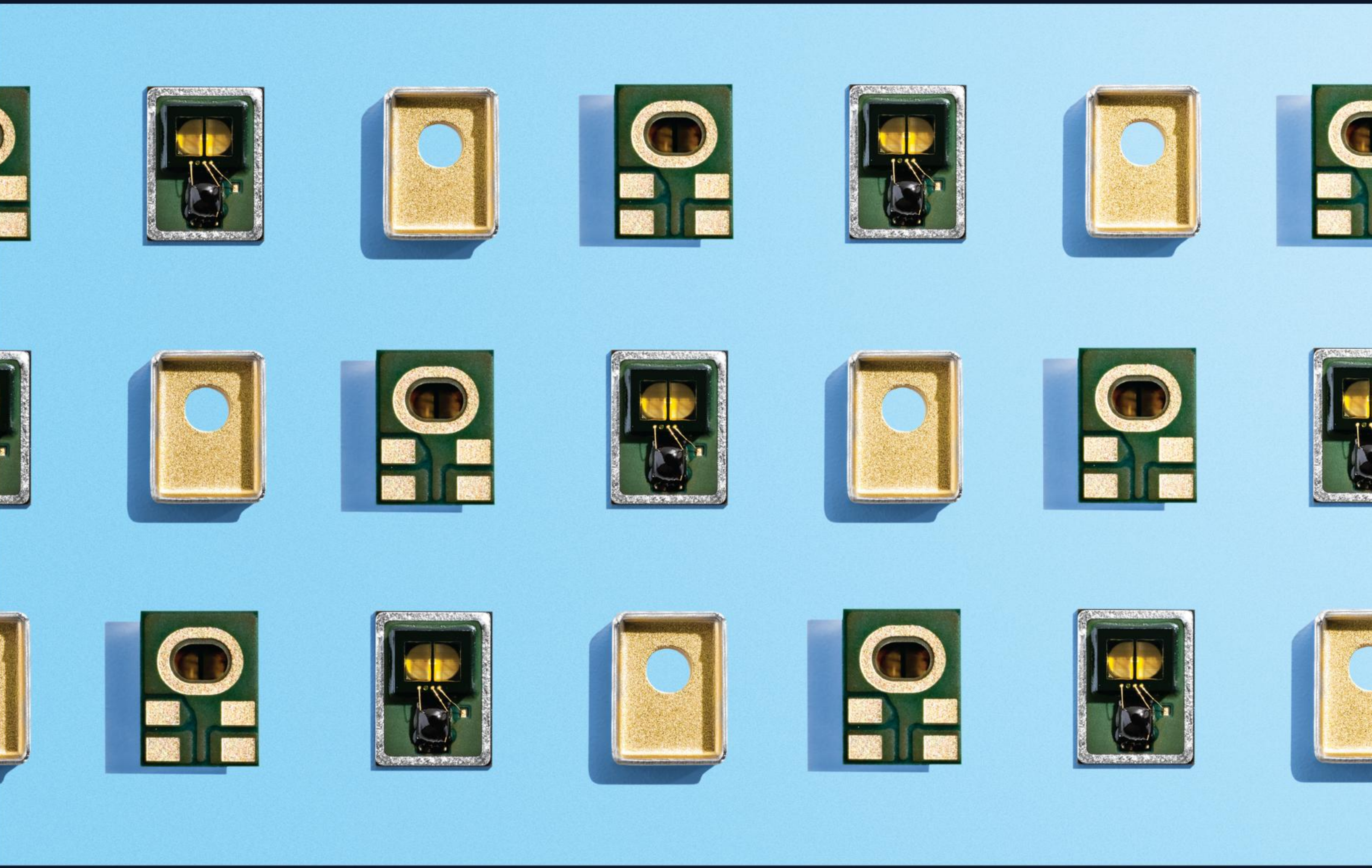
This method of calculating the Full Band SNR of a directional microphone by equalizing the frequency response allows you to compare directional and non-directional microphones' performance more effectively. For more assistance and support please reach out to us at applications@soundskrit.ca

## Revision History

| Revision Label | Revision Date | Sections Revised |
|---|---|---|
| - | December 2023 | Release |
| B | Feb 2025 | Wording improvements for clarity |

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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Hear the impossible.