Introduction to the Soundskrit Scarab Kit

Figure 1: Soundskrit Scarab Kit

The Scarab demo kit is a tool to evaluate Soundskrit microphones and their potential when combined with beamforming algorithms and Soundskrit audio processing. The kit leverages our SKM1600, a compact module with an integrated dipole and an omnidirectional MEMS microphone. The Scarab kit includes the SKM1600, an adapter board, and our PARDI board, which is a convenient interface to connect MEMS microphones over USB. The PARDI board runs embedded beamforming algorithms on the SKM1600, demonstrating how different polar patterns and non-linear algorithms can be used to enhance directionality and reduce background noise.

What's in the Box	
SKM1600	Small module with a dipole and an omnidirectional microphone
PARDI Board	MEMS microphone interface board running embedded beamforming algorithms
SKM Adapter Board	An adapter to connect the SKM1600 to the PARDI Board
Molex Cable	Preconnected cable which links the microphone module adapter board to the PARDI audio interface
USB A to USB C Cable	Cable to connect PARDI board to your computer

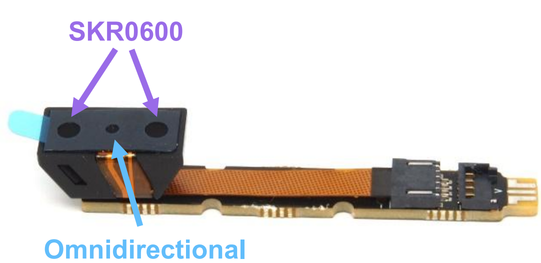


Figure 2: SKM1600 module with adapter board

Table of Contents

Board Overview and Pinouts

The PARDI board has a USB-C connector, one Molex 4-pin connector to connect two analog microphones, a 3.5 mm analog stereo output connector, and a Molex connector to connect PDM microphones (not shown in the photo below).

The board is preconfigured to connect the SKM1600 to the analog input 4-pin Molex connector of the PARDI board.

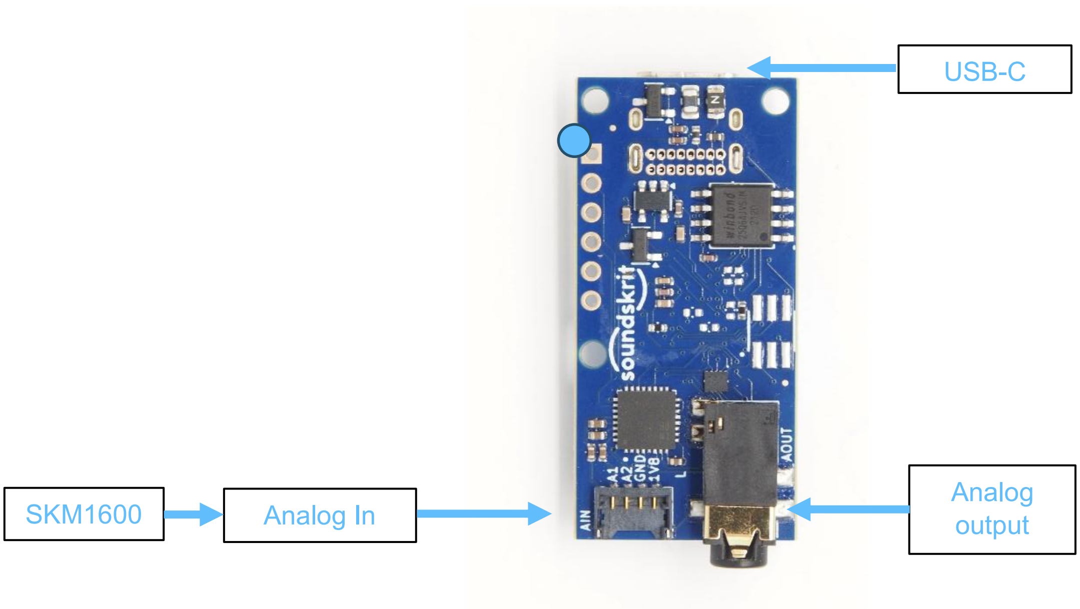


Figure 3: PARDI board connections.

SKM1600 Module

The SKM1600 is a small module with both omnidirectional and dipole microphones to demonstrate adjustable polar pattern beamforming. The SKM1600 is product ready and makes microphone integration easy, we have done all the work in designing the acoustic path, ensuring sealing, proper material selection, etc. The side opposite the flex connector is the front or 0° on the polar pattern.

To facilitate integrating the module into a product, the module has a strong double-sided adhesive tape on the top. To expose the adhesive, peel off the blue cover tape on the top. This exposes the adhesive, allowing the module to be securely attached inside a prototype. Take care to align the sound ports to ports in the prototype.

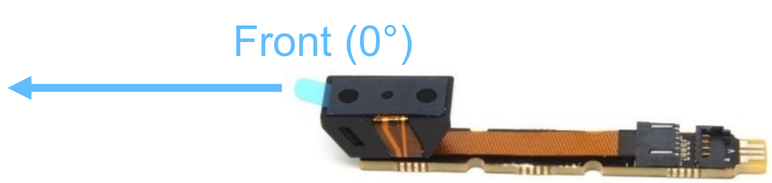


Figure 4: SKM1600 on adapter board.

Connecting analog microphones to the board

To connect the two analog microphones of the SKM1600 to the PARDI board, the cable with Molex connectors is plugged into the SKM1600 adapter board and the PARDI board's analog input connector. The cable is preconnected and only needs reattachment if disconnected.

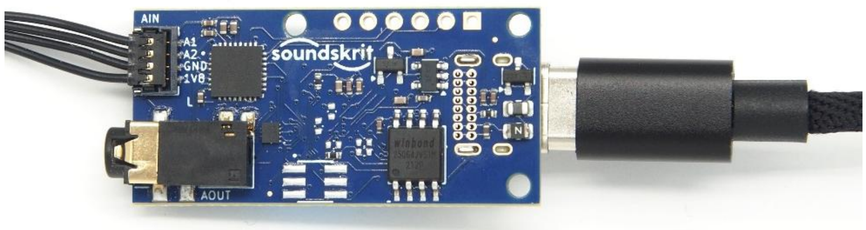


Figure 5: Scarab Kit Connections

Connecting to the analog output

The 3.5 mm jack connector is the board's analog headphone output.

Both analog output channels, left and right channels of the headphone connector, contain the same signal as the USB recording channel 1. To change the signal on the analog output, route a different beam pattern to USB channel 1 using the "set algorithm 1 <algorithm>" command as explained on pages 9-**Error! Bookmark not defined.**.

Included Beamforming

Once the SKM1600 is connected, you can take a recording. The USB output will have six output channels. The first two are the equalized microphone signals, outputs 3 to 5 are the included beamforming algorithms. Output 6 is the raw pass through of the module's dipole microphone, this can be used for measurements of the dipole. The output configuration and measurements of each of the beamforming algorithms are below.

To learn more about combining an omnidirectional and dipole microphone to create these adjustable polar patterns, check out our article Combining Microphone Polar Patterns.

All output channels implement a high-pass filter set to 80 Hz by default. This high-pass filter frequency is configurable (see Tera Term section below).

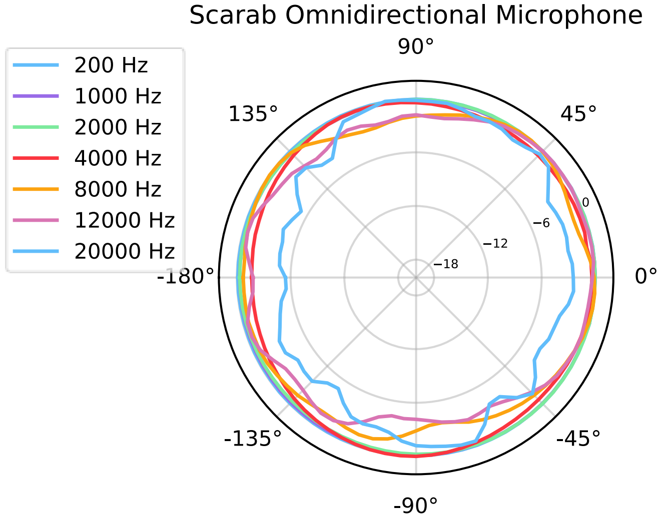
Note: The microphone input gain is set for mid-field distances above 30 cm. If used at close range, the signals might clip. If this occurs the gain can be adjusted through firmware commands.

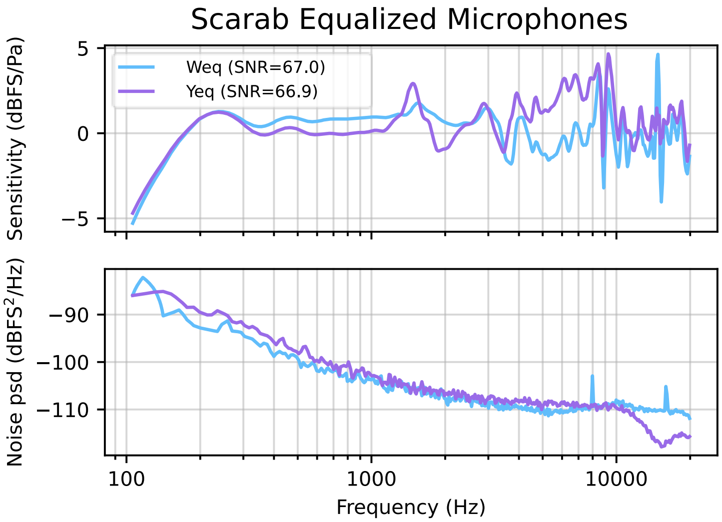
Default USB Output Channels		
Out1	Equalized SKM1600 omnidirectional microphone	The omnidirectional microphone of the SKM1600 module with an EQ applied to match the dipole
Out2	Equalized SKM1600 dipole microphone	The Soundskrit SKR0600 microphone of the SKM1600 module with an EQ applied to match the omnidirectional microphone. Directional microphones inherently have a non-flat frequency response, so an EQ is applied to flatten the frequency response.
Out3	SimplyDSP cardioid beamformer	A beamformer creating a cardioid polar pattern using both directional and omnidirectional microphones.
Out4	SimplyDSP adaptive beamformer	This beamformer maximizes the signal to background noise ratio by dynamically changing to the optimal weighing of both omnidirectional and directional microphones.
Out5	SimplyDSP non-linear beamformer + AI	Our non-linear beamformer mutes the microphone unless speech is coming from ±60° and further reduces non-speech noise when the microphone is unmuted. Includes automatic gain control and AI noise reduction.
Out6	Raw dipole microphone	Output channel six directly passes the raw signal of the module's dipole without any processing applied.

This configuration is the default, but can be changed via firmware API, e.g., to change the channel routing or enable different linear beams like a hypercardioid. Refer to pages 9-**Error! Bookmark not defined.** for instructions.

Output 1: Equalized SKM1600 Omnidirectional Microphone

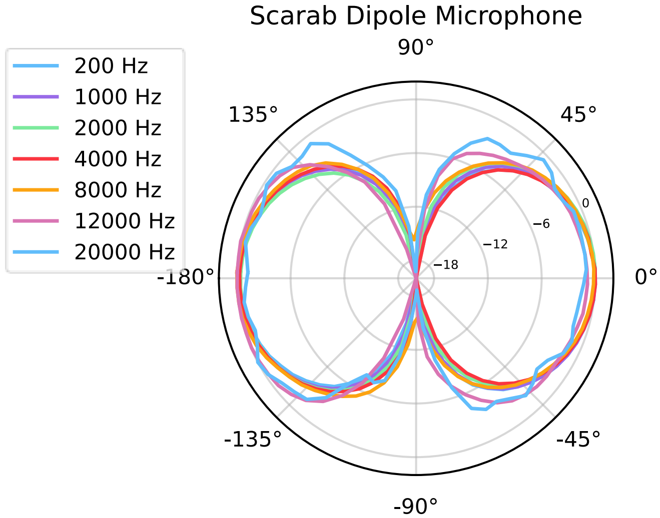
The omnidirectional microphone of the SKM1600 with an equalization (EQ) filter applied to match the dipole. It is necessary that the two microphones are equalized to match one another for further beamforming and processing.

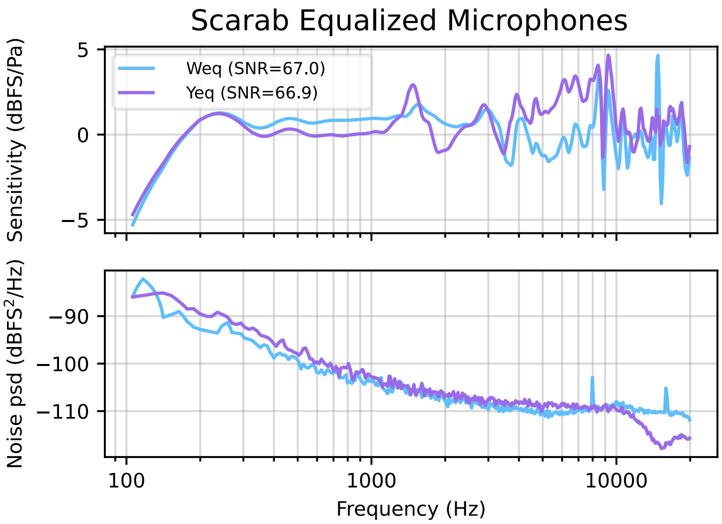




Output 2: Equalized SKM1600 Dipole Microphone

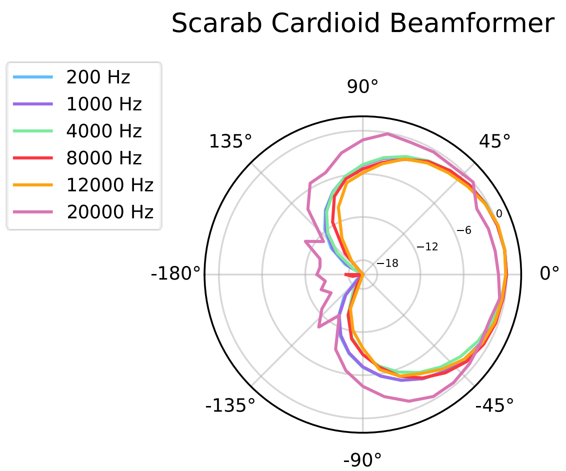
The dipole microphone of the SKM1600 with an EQ applied to flatten the response.

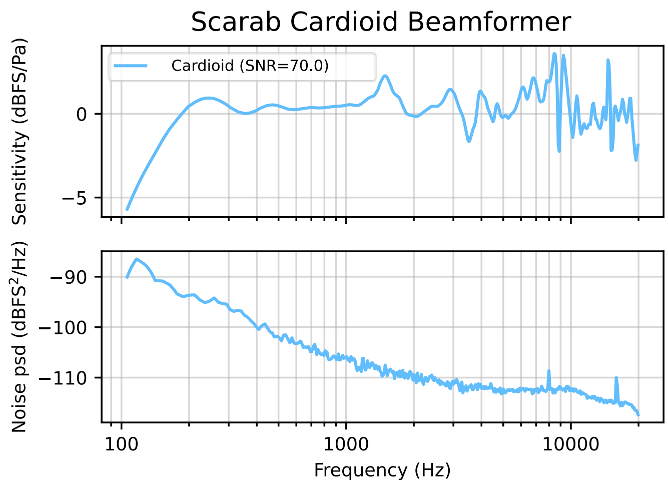




Output 3: SimplyDSP Cardioid Beamformer

A beamformer which creates a cardioid polar pattern using both directional and omnidirectional microphones. This pattern is useful to create a strong null in the rear. This does not use any noise reduction processing and just creates the cardioid pattern for distortion-free high-quality recordings.

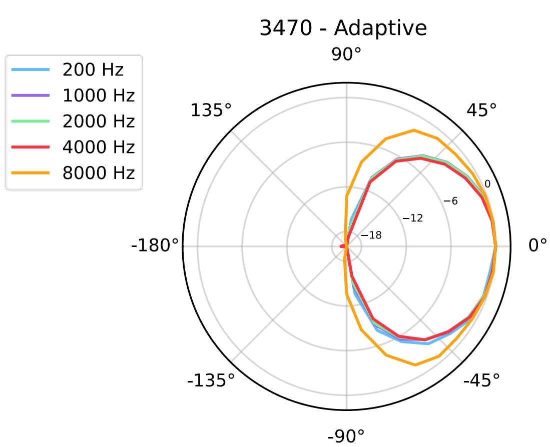


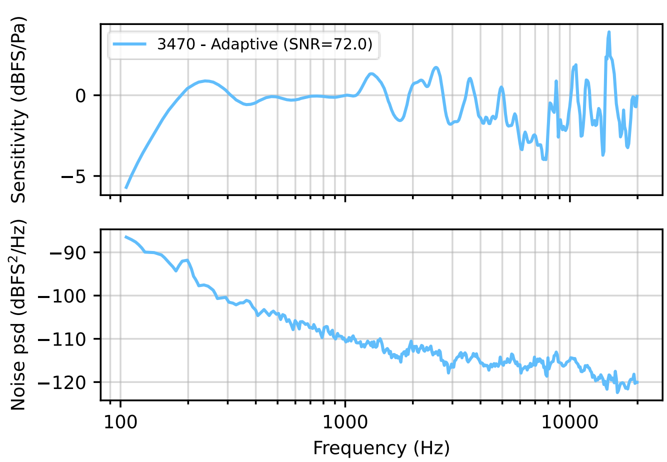


Output 4: SimplyDSP Adaptive Beamformer

The SimplyDSP Adaptive beamformer varies the polar pattern over time and frequency between omni and dipole to maximize noise rejection. This is a non-linear beamformer which rejects a large amount of noise and sounds more natural than typical non-linear processing techniques.

Note: The polar plot shown below shows the rejection of direct noise, the polar pattern in real scenarios varies over time.

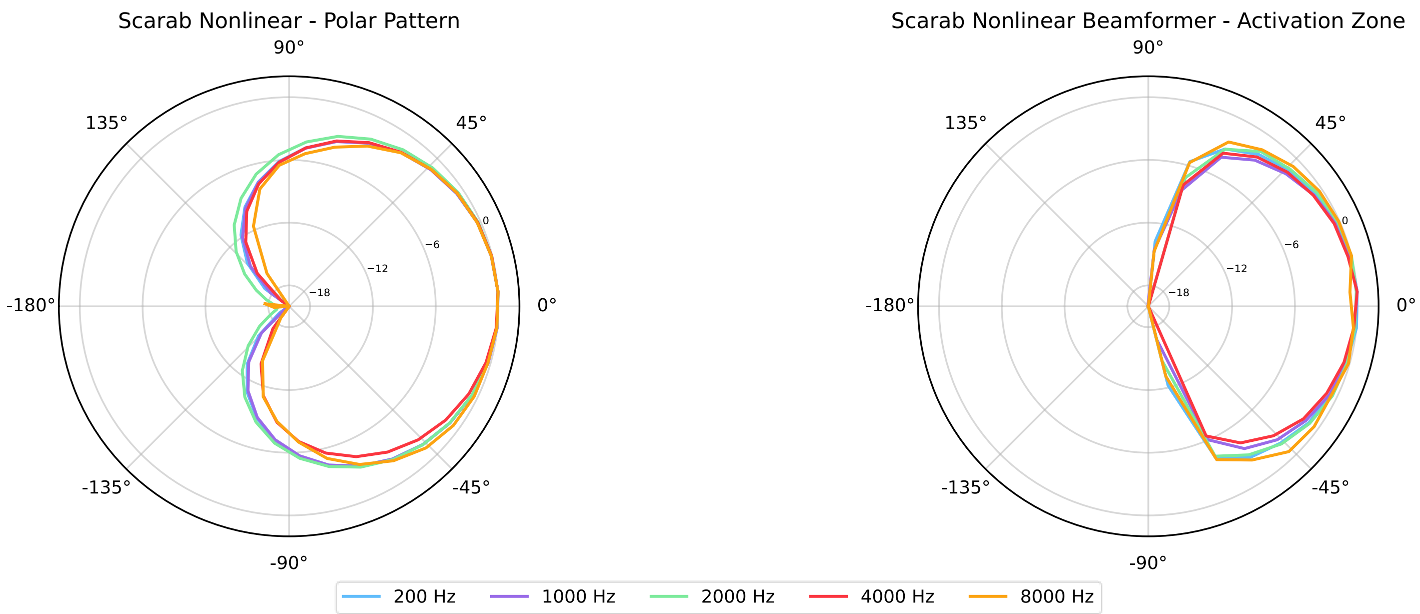




Output 5: SimplyDSP Non-linear Beamformer + AI

Our non-linear beamformer uses phase information provided by the omnidirectional-dipole pair to mute the microphone unless speech is coming from ±60°. This “activation zone” is pictured below. When the microphone is unmuted for the frequency bands containing speech, cardioid beamformers in those bands are used together with the phase information to further reduce non-speech noise.

Automatic gain control (AGC) on this beamformer optimizes signal levels for different speech distances. To further reduce noise, an AI noise reduction algorithm is enabled by default on this channel.



Output 6: Raw Dipole Microphone

Output channel six directly passes the raw signal of the module’s dipole microphone. This can be used for measurements of the raw microphone, without EQ or any other processing.

Changing the linear beam pickup pattern

The polar pattern of each channel can be changed through terminal commands. This can be used to access beam patterns which are not activated in the default configuration, e.g., a hypercardioid or supercardioid beam, or to change the order of the beams.

To change the polar pattern of output channel 3, follow these steps:

1. Download [Tera Term](https://github.com/TeraTermProject/teraterm/releases) from https://github.com/TeraTermProject/teraterm/releases. The latest release is listed on top, find the download link at the bottom of this version's box:

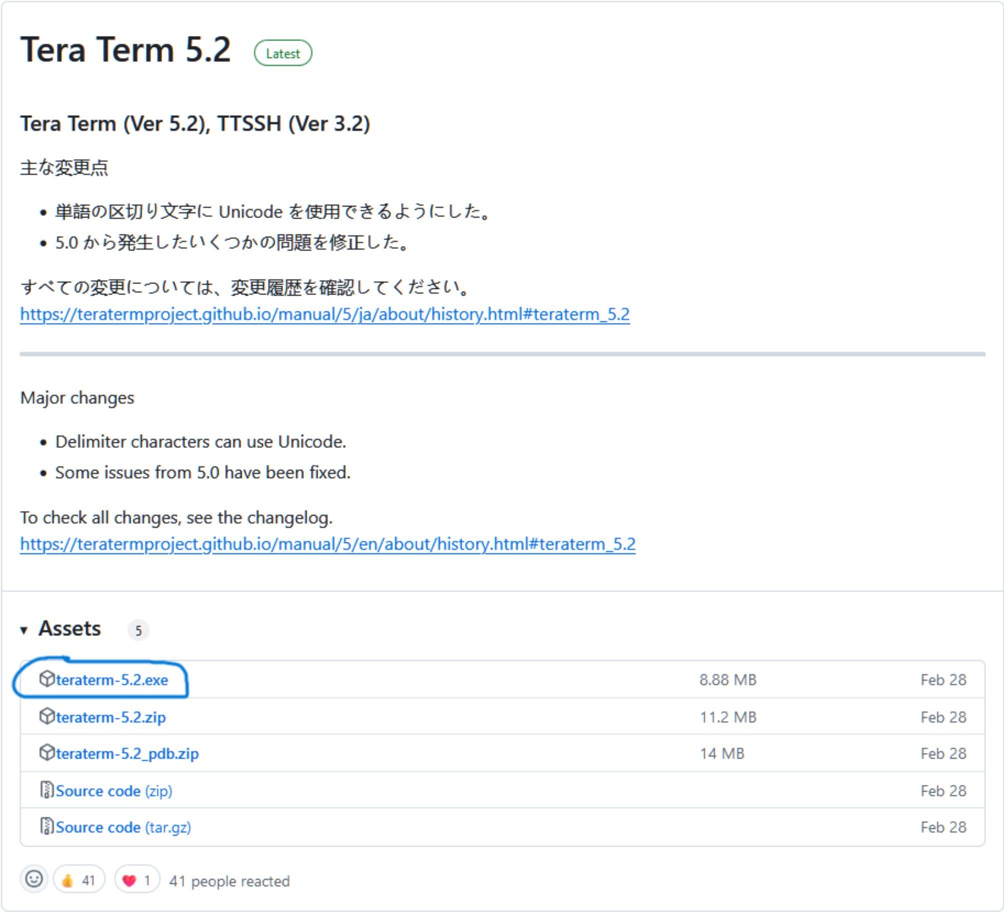


Figure 6: Tera Term download from GitHub. The Windows installer (.exe) is at the bottom of the latest version's box.

2. Install Tera Term.

3. Run Tera Term and go in the menu "Setup" → "Terminal". Change the options "New-line" both to "LF" and activate "Local echo". Click OK to apply the setting changes for this session. To save these settings permanently, use the menu save dialog in "Setup" → "Save setup".

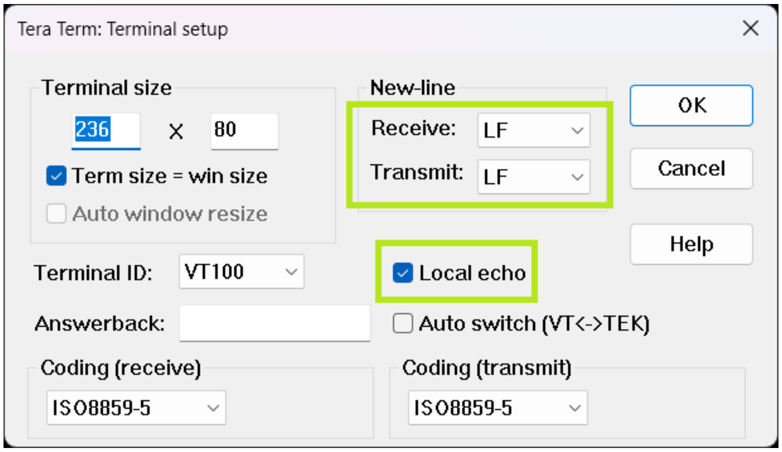


Figure 7: Terminal settings in Tera Term.

4. Connect the Soundskrit PARDI board to the PC using the USB cable.

5. In Tera Term, select the menu “File” -> “New connection”. In the popup, select the PARDI board, shown as “COMxx: USB Serial Device” (xx being a number) in the Serial port drop-down menu and click OK.

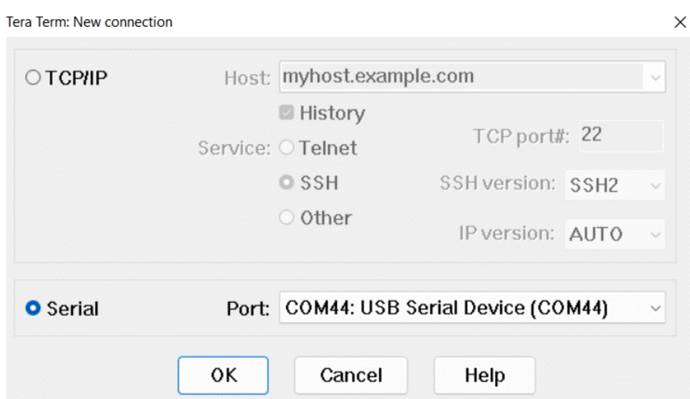


Figure 8: Connecting Tera Term to the PARDI board.

6. The current configuration is displayed when typing “get audio” and pressing ENTER. To display all commands, type “get help” or “h”, for all possible processing algorithms type “get algorithms”.

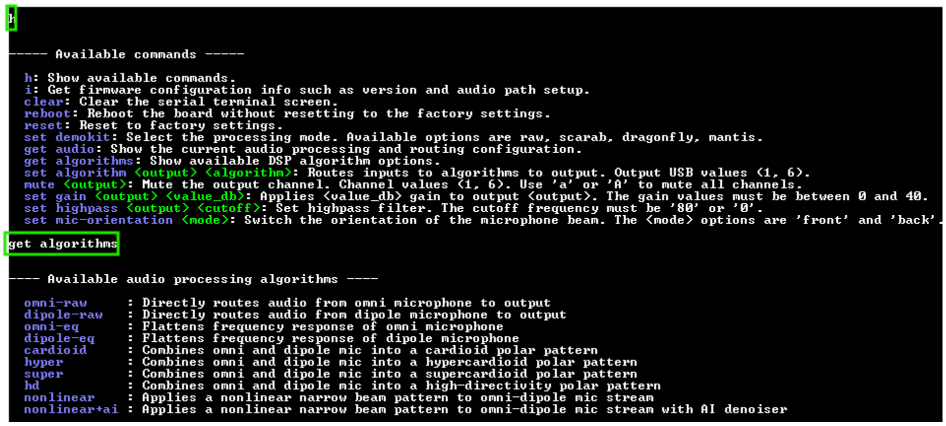


Figure 9: Help function and available algorithms listed in Tera Term.

7. To change the polar pattern on any channel, use the command "set algorithm <output> <algorithm>". <output> is the USB output channel you want to change, from 1 to 6. <algorithm> is the processing type you want to apply on this channel, for a full overview over the supported algorithms type "get algorithms" and press ENTER.

   Example: To change output channel 3, type one of the following in the terminal window:
   - For a cardioid: "set algorithm 3 cardioid"
   - For a hypercardioid: "set algorithm 3 hyper"
   - For a supercardioid: "set algorithm 3 super"
   - For the high directivity beam: "set algorithm 3 hd" (default)
   - For the nonlinear beam without AI noise reduction: "set algorithm 3 nonlinear"

   Press ENTER after the command to send the command. The new audio routing will be displayed in the terminal window after typing "get audio" as below.

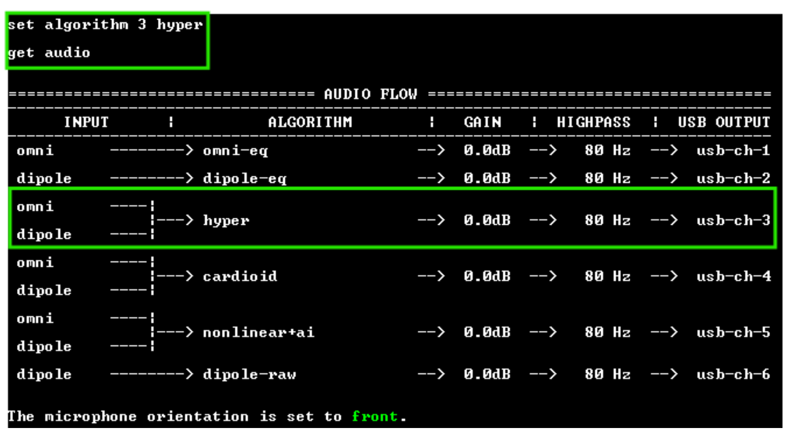


Figure 10: Changing the beam polar pattern on output channel 3.

8. To reset the PARDI board to the initial settings, type "reset", and press ENTER.

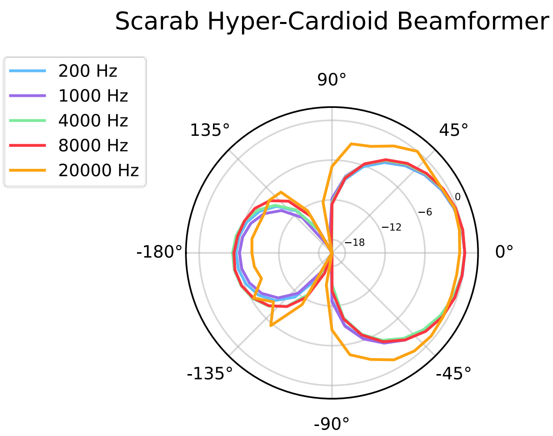
The achievable beam patterns are described below:

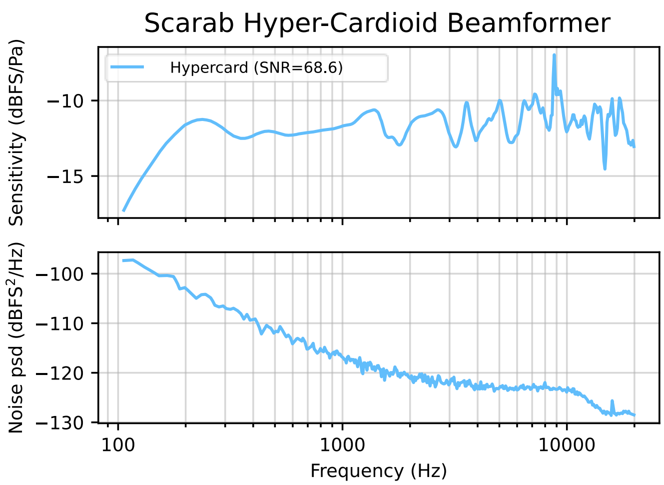
Cardioid

This is the default setting on channel 4, see “Output 3: SimplyDSP Cardioid ” on page 7 for more information.

Hypercardioid

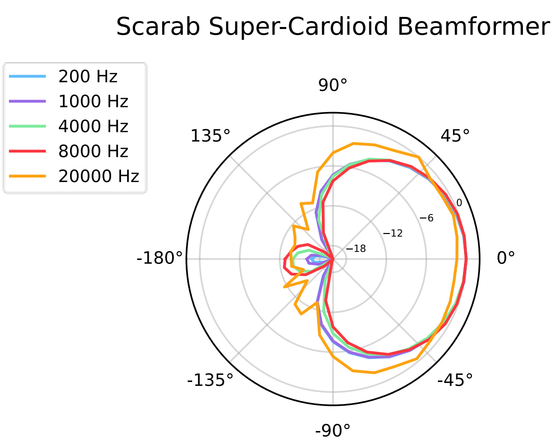
A beamformer which creates a hypercardioid polar pattern using both the directional and omnidirectional microphones. It has a narrower front-facing beam compared to the cardioid but features a rear lobe. The nulls are at 110° and 250° instead of the rear.

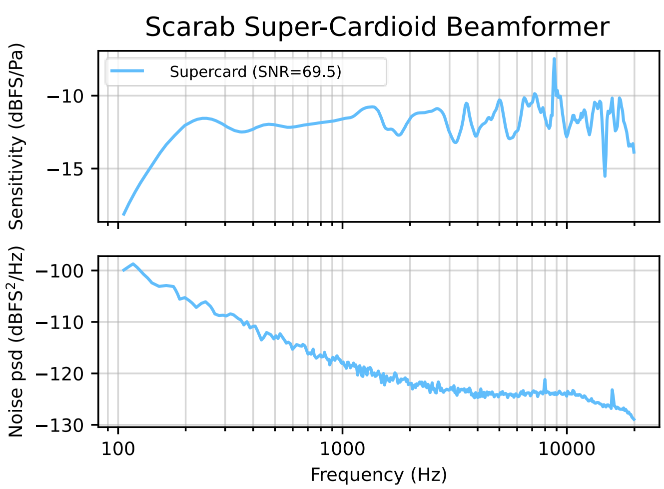




Supercardioid

A beamformer which creates a supercardioid polar pattern using both directional and omnidirectional microphones. It has a narrower front-facing beam compared to the cardioid, but a bit wider than in the hypercardioid. It features a rear lobe which is smaller than in the hypercardioid. The nulls are at 127° and 233°.





 www.soundskrit.ca

Disable/Enable the high-pass filter

By default, all channels have an 80 Hz high-pass filter activated to reduce unwanted handling noise of the bare microphone module. For measurements or low frequency applications, this can be disabled by typing the command “set highpass <output> <cutoff>”, with <output> the USB output channel from 1 to 6 and <cutoff> either “80” (filter enabled) or “0” (disabled).

Example: Disable high-pass on channel 6 (raw dipole) → “set highpass 6 0”

Changing the beamforming direction

By default, the front of the beamformers is at the side opposite of the flex cable, as shown on page 3. The direction can be reversed to the opposite direction, swapping 0° and 180°, by using firmware commands:

1. Access the terminal settings as described in #1-4 on pages 9 and 10.
2. Type “set mic-orientation back” and press Enter. This defines the side with the flex connector as the front for the beamformers.
3. To revert to the default orientation, type “set mic-orientation front”.

Recording Audio with the PARDI Board

To record audio with the PARDI board, we recommend installing Audacity. Audacity is a trusted, free to use, multiplatform suite of tools for recording and working with audio files.

Once you have installed Audacity, we need to configure the software for use with the Soundskrit PARDI board. Configure the settings as listed below:

Audio Host	Windows WASAPI
Input	*Microphone (Soundskrit Scarab)*
Output	Your listening device
Channels	6 Recording Channels

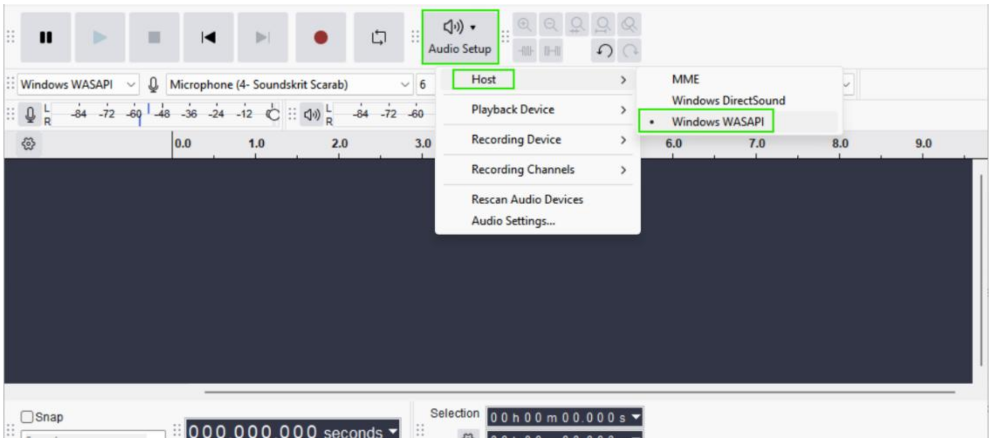


Figure 11: Audacity host selection.

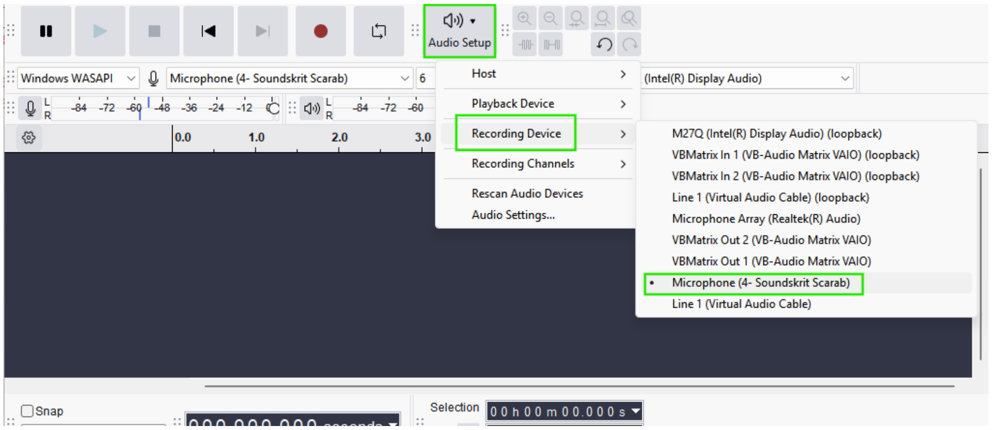


Figure 12: Audacity device selection.

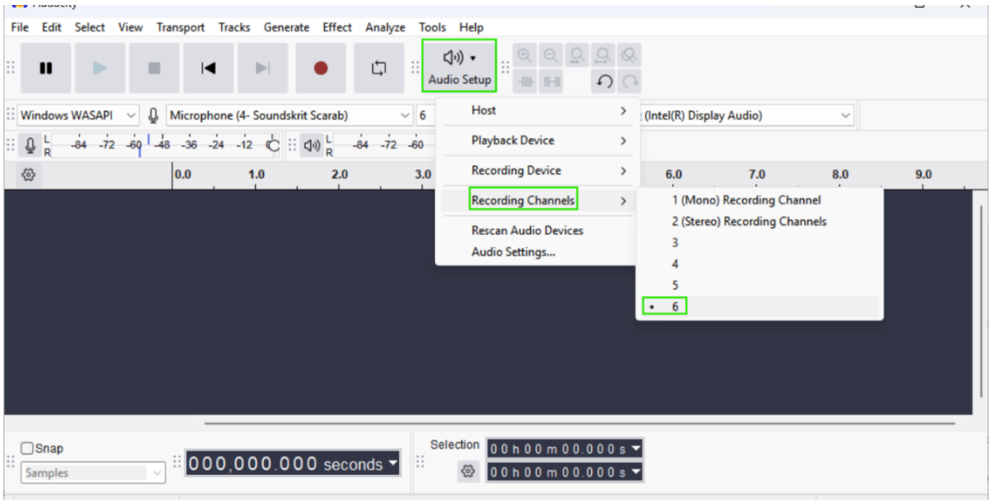


Figure 13: Audacity channel selection.

When you take a recording, all 6 channels will be recorded simultaneously. The microphone gain might be initially very low, this is on purpose the default to avoid signal clipping. You can either increase the recording gain in Windows from the default 54% to a level better suited to your recording situation or amplify the signals in Audacity. To do so, select all tracks by pressing CTRL+A and apply the same gain value to them by selecting in the top menu Effect → Amplify. In the pop up, select a gain value below the clipping threshold, e.g. by setting the “New Peak Amplitude” to -3 dB.

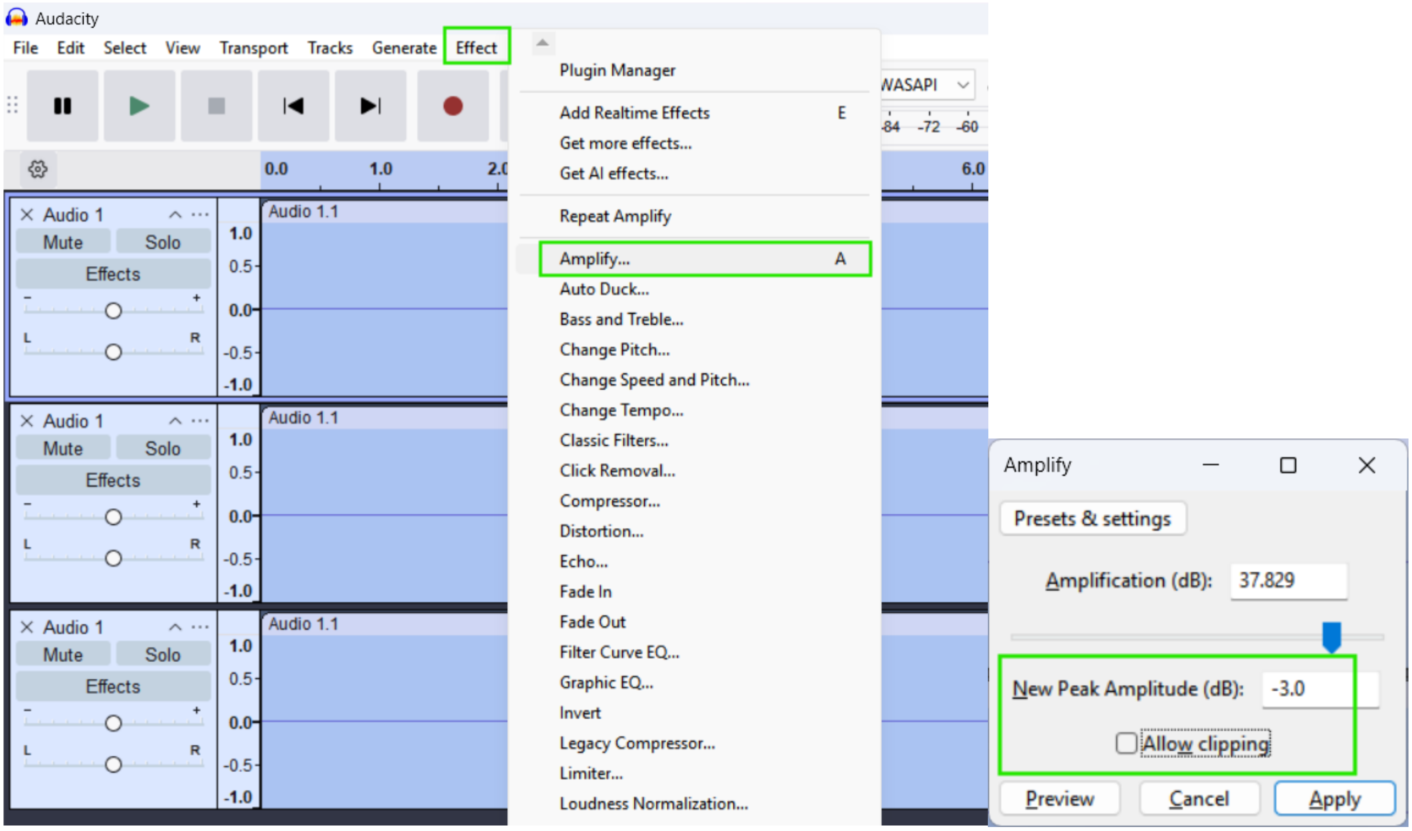


Figure 14: Applying gain to all recorded tracks.

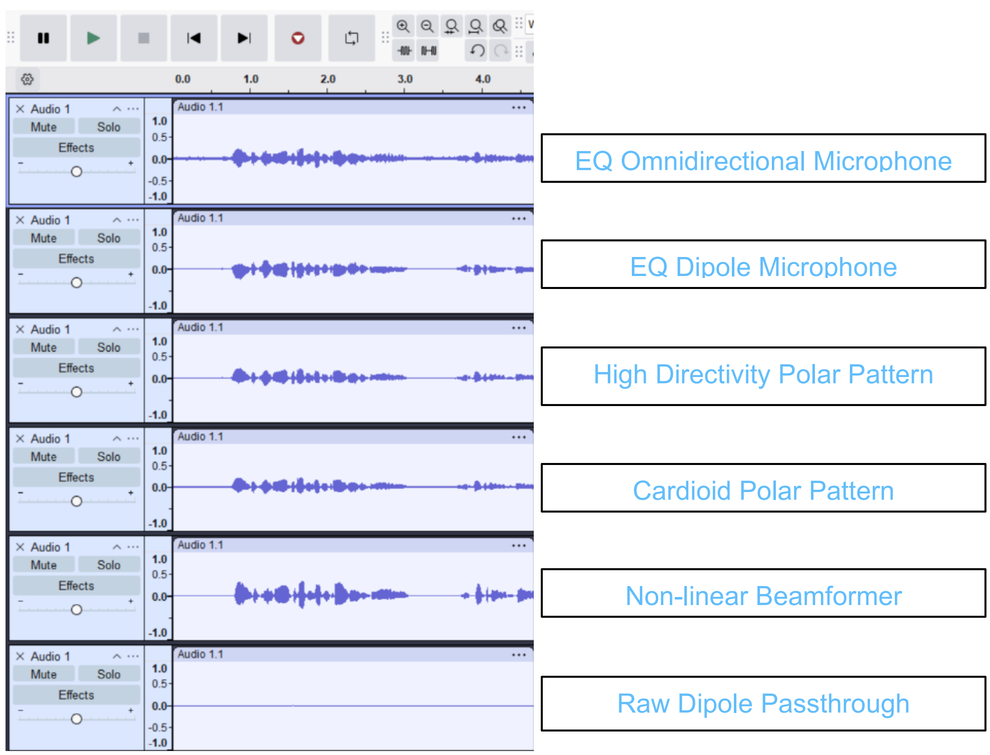


Figure 15: Output channels.

Additional Support

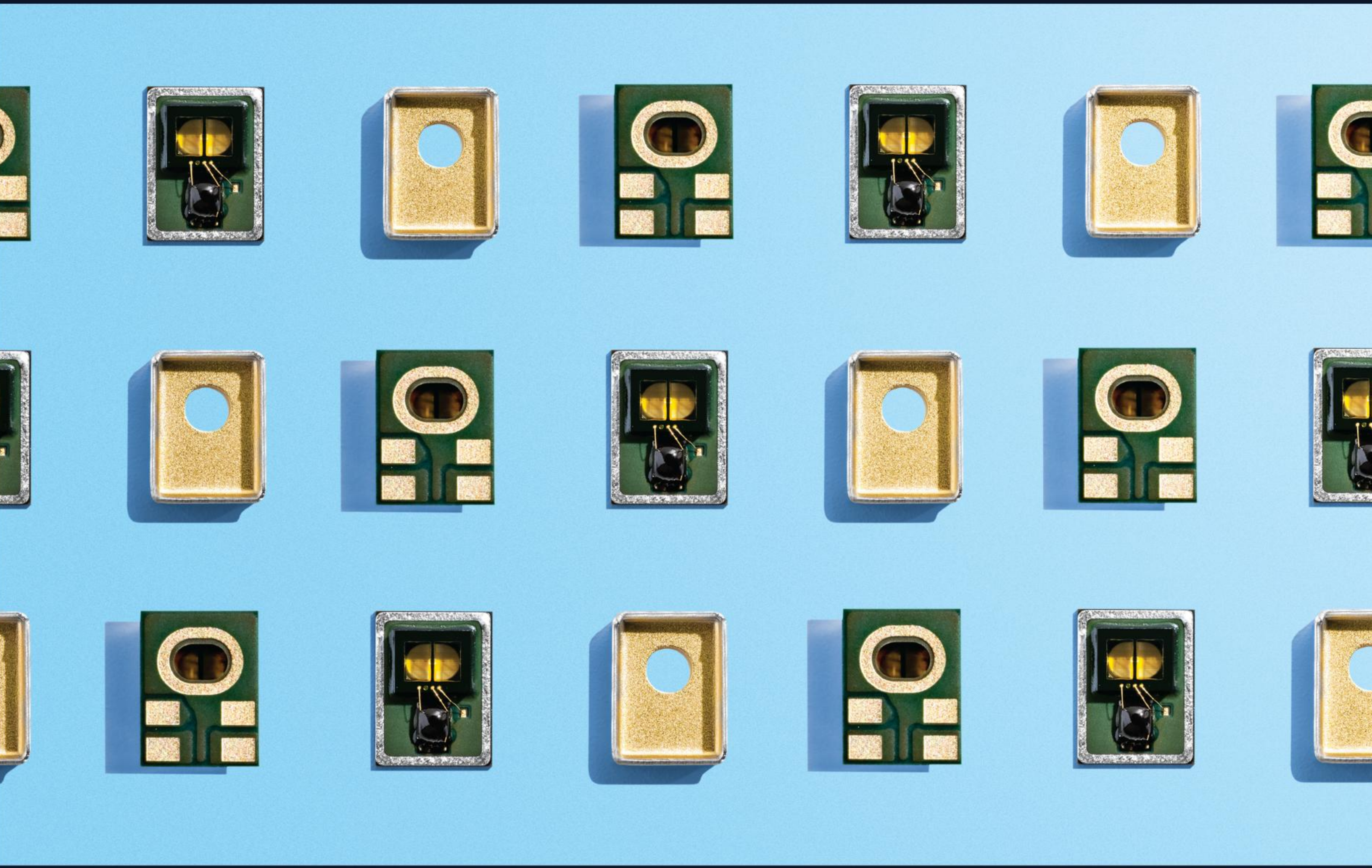
For further information on Soundskrit’s products, visit our website at http://www.soundskrit.ca where you can find more application notes, datasheets, and purchasing information. If you have any questions or need technical support, please reach out to applications@soundskrit.ca.

Revision History

Revision Label	Revision Date	Sections Revised
-	November 2024	Initial release
A	January 2025	Updated Output 5 polar plots
B	January 2025	Updated to reflect FW changes and new PARDI board
C	February 2025	Minor edits to wording

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.



soundskrit

Hear the impossible.