User manual

spc mic

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Introduction

Welcome to the spcmic user manual! The spcmic is a high-resolution 3D audio capture device which is ideally suited for a multitude of applications. Using a very large number of high-quality MEMS microphones, its characteristics can be controlled in software. This is an evolution of the ambisonic technology that was pioneered in the 1970s. The much larger number of sensors improves the spatial resolution as well as the signal-to-noise ratio of the device. Therefore, depending on your needs, the spcmic can be as quiet as the quietest studio microphones, as directional as the most directional shotgun microphones, as realistic as the best dummy heads or as versatile as a 3rd order ambisonic microphone.

Theory of operation

The spcmic is an essentially flat object with 42 microphones on each side. To avoid confusion, let us call them "sensors", and the overall array the "microphone". Each sensor is omnidirectional. It is only when the signals from all sensors are combined that you can control the directivity in any useful way.

To get an intuition about how this works, first consider the special case of a sound wave travelling parallel to the plane of the array. This sound wave will simply pass by the microphone almost without noticing it. The only scattering happens on the edge of the microphone. The edge is just 6 mm wide, much shorter than the wavelength for all but the very highest frequencies, so the microphone is barely visible to these sound waves. All sensors therefore capture the same signal, only at slightly different times. The array then behaves as a phased array, similar to what you find on military radars or beamforming satellite antennas.

Now consider a sound wave travelling at 90 degrees to the plane of the array. Two phenomena are intuitively important in this case. First, the sound hitting the front surface of the microphone will be reflected. The sound reflection causes a 6 dB amplification of the signals, since the

sensors on the front surface will sense the sum of the incoming wave and the reflected wave. This is the same effect used by boundary microphones.

Some of the energy in the sound wave passing next to the microphone will be diffracted and reach the back side of the microphone. The farther away from the edge a sensor is, the more the signal will be delayed. Generally, the signal also gets weaker as you move away from the edge. The precise shape is not simple, but it is perfectly predictable.

Sound coming from other directions can be described with combinations of these phenomena. The precise, general and important thing to remember is that the different sensors will all hear the same sound, but with a different delay and a different amplitude. The precise delay and amplitude in each sensor depends on many factors; the frequency, the placement of the sensor, the shape of the microphone, even the air temperature. However, all of these are known to the software that processes the signals (the spcmic contains a thermometer). There is only one unknown factor: The direction of the sound wave. For every possible direction, for every frequency, and for every sensor, the software knows which combination of delays and amplitudes will be observed.

With knowledge comes control. Knowing the direction-dependent response makes it possible to control the direction-dependent response. This is the process known as beamforming. Beamforming is not limited to producing narrow beams, although that is where the name comes from. You can ask a beamformer for any shape you want – that could be a cardioid response, an omni response, a narrow beam, a spherical harmonic or anything else – and the beamformer will tell you how you should delay and amplify the signals from each of the 84 sensors so that when you add them together, you get the response you asked for. A small detail is that the delay and gain are frequency-dependent, so you generally need to apply a different gain and delay at different frequencies. This is done with filters. In fact, this is the definition of what a filter does. If you pass each of the 84 signals through the correct filter and then add up the outputs, the resulting signal will have the response pattern that you asked for. Not only that,

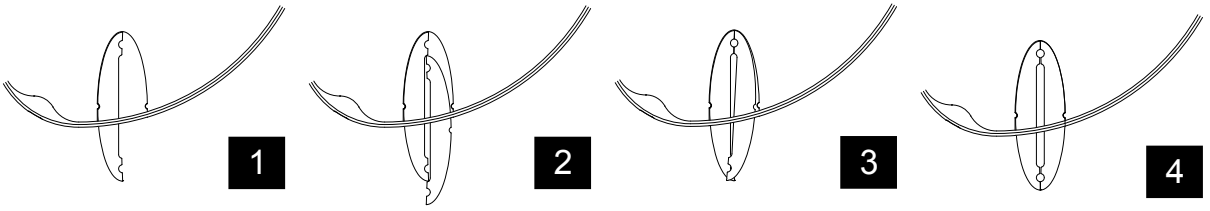
it will also have less sensor noise than any of the individual sensors and a flat frequency response. Unless you asked the beamformer for something else, of course.

Specifications

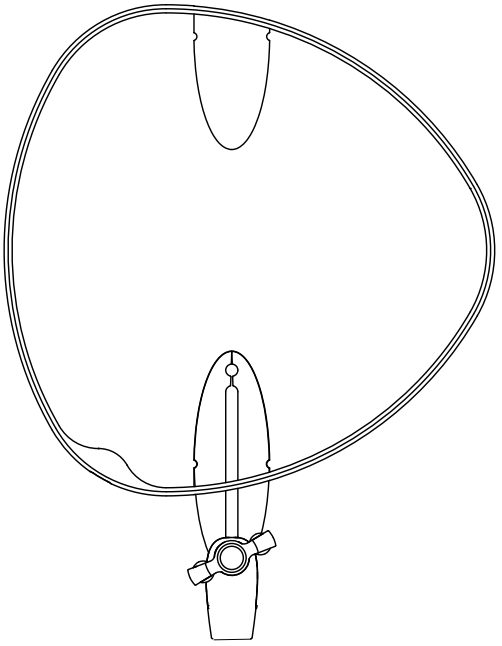
MEMS microphones	84
Individually calibrated	Yes
Frequency range	20 Hz – 20 kHz
Equivalent self noise	3 dB-A / 12 dB CCIR (in ultra-sensitive mode)
Acoustic overload point	130 dB SPL
Temperature compensation	Yes
Orientation sensor	Yes
Connector	USB-C
Diameter	170 mm
Depth	5.6 mm
Latency	3 ms
Ambisonic resolution	3rd order 3D
Beamforming resolution	5th order 3D
Field recording app	Mac OS, Windows
Studio DAW plugin	VST, AU, AAX
Sample rate	0 – 96 kHz
Sample clock synchronization	Automatic
Driver installation	No
Bottom microphone mount	3/8”
Top camera mount	Optional
Mass	168 g
Temperature range	–45 C to +85 C

Mounts

The spcmic is delivered with a rigid microphone mount. This consists of two stainless steel blades, a plastic clip, an M4 screw and an M4 wingnut. The blades can be inserted in the mounting slot on the bottom or the top of the microphone. The blades should have a small amount of twist in them so that they make contact with the inside of the spcmic when inserted and tightened. This prevents rattle during movement.



To insert the blades, first insert one blade, then the other. To fully seat the second blade, cross it over the first one, as shown in the drawings above. A small amount of force may be required, but do not use tools, as that might impart an excessive amount of force. Finally, slide the plastic clip over the blades, insert the screw and tighten the wingnut.



The plastic clip has 3/8”-16 internal threads, which is compatible with many microphone stands. If your stand has a different thread, adaptors are readily available. Additional sets of microphone mounts can be ordered from your

supplier. A second set can for example be used to mount a light action camera on top of the spcmic.

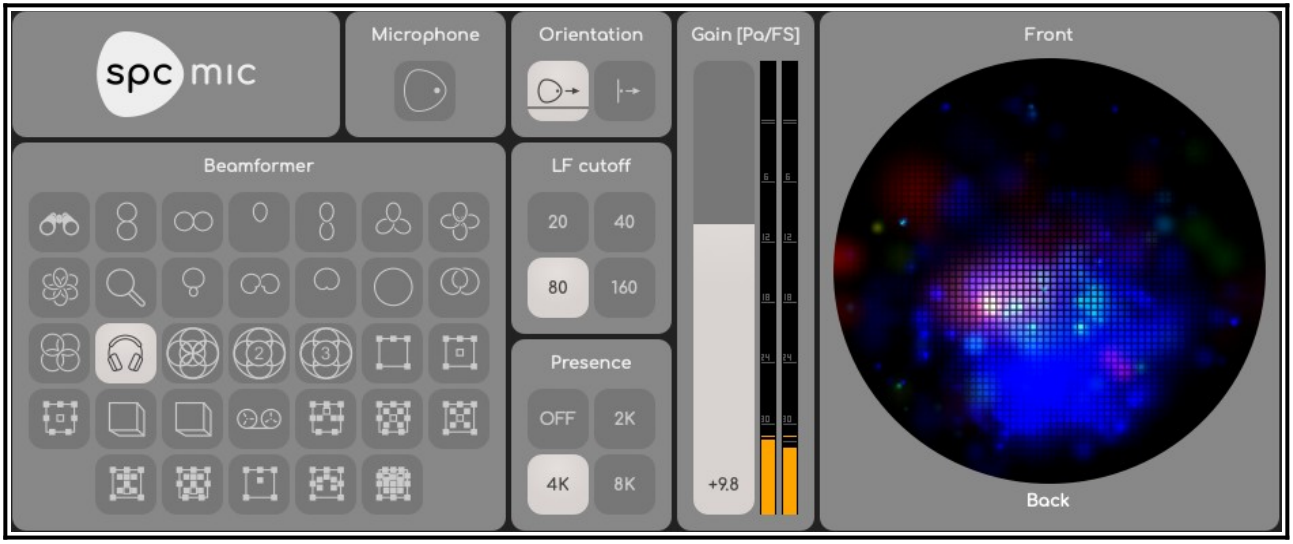
Maintenance

The spcmic does not require much maintenance. Do not let the spcmic get wet. Dirt and smudges on the fabric can be removed by applying and removing removable scotch tape. Do not use stronger tape, since this may rip the fabric off the spcmic. Over time, the edge might attain a lighter shade. It can be brought back to its original shade by rubbing it with a small amount of mineral oil. Each sensor of every spcmic which is sold is calibrated. The calibration data is either programmed into the spcmic prior to shipping or delivered as a separate download (see the spcmic app section). The spcmic can sent back for recalibration if necessary. Contact Harpex Audio GmbH for shipping arrangements.

Software

The spcmic is delivered with two programs: The spcmic plugin-in and the spcmic app. Both are available for Windows and MacOS. The spcmic app can make raw, 84 channel recordings using the spcmic and later decode such recordings. This is easy to use and provides maximum flexibility in post-production. The spcmic plug-in is used to inject real-time decoded audio into a digital audio workstation, DAW, and requires more choices to be made prior to recording. There are many cases where either of these programs could be used, and only by understanding the pros and cons of each one can you make the best choice.

Plug-in



Inserting the plugin

The spcmic does not show up as an audio device. Instead, you need to insert the spcmic plugin on a track in a DAW in order to access the spcmic. This allows you to use the spcmic together with your usual audio interface. This is an instrument plugin, meaning that it doesn't use any inputs, only outputs.

If the plugin shows "N.C." in the microphone box, it probably means that the microphone is not connected, or that the driver isn't installed. This is only an issue on Windows. To solve it, plug in the microphone, open the app and select Help → Install driver. Restart the app after installation completes.

Recording the output of the plugin

Most DAWs are set up to record the input to a track, before it is passed through any plugins. Therefore, you will normally have to route the track with the spcmic plugin into a second track and record that track. Some DAWs, like Reaper, can be set up to record the output of the plugins on a track. There is a dropdown menu on the track which by default reads "in". This must be changed to "out". Other DAWs, like Pro Tools, have special "Instrument tracks" which have no inputs. These may be appropriate to use with the spcmic plugin, but you still need to route the output to a second track for recording.

Connect the microphone before starting your DAW and keep it connected during the session.

Spatial audio format

There are many different beamformers available in the plugin. Hover the mouse pointer over them to get a little bit more information. They include higher order ambisonics, dolby atmos, binaural and many others. See the beamformer section for a list.

Frequency response

There are two settings which affect the frequency response of the microphone. The LF cutoff sets the corner frequency of a low cut filter, similar to what you will find on many traditional microphones. The default is 80 Hz. You may set it as low as 20 Hz, but depending on the beamformer you are using, that might make the microphone very susceptible to handling noise and wind noise, just as in traditional microphones.

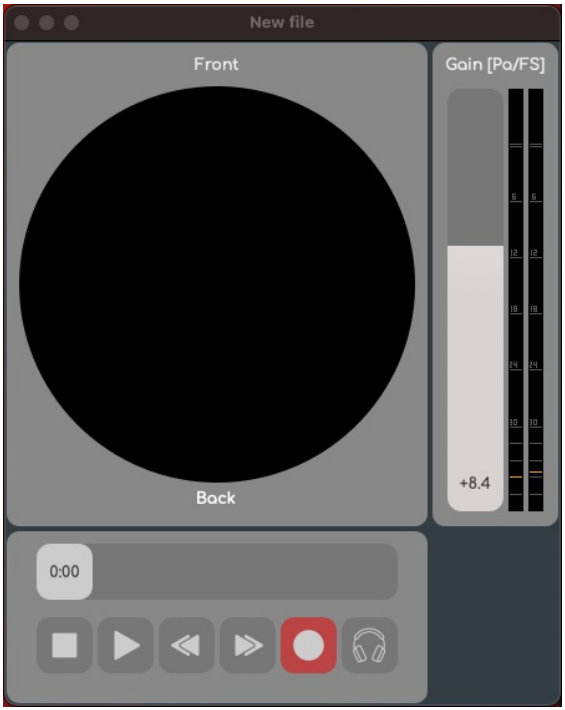
Most microphones do not have a flat frequency response, but instead have a slight peak around 2-8 kHz. This may or may not be intentional, but over the years sound engineers have gotten used to it and will complain if it is missing. It is known as the "presence peak". By default, the plugin adds a 2 dB peak at 4 kHz to mimic this. If you turn it off, the frequency response will be flat.

The LF cutoff affects all channels, including the LFE channel in surround beamformers. The crossover frequency is 80 Hz, so if you plan to make use of the LFE channel, the LF cutoff should be lower than that.

Orientation

The correct orientation of the microphone is explained in the mouseover text for the two "orientation" options. The sound will be very much the same in both orientations, but there might be slightly less noise in one or the other of the two orientations. Usually this difference is small enough that the orientation will be determined based on practical considerations like the visibility of the microphone etc. If the microphone is mounted in the wrong orientation, the orientation icon will turn red. The microphone can only detect its orientation relative to the vertical, so the orientation relative to the source is the user's responsibility.

Application



The spcmic application has several functions:

- It can decode and monitor the microphone input in real time
- It can record in raw format
- It can decode and play back raw recordings
- It can decode and export raw recordings to wav files

The application is very simple to use. It has record, play, rewind and ff buttons that work the way you would expect. It also has a monitor button, which activates real-time decoding and monitoring through an output device which you can choose.

Device driver

If the record and monitor buttons in are crossed out, it probably means that the microphone is not connected, or that the driver isn't installed. This is only an issue on Windows. To solve it, plug in the microphone, open the app and select Help → Install driver. Restart the app after installation completes.

Spatial audio format

To select the format, open the settings window (⌘ + comma on Mac, ctrl + p on windows). Here, you can select the spatial audio format. By hovering the mouse over the different options,

you get a brief description of each. They include higher order ambisonics, dolby atmos, binaural and many others. See the beamformer section for a list.

Exporting

Files saved by the application have the extension ".spcmic". These are actually 84 channel wav files, so you could open them in other audio software if you wanted to, but you will usually not. Normally, you would choose to export the recordings to another format.

Sample rate

The sample rate can be selected in the settings window. This applies to the real-time monitoring, saved files and exported files. It only applies to files created after you have changed it, though.

Block size

This only applies to the real-time monitoring. A shorter block size gives less latency, but with a higher risk of dropouts. These dropouts should not be present in the recording, though.

Gain

The gain control applies to real-time monitoring and exported files. It does not affect the saved raw files, since these files always cover the full dynamic range of the microphone. When exporting, the peak post-fader level is reported. If the peak level is a positive number, some

samples will be clipped and you need to reduce the level and re-export. The exported files are in 32 bit float broadcast wav format, so you can hardly set the level too low (quantization noise only appears below -600 dB).

Temporary directory

When recording, your audio data is saved to a temporary file until you save it or exit the application without saving. You can choose where to store these files, since they can get very large. It might be a good idea to place the temporary files on the same disk where you are planning to save your recordings. If you're using an internal disk for everything, you don't need to worry about this. The remaining recording time on the selected disk is displayed and updated as you record.

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Update calibration data

If your spcmic was delivered without calibration data, you can download and program the calibration data into the spcmic by selecting spcmic app → Update calibration... (on Mac), or Help → Update calibration... (on Windows). The calibration data for your microphone will be downloaded and programmed into non-volatile memory in the spcmic. After performing this operation, the app must be restarted.

Update firmware

You can update the firmware in the spcmic by selecting spcmic app → Update firmware... (on Mac), or Help → Update firmware... (on Windows). The latest firmware for your microphone will be downloaded and programmed into non-volatile memory in the spcmic. It is important not to disconnect the spcmic from the computer until the operation has finished, which may take a couple of minutes. After performing this operation, the spcmic must be disconnected and the app must be restarted.

Orientations

The following orientations can be selected in the plugin or in the application:

	vertical	Vertical array, logo points towards source, source in array plane
	side-address	Vertical array, logo towards source, source normal to array plane

Beamformers

The following beamformers can be selected in the plugin or in the application:

	Long range	Highly directional microphone pattern for use in noisy environments or at long range. If coloration of off-axis sound is not permissible, consider the "solo" pattern.
	Figure-of-eight front	Figure of eight, or bidirectional pattern usually used in pairs (see Blumlein setup). Since this pattern has zero side response, it can be used to suppress strong, interfering sources.
	Figure-of-eight side	Figure of eight, or bidirectional pattern usually used in pairs (see Blumlein setup). Since this pattern has zero front and back response, it can be used to suppress strong, interfering sources.
	Solo	A good all-round pattern for recording a single voice or instrument. Similar to a cardioid, but narrower and hence with a longer reach.

 Duet	A dual beam pattern for recording two voices or instruments, one in front of and one behind the microphone.
 Trio	A triple beam pattern for recording three voices or instruments distributed evenly around the microphone, one being straight ahead.
 Quartet	A quadruple beam pattern for recording four voices or instruments distributed evenly around the microphone, one being straight ahead.
 Sextet	A six-beam pattern for recording six voices or instruments distributed evenly around the microphone, one being straight ahead.
 Ultra sensitive	A specialized beam pattern with ultra-low self noise for recording very quiet sounds. If coloration of off-axis sound is not permissible, consider the "solo" pattern. For optimal results, use the side-address orientation
 Hypercardioid	A classical beam pattern with longer reach than a cardioid, but with a strong rear lobe. For even longer reach, consider the "solo" or "long range" patterns.
 ORTF close-spaced stereo	A popular stereo technique using two cardioid microphones placed 17 cm apart at an angle of 110 degrees. For mono compatibility, consider the XY setup. For 3D recording, consider the "KU100" pattern.

 Cardioid	Cardioid, or unidirectional pattern. A good general-purpose pattern with zero rear response. For longer reach, consider the "solo" pattern. For improved wind-noise rejection and bass response, consider the "omnidirectional" pattern.
 Omnidirectional	A pattern with flat and equal frequency response in all directions
 X/Y Stereo	A popular stereo technique using two coincident cardioid microphones at 90 degrees angle.
 Blumlein Stereo	A popular stereo technique using two figure-of-eight microphones at 90 degrees angle.
 KU-100 dummy head	An accurate emulation of the KU-100 dummy head from Neumann. Inteded for playback over headphones.
 Ambisonics 1st order	AmbiX (4 channels)
 Ambisonics 2nd order	AmbiX (9 channels)
 Ambisonics 3rd order	AmbiX (16 channels)
 Surround 5.0	Five full-bandwidth directional channels
 Surround 5.1	Five directional channels and one LFE channel attenuated by 10 dB.

 Surround 7.1	Seven full-bandwidth directional channels in the ITU standard channel order. The LFE channel is redundant and contains an omni signal below 120 Hz attenuated by 10 dB.
 Mach 1	Mach1 (cube)
 SPS32	Mach1 (cube)
 Raw array input	The 84-channel signal from the array, synchronized and calibrated. Temperature and orientation information is embedded in the signal.
 Atmos 7.1.2	Nine directional channels and an LFE channel attenuated by 10 dB.
 Atmos 7.1.4	Eleven directional channels and an LFE channel attenuated by 10 dB.
 Auro 9.1	Nine directional channels and an LFE channel attenuated by 10 dB.
 Auro 11.1	Eleven directional channels and an LFE channel attenuated by 10 dB.
 Auro 13.1	Thirteen directional channels and an LFE channel attenuated by 10 dB.
 IMAX 6	Six full-bandwidth directional channels.

 IMAX 12	Twelve full-bandwidth directional channels.
 Surround 22.2	Twenty-two directional channels and two LFE channels attenuated by 10 dB.