GETTING STARTED GUIDE

EM64 Eigenmike®
Microphone Array and
EigenStudio® 3

Version 1

Rev. G

mh acoustics LLC

TABLE OF CONTENTS

1 Introduction

This manual serves as a "Getting Started Guide" to mh acoustics' em64 Eigenmike® microphone array and EigenStudio® 3 software. In the next section, there is a summary of included hardware. Section 3 details the hardware setup and required software installations while Section 4, comprehensively describes the EigenStudio® 3 software. An abbreviated "quick guide" for the usage of EigenStudio® 3 can be found in Appendix A. Appendix B provides the physical feature data of em64.

2 What is in the box

The em64 Eigenmike® microphone array is packaged and shipped in a Pelican 1535TRVL Air Case. The case has a foam insert for the em64 array, and the Rycote Universal shock mount. The foam insert also has cutouts for the optional Ubiquity PoE Gb switch and associated power supplies. Also shown is the optional RedNet AM2 Dante headphone amplifier as well as a few other cutouts for an external SSD drive and ancillary small devices. The zipper pouch in the lid that can hold cables and the optional windscreen for the em64. Figure 1 shows the em64 inside the case with the above-mentioned options.

Figure 1: em64 in the supplied custom Pelican Air 1535TRVL case with Rednet AM2 and Ubiquity USW Flex PoE switch options. Note that an additional protective foam piece used to cover the em64 when packed is not shown in the photograph. Windscreens and frame as well as Ethernet cables and other items can be placed in the zipper pouches in the lid of the Pelican case.

Figure 2: Photograph of the em64 array connected to a MacBook Pro with Gb ethernet dongle and running EigenStudio3. Picture also shows the Netgear GS108PEv3 PoE ethernet switch and RedNet AM2 Dante headphone amplifier.

3 Connecting the em64 Microphone Array

The em64 Eigenmike® array uses a gigabit ethernet connection to transport the microphone array audio to a laptop or other storage device. The em64 uses Audinate's[1] Dante ethernet protocol solution. Dante is a combination of software and network protocols that delivers uncompressed, multi-channel, low-latency digital audio over a standard Ethernet network using Layer 3 IP packets. The em64 has two ethernet RJ45 jacks at the bottom of the shaft on the array. These two ethernet ports are denoted the primary "P" and secondary "S" Power-over-Ethernet (PoE) connectors. In a standard setup that does not require network redundancy, only a single PoE ethernet cable is used to the em64, and for this configuration, only *the "P" (primary) port should be used*. More on the secondary port below.

The em64 network connection requires a CAT 5e [or better] cable connected to a PoE switch. This PoE switch must be capable of supplying at least 10 Watts of PoE power at 48 Volts. The maximum length of the ethernet cable from em64 to the PoE switch is 100 meters. A photograph showing how to physically connect em64 via ethernet to a MacBook Pro laptop is shown in Figure 2.

When the em64 is connected to a powered PoE switch, two RGB LED lights along the equator on the front and back of the em64 flash red as the internal firmware begins to boot. After a few seconds, the two LEDs begin to flash green while the internal FPGA

[1] https://www.audinate.com

fully boots. When fully booted, the LEDs flash blue as the internal A/D converters are configured and loaded with calibration filters for each microphone (unique calibration filters for each microphone are stored inside each em64 array and loaded into the em64 during manufacturing). When fully booted, the two LEDs on the em64 turn solid blue indicating that the em64 is ready for use. The total boot sequence takes approximately 20 seconds.

The secondary ("S") port is for professional applications that need redundancy in the network. Dante supports a redundant network configuration in combination with an external ethernet switch chip or dual ethernet PHY. Dante redundancy uses two network interfaces, both of which are used to transmit/receive audio during normal operation. When a redundant audio connection is made between two Dante devices using textual channel labels, two streams of packets containing audio are automatically configured on both the primary and secondary interfaces. Redundancy allows glitch free audio in case of failure on one of the network paths. For networks that do not require redundancy the user should only connect to the primary network port "P" on the bottom of the em64 shaft. Please refer to the Dante documentation on how to set up Dante Controller to handle redundant networks.

3.1 Configuring the em64 in the Dante Network

Two main Audinate Dante support applications are required to set up the em64 on a Dante Network. These are the Dante Controller (DC), that controls the routing of signals on the Dante Network, and the Dante Virtual Sound (DVS) Card, which is the software interface to the PC or Mac sound system. With DVS, the em64 appears as a 64 channel in/out Dante soundcard to the PC/Mac.

The Dante Controller application is used to configure Dante devices on a Dante network. Dante Controller for Windows or macOS can be downloaded from www.audinate.com/DownloadDC.

Dante Controller is free to the user, but DVS must be purchased. Each em64 is shipped with an included coupon code that is affixed to the top right of the first page of the printed Getting Started Guide (this document). The supplied coupon code can be used to activate a DVS license for a single Windows or Mac machine from Audinate's website.

You must redeem your Dante Virtual Soundcard (DVS) token to receive a license key to activate the DVS app. To redeem your token, go to the Audinate website www.audinate.com and create a user account. After your account has been created, login, go to www.audinate.com/redeem-token and follow the instructions found there. You must be logged in to be able to redeem a token. The Dante Virtual Soundcard application for Windows or macOS can be downloaded from www.audinate.com/DownloadDVS.

DVS setup requires the user to select the correct PC/Mac 1Gb Ethernet network interface that is used by Dante. DVS allows the user to set the desired sound card latency (in milliseconds) and the number of input and output audio channels which is 64x64 for the em64 (note that the em64 in solely a "transmitter" and only outputs 64 channels). Figure 3 shows the DVS pop-up window with typical settings for the em64. Note that the IP address in Figure 3 is an autoconfigured address given by the Gb Ethernet PoE switch since the network was not connected to a network with a DHCP server. Using autoconfiguration in an isolated network allows for an extremely simple setup without the need for the user to manage a network or network switch.

Note that although Dante Controller supports redundant networks for robustness, DVS version 4.2.4 and earlier versions do not support redundant networks. mh's EigenStudio® 3 (ES3) application utilizes DVS to route Dante audio from the em64 Eigenmike array to the application (or Dante Application Library which is also based on DVS). Therefore, mh's ES3 application does not currently support redundant networks.

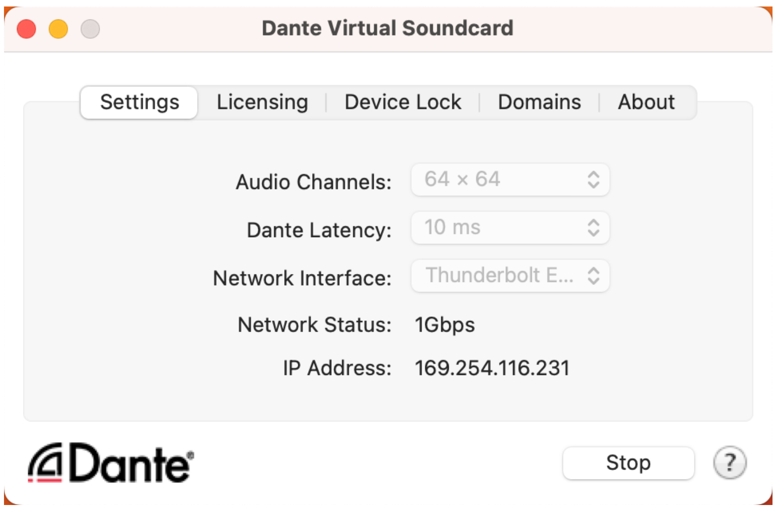

Figure 3: DVS 4.2.4 popup screen.

Figure 4 shows the Dante Controller (DC) version 4.7.1.1 UI screen displaying the signal "Routing" matrix screen (Router tab). The em64 array will show up in Dante Controller as a transmitter (since it can only transmit audio) with the name that contains the em64 serial number, (em64-000121 in the figure). If DVS is running, DC shows the name of the PC or Mac (GWES-M1-MBP in Figure 4) in both "transmitter" and "receiver" tabs. For audio to be routed to the DVS application, DC must connect the em64 to the PC/Mac DVS sound card for all 64 channels. The matrix connection in DC should be the identity matrix where microphone 1 is routed to DVS input 1 and microphone 2 to DVS input 2, and so on, for all 64 microphone channels. This can be accomplished quickly with one mouse press by hovering the mouse over the "-" (minus) icon in the expanded PC/Mac DVS receiver tab and holding the "Control" key down with a mouse click. Note that the channels have been expanded to show all output channels for the em64 and all DVS receiver channels on the Mac.

DC is the main tool for checking the routing and status of the connections in a Dante network. DC can show the clock status of all connected devices, latency, and detailed device information such as network bitrate and any transmit and receive packet errors.

For more information on the setup, control, and status of the Dante Controller and Dante Virtual Sound Card applications, please refer to the Dante Controller User Guide and Dante Virtual Sound Card user guides on Audinate's web site (note that the version number in the hyperlinks below can be different as updates are made available):

https://dev.audinate.com/GA/dante-controller/userguide/pdf/latest/AUD-MAN-DanteController-4.7.x-v1.0.pdf https://dev.audinate.com/GA/dvs/userguide/pdf/latest/AUD-MAN-DanteVirtualSoundcard_4.2.x-v1.1.pdf

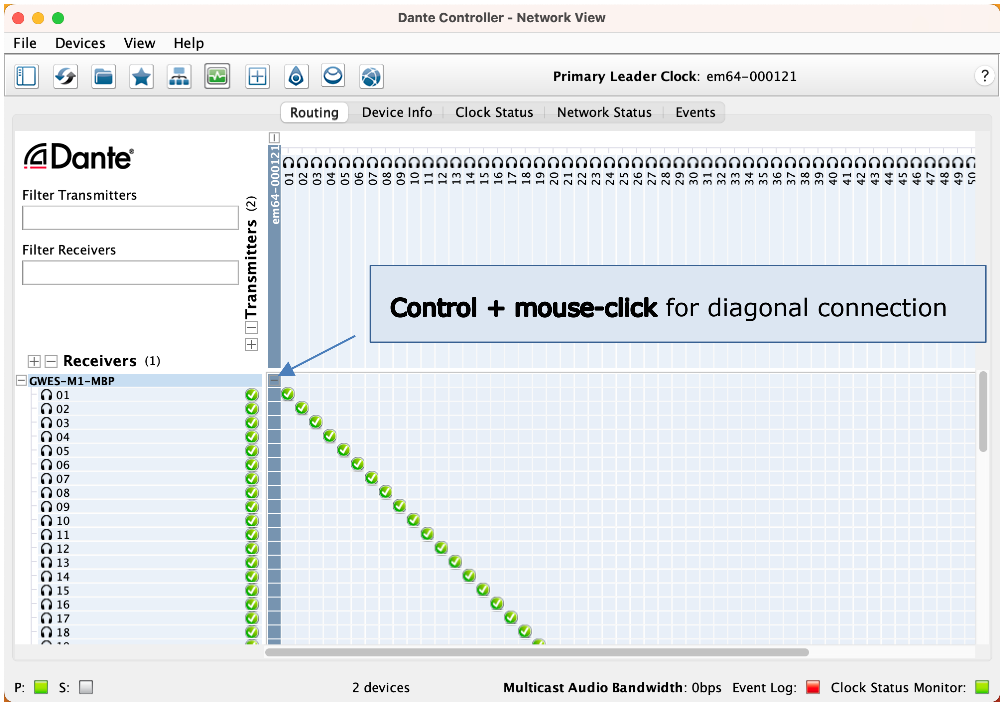

Figure 4: Dante Controller 4.7.1.1 Routing UI screen.

3.2 Installing the EigenStudio® 3 Application

The EigenStudio® 3 (ES3) application software is available on the mh website Download tab [or on USB sick if enclosed].

On macOS, it is recommended to open the .dmg file and drag the ES3 application into the Applications folder.

On Windows, the ES3 application can be installed by extracting the .zip file, double-clicking on the file `EigenStudio3Installer.exe`, and following the on-screen instructions[2]. By default, the application will be installed in `%HOMEPATH%\mhacoustics\EigenStudio\`.

Additionally, you may wish to copy the documentation included in the zip file to another folder on your PC for later reference.

[2] NOTE: In some cases, Windows Defender Smartscreen may attempt to prevent installation. It is safe to bypass this and install our application. To bypass Defender and install the application, click "More Info" in the warning screen, and then click the "Run Anyway" option.

4 EigenStudio® 3 Application Details

mh acoustics' EigenStudio® 3 application (ES3) is a stand-alone software application written by mh acoustics for the em64 Eigenmike® microphone array (em64). The application allows for recording, processing, and control of the em64.

The controls in ES3 are grouped in 5 *tabs* (shows the tab bar):

1. LIVE: This tab is only available when an em64 array is connected. It allows some control of the em64 (e.g., PGA volume) and the display of some microphone information (e.g., orientation, S/N).
2. AMBISONIC: In this tab, the user has control over the higher order Ambisonics signals, (e.g., max. order, SNR, rotation).
3. BEAMFORMER: This tab is where the user can control the beamformed outputs. (e.g., number of output beams, beampatterns, beam steering direction).
4. RECORDING: Tab gives an overview of which signals are recorded (microphone inputs, Ambisonics and/or beamformer).
5. SETTINGS: Audio device setup (e.g., audio device selection, number of channels).

The tabs in the bottom audio transport area allow for control of the *audio transport*, *file navigation*, *audio monitoring* and *level meters*. See the corresponding sections for more detail.

Figure 5: Tab bar in ES3 with the BEAMFORMER tab being the active tab.

4.1 Settings Tab

The SETTINGS tab brings up a general setup UI area that allows the user to change the audio and network settings as well as the location of the Pool directory were any recordings will be stored on the computer. Figure 6 shows the UI panel for the settings tab. The current settings for the network and audio settings are shown in the upper right of the panel. The em64 array has 64 microphones so the number of input channels should be 64. The number of output channels can vary depending on the maximum number of output beams or HOA channels that the user desires or the maximum number of hardware output channels that are available on the Dante network. The "Monitor Channels" entry allows the user to move the monitored output channel. Typically, the user will locate the two Monitor output channels to upper Dante output channels. This may be desired, so the monitor channels do not overwrite any beamformer or HOA output channels which are routed beginning at channel 1. It is recommended to set the monitor channel offset to the upper two channels on the Dante network.

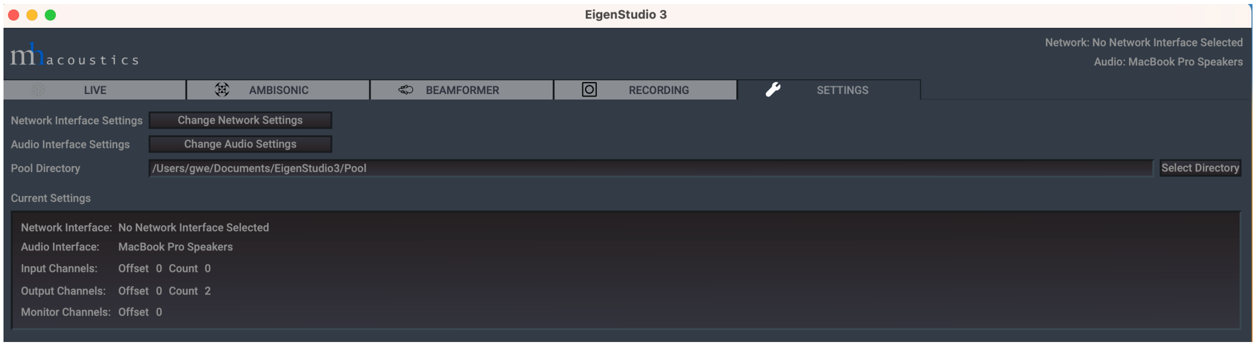

Figure 6: Audio Setting Tab section of the UI.

Figure 7: Live tab.

4.2 Live Tab

After the em64 array is connected via DVS (see Section 3.1), the user can select the Microphone tab[3] to show the signal levels for each of the 64 microphone capsules. Refer to the Section 4.6.2 to learn how to audition a particular microphone capsule.

The main feature of the Microphone tab is to allow the user to properly set the microphone input analog gain stages to maximize recording SNR in the analog-to-digital (A/D) conversion process. The “Live" tab provides three ways to adjust the input volume:

- 15 dB Pad: To allow for high SPL environments, the user can insert 15 dB of attenuation to the input analog signal path. With the pad active, a maximum SPL of 130 dB can be recorded before the physical microphone input channel begins to clip the A/D converter. Without the input pad attenuation, the maximum input SPL is 115 dB. For best SNR performance, the pad status should reflect the recording setup, i.e., enable the pad only in high SPL environments.
- PGA volume: The Programmable Gain Amplifier (PGA) allows the adjustment of the analog microphone input signal volume by 0 – 30 dB.
- Volume: Digital volume adjustments between -20 and +20 dB.

In addition to the volume controls, the Microphone tab also displays some information about the connected em64 array:

[3] Note that the Microphone tab is only available when an em64 is connected.

- S/N: This is the serial number of the connected em64 microphone.
- The em64's IP address.
- The em64's Dante device name.

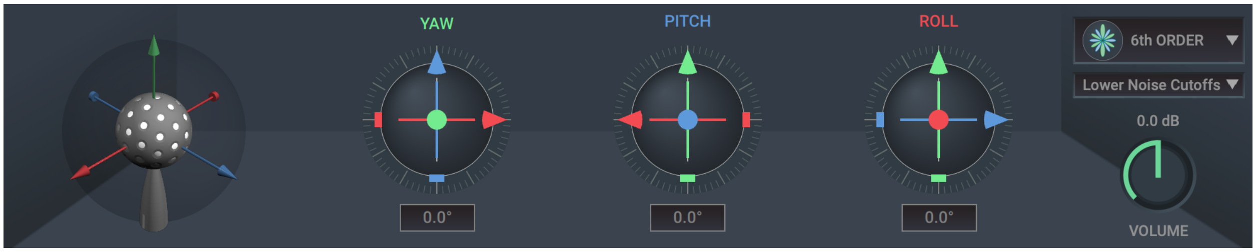

Figure 8: Ambisonics tab.

4.3 Ambisonic Tab

The AMBISONIC tab (Figure 8) provides controls for the Ambisonics Encoder (also referred to as Eigenbeamformer depending on the use context Ambisonics vs. Beamforming). The Ambisonic tab gives control over the Higher-Order-Ambisonic Signals (HOA, or Eigenbeams) generated by the Encoder. The control parameters in this tab are HOA-order, cutoff frequencies, rotation, and volume level. Except for the volume control, these parameters are described in the following subsections.

4.3.1 HOA-Order

The em64 array can generate Ambisonic signals up to 6th-order which has 49 HOA channels. The HOA-order – and therefore the number of HOA channels - can be reduced. Reasons to do so might be:

- 6th-order spatial resolution is not required for the application.
- To save storage space.
- To reduce computational load.

4.3.2 Cutoff Frequencies

An important design parameter of the Encoder (Eigenbeamformer) is the compromise between noise performance and supported maximum attainable beamformer directivity. Beamformer attainable directivity is controlled in the cutoff frequencies of the Ambisonic signals: The lower the cutoff frequency the higher the noise gain through the beamformer and the higher potential directivity. On the other hand, the higher the cutoff frequencies the lower the noise and commensurately the lower attainable maximum beamformer directivity. The user can select between two presets:

1. Normal Cutoffs: This is the nominal preferred setting. It is especially suited for normal level input signals (not low-level input signals), or for the cases when there is further processing of the signals that allows overall noise control of the output signal, e.g., Decoding or Modal Beamforming. When using the "Beamformer" tab to form the output signals, this is the preferred setting.

2. Lower Noise Cutoffs: This setting reduces output noise level and is preferred when the recorded signals are low-level and/or Ambisonic signals are further processed with 3rd party tools. Noise reduction is achieved by moderately trading off Ambisonic signal bandwidth.

In general, it is recommended to select the *Normal Cutoff* setting. When the *Lower Noise Cutoffs* is chosen the *Noise Robustness* control in the "Beamformer" tab is disabled (see Section 4.4 for details).

4.3.3 Rotation

Rotation control (Yaw, Pitch, Roll allows the user to align the orientation of the Ambisonic signals with the room, camera, or other desired coordinate system orientation. Being able to rotate the coordinate system to another orientation can simplify the setup of a Decoder, or the Modal Beamformer. Note that the em64 Cartesian coordinate system has the origin at the center of the em64. The z-axis is along the direction of the microphone shaft axis with positive z in the direction of the top of the em64. The positive x-axis points out normal from the shaft logo side. The y-axis is normal to the x and z axes and is defined by the normal Cartesian coordinate where positive y is pointing to the right when observing the em64 from the positive x-axis. Yaw is a rotation about the z-axis (rotation around the shaft), Pitch is rotation about the y-axis, and Roll is rotation about the x-axis.

The process is best explained in a simple example: Assume the em64 had to be hung from the ceiling during recording. As a result, the em64 coordinate system is upside-down relative to the room coordinate system. Without any correction of the orientation the steering in the elevation for any output beams needs to flip, i.e., steering a beam towards the ceiling requires steering the elevation down (south-pole). However, when the em64 coordinate system (which is equal to the Ambisonics coordinate system) is aligned with the room coordinate system, rotating the Ambisonic signals before the beamformer allows more intuitive steering from a room coordinate perspective, i.e., when steering towards the ceiling beam will steer up in elevation in the room coordinate system. To align the two coordinate systems, apply a Pitch of 180° (compare Figure 9). Note that the oriented em64 to the left in the Ambisonics tab reflects the rotation settings. The user can set the yaw, pitch, and roll so that results in the reoriented em64 being in the same orientation as it was during recording.

Note the color coding in the rotation control. The x-axis is red, the y-axis is blue, and the z-axis is green. Again, yaw represents a rotation around the z-axis, pitch represents a rotation around the y-axis and roll represents a rotation around x-axis. The values for the rotation can be changed in two ways: by clicking and dragging the graphics or by entering the angle directly in the edit box below the graphics. A double-click on the graphic will reset the value to 0°.

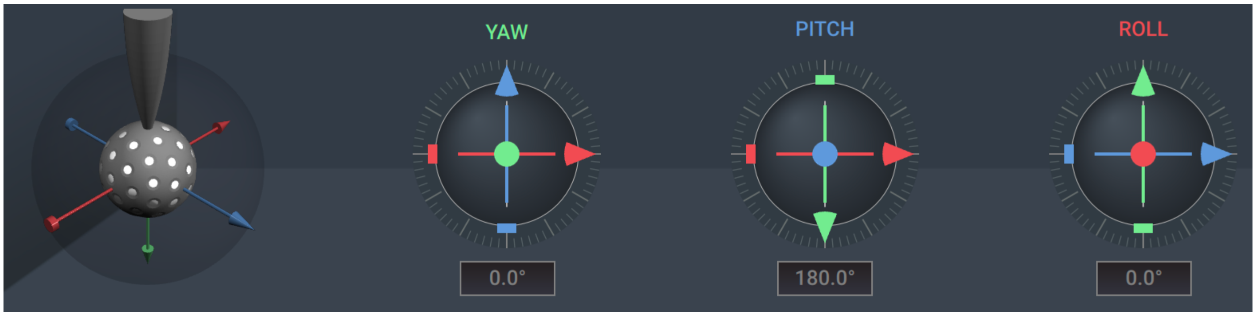

Figure 9: Ambisonic rotation with Pitch set to 180°.

4.4 Beamformer Tab

The BEAMFORMER tab opens the UI area where the user can form as many output beamformers as desired. The following predefined beampatterns are available for beamformers up to 6-th order:

- Hypercardioid – Maximum directivity beamformer for the chosen order.
- Supercardioid – Maximum front-to-back energy ratio for the chosen order.
- Cardioid – Beampattern with a single, or higher-order, null at 180 degrees for the chosen order.
- Dipole – Cosine to the power of the order beampattern.
- MaxRE – *Maximum Energy Vector Magnitude* pattern. Beampattern popular in Ambisonics that maximizes direction weighted energy response (design approximately between Hypercardioid and Supercardioid beampatterns).
- Omnidirectional – Self-explanatory.

Figure 10 shows the upper area part of the ES3 UI graphic window related to the Beamformer tab. In this figure there are two (2) defined output beams with 6th-order hypercardioid beampatterns. Note that the output beam names in the table to the right, can be customized by double tapping the mouse in the name box and entering in the desired name. Each beam has a steering direction, beampattern type (as defined above), order, offset delay and gain, as well as the Noise Robustness setting, the cutoff, and equalization filters. Control of these parameters is in the region shown in the panel to the left of the Beamformer table. The horizontal beampattern is shown in the circular graphic and the vertical steering is shown in the half circle graphic (with just an arrow to indicate steering up or down. The beampattern graphics can be "grabbed" with a mouse press and hold and then drag the arrow to the desired angle for horizontal and vertical angles. The edit boxes below these two graphic displays allow the user to type in the desired angles. In the upper left corner of the beamformer control panel, the user can select beampattern and order for the beam highlighted in the beampattern table.

The Noise Robustness knob allows the user to tradeoff the amount of noise-growth that is allowed in beamformer processing in exchange for lower cutoff frequencies of the beamformer. It should be noted that to control [i.e., lower] the amount of noise-growth in the beamformer, the maximum order used by the beamformer is progressively reduced at lower frequencies. This is due to the high-pass nature of the Ambisonic signals produced by the em64 array and the underlying physics of using spherical harmonics for sound field decomposition.

The representative HOA frequency response graphics on the lower left of the beamformer control panel gives the user some qualitative visual feedback on where the higher-order cutoff filters change the cross-over frequencies when changing the Noise Robustness setting.

The Beamformer table to the right shows the active beamformer outputs. The "+" sign allows the user to add a beam, and the trash-can, to delete a beam. The user can store the beamformer setup to a file for later recall using the file download and file upload icons. The chain-link icon allows the user to lock all beams so that if one is steered then all beams are steered relatively to the steering of any beam. The eye icon shows all beams in the beampattern graphic. Finally, to the far right, the user can select to use either a low-delay time-domain beamformer processing or a "normal" frequency domain implementation. Remember, the order is changing as a function of frequency so that noise growth through the modal beamformer processing can be controlled. However, note that the low-delay time-domain beamformer does not equalize the beamformer output for beampattern and order variation.

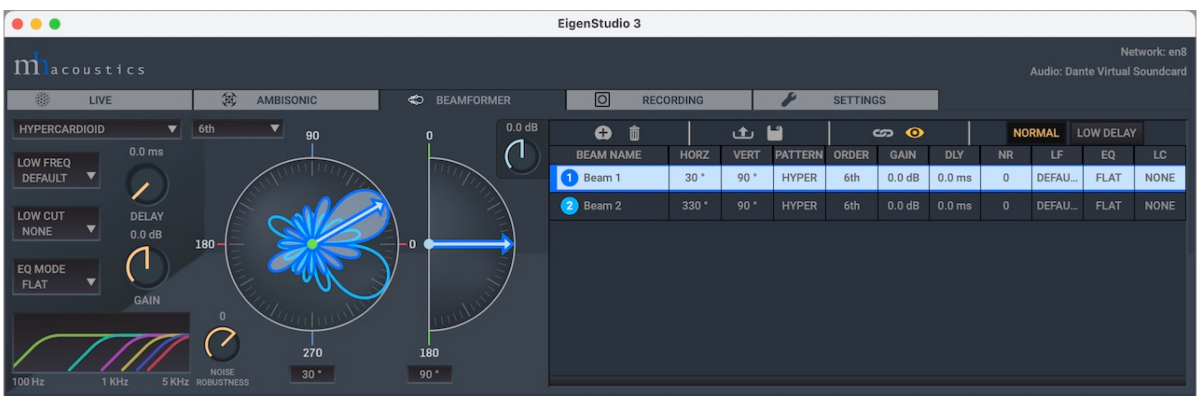

Figure 10: Beamformer tab showing 2 beamformer outputs with hypercardioid beampatterns.

4.5 Recording Tab

Figure 11 shows the upper section of the ES3 UI when the RECORDING tab is selected and active. In this window, the user can select what signal or signals to "arm" for recording by clicking the mouse in the desired select box or boxes. The choices for recording the various outputs in the selection boxes are labeled: "MIC", "AMBISONIC" and "BEAMFORMER". The [arm] box background will change color to red to indicate that the signals associated with the selected box will be recorded when the recording button is pressed in the Audio Transport section (described in more detail below). As can be seen in the Figure 11, each output has an independent digital volume control.

ES3 can record all three of these choices simultaneously. Each output is written into different multichannel output WAV files with a filename containing the recording type. Since the Ambisonic and Beamformer outputs can be generated off-line from a raw Mic recording, a user usually only needs to record the raw microphone signals. This is the most common user scenario. There may be instances where a user is interested in recording lower-order Ambisonic signals only and at the same time wants to reduce the number and size of the recorded files. In this situation, the user can select to record only the lower-order Ambisonic signals which could have a much lower channel count that the 64 microphone input signals from the em64 array. Similarly, it may be desirable to only record the output beams.

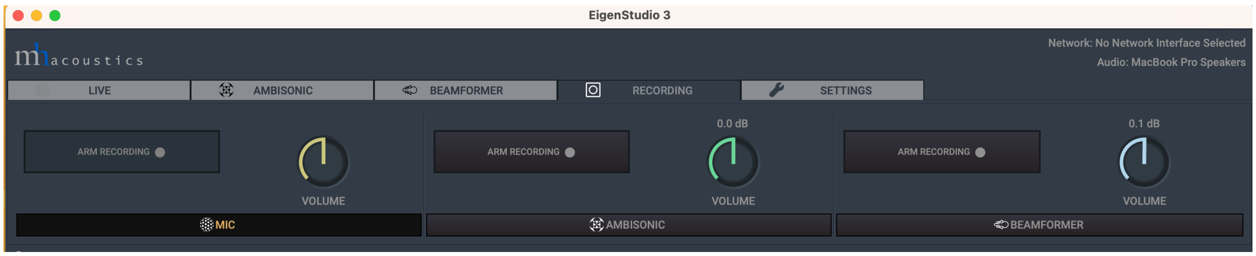

Figure 11: Recording Tab area of the ES3 UI.

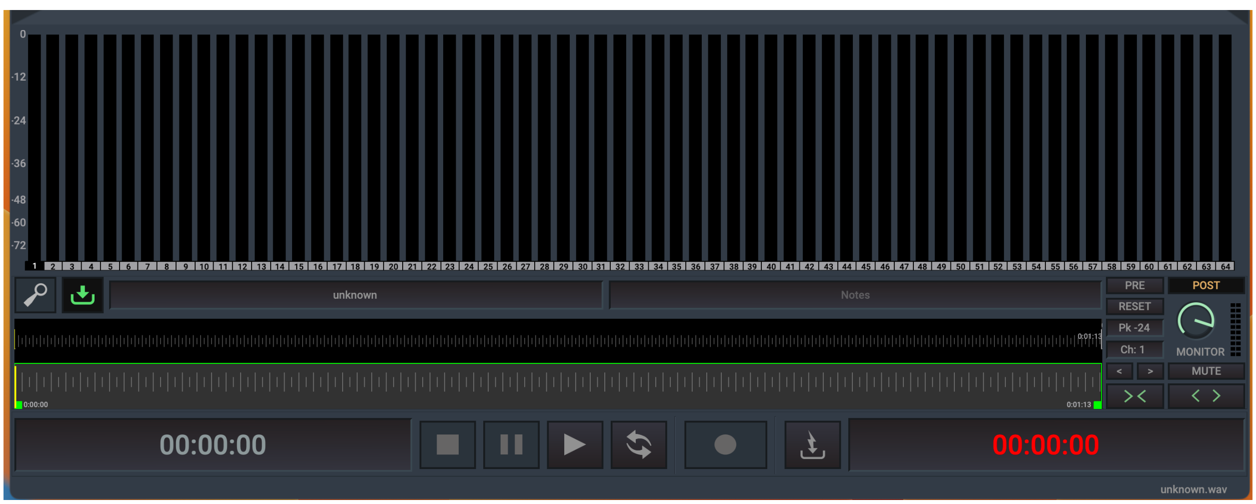

Figure 12: The "Common" section of the ES3 UI.

4.6 Common Sections

Below the tab UI section, are the "Common" sections that consist of the Level Meters, the Audio Transport, Monitor, and File Navigation. Figure 12 shows the "Common" section that is in the lower half of the ES3 main window for all the tabs except the Settings tab. The following sections describe the individual sub-UI components of this larger UI panel.

4.6.1 Audio Transport Section

The Audio Transport section is at the bottom of the UI panel shown in Figure 12. There are two timer counter panels shown in this section. The left timer indicates the current cursor location in the timeline UI section above the Audio Transport section. The red timer on the right indicates the amount of time from the start of an active armed recording. In between these two timer counter panels are the standard audio transport controls; where the square icon means "stop", the two vertical parallel lines means "pause", the triangle means "play", the two arrows pointing in different directions means "loop" and the circle means "start recording". The icons will change brightness when they are pressed and activated. The down array lightning-bolt arrow pointing into a U-shaped icon, is used to enable processing of recorded microphone signals as fast as possible and render Ambisonic and/or Beamformer outputs as fast as the machine can process (much faster than real-time). During fast rendering, the VU meters and timeline cursor position graphics are not updated.

4.6.2 Monitor Section

The user may want to monitor the audio streaming from the microphones, Ambisonic signals, or the Beamformer signals in realtime. This can be done in the Monitor section which is at the far right of the "Common" UI panel. The user can listen to "Pre" or "Post" digital gain stages for the various signals by selecting these buttons with a mouse press. The monitored channel depends on what tab is open in the upper UI window (Mic, Ambisonics, Beamformer) and what channel is selected in the VU meters or input into the Monitor channel text exit box. The monitor output is dual channel and is diotic (same signal in both channels) for headphone listening.

On the Mic tab, the user can monitor the individual input microphones by either clicking the mouse in the channel VU meter or entering the channel in the "Ch" edit box, or by using the left/right arrow buttons to sequentially move up or down the channels. The "Pk" text box shows the Peak levels for the selected channel and computes and holds them for all channels. The "Reset" button resets all the Peak levels. The Monitor section allows the user to change the Monitor output level or to Mute the output.

Note that the Monitor output should be routed using an offset in the Settings tab so that the Monitor signal does not overwrite a Beamformer or Ambisonic output (which can themselves be offset but default starting with Output channel 1). <u>If the Monitor channel is not offset and starts at channel 1, then the Monitor signal will overwrite any signal that is routed to output channels 1 and 2</u>.

4.6.3 File Navigation

The "File Navigation" panel just above the timers provides graphical display of the overall recording timeline and the current position cursor in the audio file during playback. The panel allows the user to select specific audio sections, to move to a certain position, and more. There are two timeline traces, one over the other, that are shown in this part of the UI. The lower timeline shows full length of the recording. The two green flags on the lower timeline; one on the left and one on the right, allow the user to select a section in the overall recording. The selected section is then shown in the upper timeline plot. Clicking anywhere inside the selected area will update the current position cursor to that location.

To the right of the bottom timeline are two sets of green arrows that allow for zoom-in and zoom-out of the audio timeline graphs. The zoom-in function is controlled by the set of green arrows that point towards each other ><. The zoom-out function has two green arrows that point away from each other <>. Selecting the zoom-in icon sets the bottom timeline to show only what is between the selection indicated between the left and right green time select flags. To switch back to the full view of the recording, click the zoom-out arrows (<>) and the bottom timeline with return to the full timeline view.

4.6.4 Level Meters

The level meters are standard VU level meters with attack and decay times to allow the user to visualize how the A/D converters are being driven on the Mic inputs, and the output levels from the Ambisonic and Beamformer processing. The level meters are useful to allow realtime adjustment of the microphone input PGA gain and/or 15 dB Pad on the inputs for very high sound level environments.

5 Appendix A – Essentials Guide to ES3

5.1 Configuring EigenStudio® 3 for em64 Input

The following appendix section provides an abbreviated guide on how to set up and record audio from the em64 using EigenStudio® 3 (ES3). For a more details, see Section 4 above. It is assumed the Dante Controller, Dante Virtual Soundcard, and the EigenStudio 3 application all have been installed according to Section 3.

5.1.1 Settings tab

Open ES3 by double-clicking its icon in your applications folder and start by making sure the audio connections are set up correctly. This is done in the *Settings* tab.

5.1.1.1 Configuring Audio Connections

The "*Current Settings*" field displays the active audio device[4] and channel routing information:

- Audio Interface: Name of the audio device used for audio input and output. To receive audio from a connected em64 the audio interface must be the Dante Virtual Soundcard (DVS). When working on recorded data (audio input data is provided by file) any audio device with output channels can be selected.
- Input Channels: Number of input channels (count) and the channel index of the first channel (offset). For audio input from a connected em64 the number of channels must be 64. In general, the offset depends on the configuration of the Dante network or other audio device. However, since the DVS only supports 64 channels, the offset should be set to 0.

[4] Note that the current audio device is also displayed in the top right corner of the UI.

- Output Channels: Number of output channels (count) and the channel index of the first channel (offset). Note that this is the number of physical output channels that will be used, e.g., for audio playback. This channel count can be different from the number of beamformers in the "BEAMFORMER" tab.
- Monitor Channels: The channel index (offset) of the monitor channel. There are always two monitor channels containing a single channel output (for diotic headphone listening)

Note that the monitor channels occupy two physical output channels and can therefore overlap with the output channels. If the monitor channels and the output channel overlap, the monitor channel will take precedence. The user should offset the monitor channel to a high channel to avoid any potential confusion. The audio settings can be changed by clicking on the "Change Audio Interface" button[5]. This will open the "Audio Interface Setup" dialog.

5.1.1.2 Setting the Pool directory

All recorded data is stored in the Pool directory. The default directory is Documents/EigeStudio3/Pool in the users' home directory. This directory can be changed by clicking on the "select directory" button.

5.2 Recording the Raw Microphone Signals

When making recordings with the em64 it is always recommended to record the 64 "raw" microphone inputs. By storing the "raw" signals the user will be able to benefit from future modification, improvements, or additions to the signal processing chain.

The following sub-sections describe a step-by-step process to record the "raw" microphone inputs.

5.2.1 Adjust the Input Gain

Navigate to the "LIVE" tab (note that this tab is only available when an em64 is connected). The level meters indicate the signal level relative to full-scale levels (FS). For best Signal-to-Noise-Ratio (SNR), set the input gain so that the signal level is appropriate signal (i.e., roughly –20 dB re FS). The "LIVE" tab offers three gain stages (Figure 13), two of which are analog gain adjustments, and one is a digital gain (see Section 4.2 for more details). The gains should be adjusted clockwise (turn from left to right).

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[5] The Audio Interface Setup dialog can also be accessed by clicking on the current audio interface name displayed in the top right of the UI.

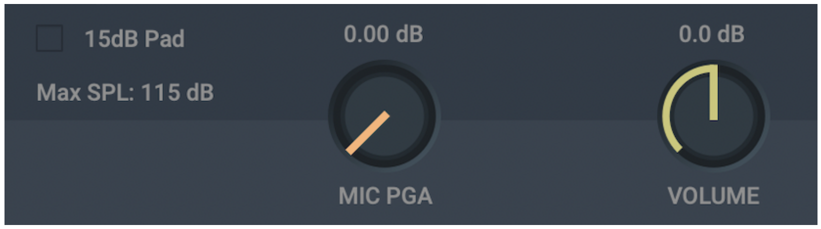

Figure 13: Gain stages in the "Live" tab.

1. 15dB Pad: For high SPL (peaks exceeding 115 dB), the user can apply a 15 dB attenuation pad in the microphone input gain stage. Applying the pad allows a higher maximum Sound Pressure Level (130dB SPL) with less distortion. If the SPL is not expected to exceed 115 dB SPL, the 15 dB attenuation Pad should be disabled for best SNR recording.
2. Mic PGA: The Programmable Gain Amplifier allows changes in the input gain between 0 dB and 30 dB. The PGA gain is applied to the signal before A/D conversion.
3. Volume: The digital volume control allows further gain adjustment for the input signals after A/D conversion.

5.2.2 Listening to the Input Signals

While adjusting the input gain it is useful to audition the input signals. This can be accomplished through the monitor section of the UI (details in Section 4.6.2). Make sure that the Dante channel routing is set up according to Sections 3.1 and 5.1.1.

The easiest way to select a microphone channel is by clicking anywhere in the desired channels level meter. The dark-colored number in the channel numbers along the bottom of the VU meters is the one routed to the monitor output. The volume knob in the monitor section allows for level adjustment of the monitor signal.

5.2.3 Record the Signals

Once the above preparations are complete, the signals are ready for recording. Go to the MIC section (the one to the left, see Figure 14) under the "RECORDING" tab and click the ARM RECORDING button. The button will turn red indicating that microphone audio *will be* recorded. The VOLUME dial is a copy of the volume dial in the "LIVE" section as the two volume dials work in tandem. The wide buttons (MIC, AMBISONICS, BEAMFORMER) set the level meter source. With the MIC button active the level meters show the raw microphone input levels.

The last step to finally begin recording is to click the *record* button in the "Common" section panel. This button is at the bottom of the UI (see Section 4.6), together with the other audio transport buttons. The red colored record timer in the lower right of the UI will now begin displaying the time into the recording.

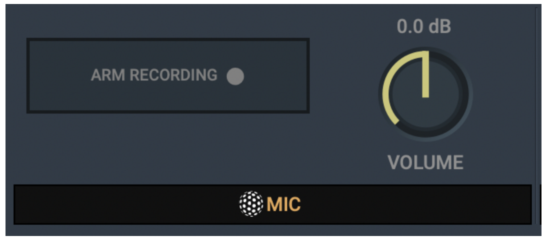

Figure 14: Microphone section in Recording tab.

The signals are recorded to a file in the pool directory (see Section 5.1.1.2 for setting the pool directory). The file name is displayed in the edit box below the level meters. The default file format is "ES3_*date_time*". The files are stored as multichannel WAV files.

5.3 Rendering the Ambisonic Signals

The AMBISONIC tab provides controls for the Ambisonic Encoder (also called Eigenbeamformer, depending on the context Ambisonics vs. Beamforming). The following sub-sections describe a step-by-step process to record the Ambisonics signals.

5.3.1 Select the Input Signals

Input signals required to render Ambisonic signals can be live "raw" or previously recorded "raw" signals from an em64. Figure 15 shows the input source selection buttons. Selecting the Eigenmike icon will rout the live inputs to the Ambisonics (Encoder) section while the tray symbol will trigger a file input dialog. After loading the file, the signals must be routed from the selected file to the Ambisonics section.

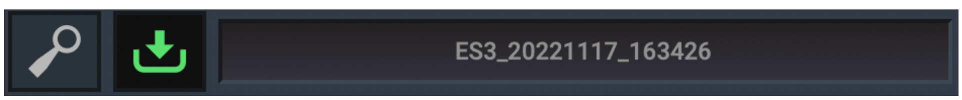

Figure 15: Input source selection. Here the input from file is active.

5.3.2 Adjust the Ambisonics Settings

Make sure the "AMBISONIC" tab is open by clicking on it. There are four settings: Orientation, order, cutoff (frequencies) and volume. The orientation is adjusted via the Yaw, Pitch, Roll dials. This can be useful to compensate for any physical rotation of em64 during the recording (see Section 4.3.3 for details). The Volume dial applies a digital gain to the Ambisonic signals before they are recorded.

The main settings in this tab are the ambisonics order and cutoff (frequencies). em64 supports High-Order-Ambisonics (HOA). Setting controls are shown in Figure 16. The order of the Ambisonics signals determines the number of channels recorded and will indirectly affect the resulting file size. You should set the HOA-order to whichever order you plan to use for your further processing.

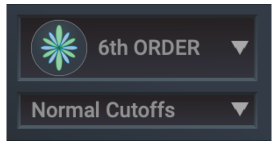

Figure 16: Ambisonics tab with setting controls of Order and Cutoff (frequencies).

The cutoff frequencies affect the SNR in the resulting Ambisonic signals. ES3 provides two configurations:

1. Normal cutoffs: This is the nominal preferred setting. It is especially suited for loud input signals, or for the cases when there is further processing of the signals that allows overall noise control of the output signal, e.g., Decoding or Modal Beamforming. When using the "BEAMFORMER" tab to form the output signals, this is the preferred setting.
2. Lower Noise Cutoffs: This setting reduces output noise level and is preferred when the recorded Ambisonic signals are further processed with 3rd party tools. Noise reduction is achieved by moderately trading off Ambisonic signal bandwidth.

For more details on the order and cutoff setting see Section 4.3.1 and 4.3.2.

5.3.3 Record the Signals

Refer to Section 5.2.3 for recording the signals. The only difference when recording the Ambisonic signals is to click on the ARM RECORDING button in the AMBISONIC (center) panel of the RECORDING tab instead of arming the microphone panel as done in the previous section.

5.4 Rendering a Beamformer Output

The BEAMFORMER tab provides the controls for the Modal Beamformer. A Modal Beamformer is equivalent to an Ambisonics Decoder, but it has ability to control each beam independently. The following sub-sections describe a step-by-step process to record Beamformer outputs.

5.4.1 Select the Input Source

Input signals for the beamformer can either come from a file that contains previously rendered Ambisonic signals, or from "raw" sensor signals that are processed by the Encoder [Eigenbeamformer]. Note that the Ambisonics tab becomes disabled if Ambisonic signals are loaded from file. See Section 5.3.1 on how to load a file and file versus live "raw" input signals. It is also possible to load the file after all the beamformer adjustment are done.

5.4.2 Add a New Beamformer Output Channel

Make sure the BEAMFORMER tab is active. A beamformer channel can be added (and deleted) via the global beamformer controls (compare Figure 17). These controls affect the overall beamformer section rather than just a single channel. To add a channel, click the "+" icon. A new channel will be added to the beamformer channel list. The new channel will get a default name and the channel-specific settings are copied from the currently active (highlighted) channel. You can change the name by double-clicking in the "Beam Name" field. To remove a channel, you first select it and then click the "trash" icon.

Figure 17: The global beamformer settings.

For information on the other operations, refer to Section 4.4. Normal vs. Low Delay, will be explained below in Section 5.4.4.

5.4.3 Configure the Beamformer Output Channel

All channel settings are listed in the beamformer channel list. To adjust those setting select the channel you want to adjust. The UI part to the left of the table now displays the settings for this channel. The available settings are:

1. Noise Robustness: For robustness reasons, all modal beamformer patterns transition towards lower orders with decreasing frequencies. The graphic next to the *Noise Robustness* dial (compare Figure 18) shows abstracted highpass filters to provide an idea of the transition frequencies for the different orders (1 to max. order). A higher *Noise Robustness* will result in higher transition frequencies and therefore a reduction in the Directivity Index. On the other hand, a lower *Noise Robustness* will result in a higher Directivity Index but also in higher noise levels (lower SNR). Note that the Noise Robustness setting will be inactive if "Low Noise Cutoffs" are selected for the Ambisonic signals (compare Section 5.3.2). That is because the "Low Noise Cutoffs" setting results in a similar or better noise robustness than the Noise Robustness control in the modal beamformer can provide.
2. Low Freq: This is the low frequency behavior. The user can choose between three modes:
    a. Default: This setting honors the *Noise Robustness* at lower frequencies and might result in the transition of the current pattern to omni at lower frequencies. See the *Noise Robustness* setting above for more information.
    b. Pad: Like a) but it will also add some attenuation to the omni band. The amount of attenuation is equivalent to the Directivity Index of the first order pattern.
    c. 1^{st} Order: This setting will keep a first order pattern at low frequencies and ignore the *Noise Robustness* setting. As a result, this beamformer channel might be noisier. This might be partly corrected by changing the *Low Cut* setting by avoiding the very low frequencies.
3. EQ Mode: Most beamformer beampatterns will increase in beamwidth towards lower frequencies to control noise gain through the beamforming operation. As a result, the Directivity Index of these patterns will decrease. This can result in "boomy" sound especially in reverberant environments. Therefore, the frequency response of the beamformer can be adjusted. The three options are:
    a. Flat: This setting results in a flat on-axis response in an anechoic environment.
    b. Semi Diff: This is somewhere between Flat and Diffuse.
    c. Diffuse: This setting results in a flat response in a diffuse environment and commensurately exhibits high-pass behavior in an anechoic environment.

4. Gain and Delay: This knob controls the Gain and Delay for the selected beamformer channel. These settings can be used to compensate for a non-ideal playback speaker setup.

Figure 18: Noise Robustness control in Beamformer tab.

For more details on these and all settings see Section 4.4.

5.4.4 Adjust the Global Beamformer Settings

The two global beamformer settings that influence processing of the signals are:

1. Beamformer Mode: The controls for the beamformer mode are two buttons to right of the "global beamformer settings" (compare Figure 17):
    a. *Normal*: The *Normal* mode applies a Filter & Sum beamformer to the Ambisonic signals. The frequency dependent processing allows to compensate the frequency response and the directivity as the orders decrease towards lower frequencies.
    b. *Low Delay*: This mode applies a time domain Weight & Sum beamformer to the Ambisonic signals. Due to the computational simplicity of the time domain filtering operation, this mode does not add delay to the processing. However, using higher-order beamformer patterns for the output signals, the sound will be highpassed and the directivity pattern is not as well controlled. Note that not all settings are active in *Low Delay* mode.
2. Volume: In addition to the "per channel" gain setting, there is a global Volume setting that acts on all channels. This global gain is controlled by the light blue colored knob immediately to the left of the beamformer channel table.

5.4.5 Record the Beamformer Outputs

Refer to Section 5.2.3 or recording the signals. The only difference when recording the Beamformer signals is to click on the ARM RECORDING button in the BEAMFORMER (right) panel of the RECORDING tab instead of arming the microphone or Ambisonics panels as done in the previous sections.

6 Appendix B – em64 Physical Attributes

Figure 19: and Figure 20: show the microphone channel locations from the logo and mount sides of the em64. The upper half contains the lower 32 microphone channels, and the lower half contains the upper 32 microphone channels.

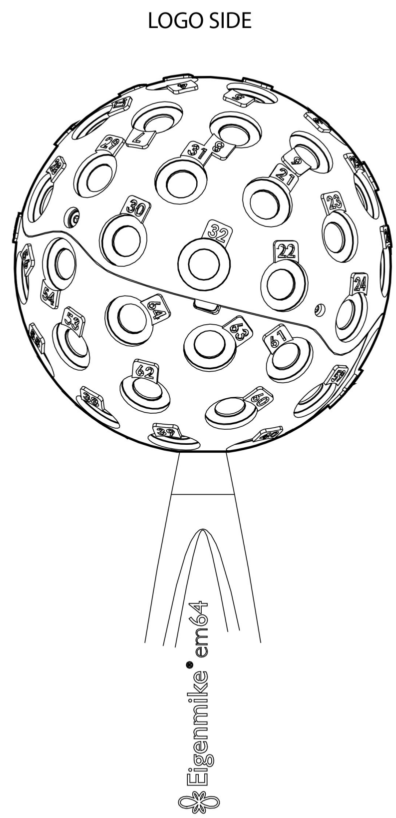

Figure 19: Logo-side Microphone numbers.

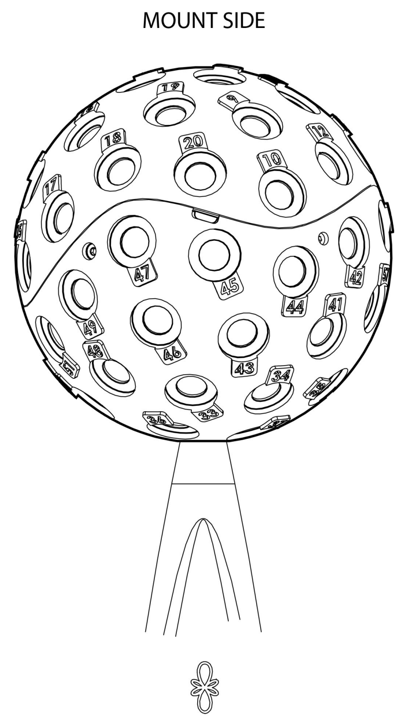

Figure 20: Mount-side Microphone locations.

Figure 21: and Figure 22: show the microphone positions from the top and bottom views of the em64. Figure 23 shows the positions of the two M3 x 0.5 mm thread tapped shaft mounting holes.

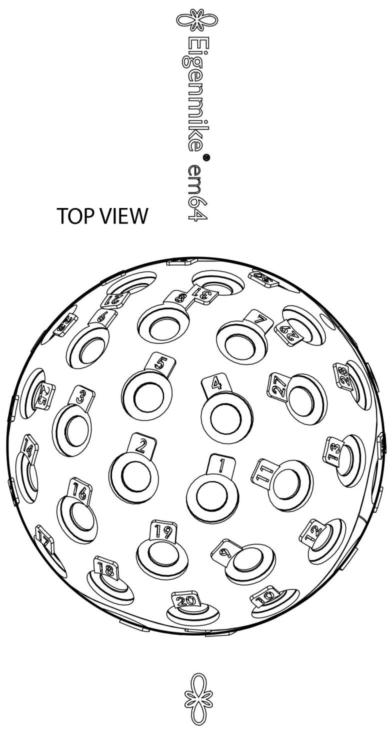

Figure 21: Top view of em64.

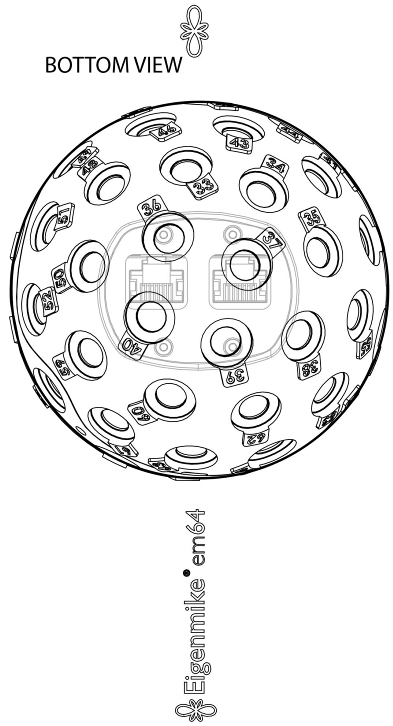

Figure 22: Bottom view of em64.

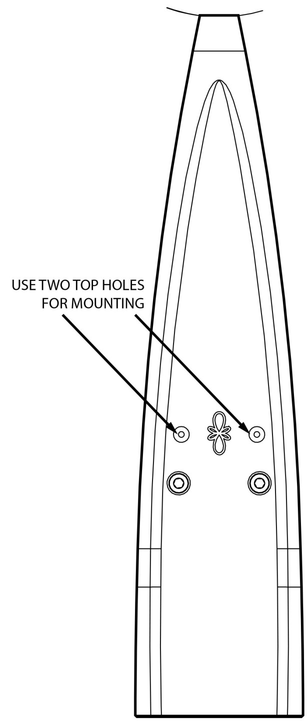

Figure 23. em64 M3 x 0.5 mm thread tapped mounting hole locations

Table 1 below contains the microphone positions relative to the center of the sphere as well as the standard physics spherical angles for the microphone locations. Theta is the angle from the positive Z-axis (the em64 shaft axis) and Phi is the angle along the equator counterclockwise from the X-axis (direction normal to the mh logo on the shaft). Table 1 also contains the quadrature weights for the 64 microphone positions.

mic	mic X (m)	mic Y (m)	mic Z (m)	Theta (degrees)	Phi (degrees)	Quad. Weight
1	-0.011557241	-0.003634259	0.040214702	16.7656	197.4561	0.954
2	-0.006821829	0.014153239	0.038950591	21.9677	115.734	0.9738
3	0.003984595	0.028035747	0.031018059	42.3941	81.911	1.0029
4	0.006624713	-0.007015439	0.040876603	13.2817	313.3592	1.0426
5	0.01180592	0.011078169	0.038754281	22.6728	43.1785	1.0426
6	0.022897066	0.02432533	0.025455896	52.6925	46.7324	1.0024
7	0.023518981	-0.010473407	0.033183809	37.806	335.9958	0.9738
8	0.027930555	0.007244038	0.030518977	43.3944	14.5398	0.954
9	-0.026528792	-0.012064556	0.030243505	43.9386	204.4547	1.009
10	-0.035377244	-0.017670872	0.014148888	70.3132	206.542	0.9932
11	-0.008872276	-0.021232674	0.035134829	33.2231	247.3219	1.0024
12	-0.02147899	-0.029365592	0.020983685	60.0257	233.817	1.0324
13	-0.003329298	-0.034854965	0.023195849	56.4763	264.5437	0.954
14	-0.006515458	0.038250201	0.01607703	67.4936	99.6669	1.0024
15	-0.010629237	0.040561896	-0.002398308	93.2735	104.6842	1.0079
16	-0.016145501	0.026952875	0.027872303	48.423	120.9227	1.0268
17	-0.024451299	0.033028326	0.008675458	78.0793	126.513	1.0151
18	-0.03154981	0.019533694	0.019673441	62.0685	148.2368	0.9463
19	-0.025073092	0.007839137	0.03277023	38.7171	162.6381	1.012
20	-0.037672903	0.000953744	0.018542999	63.8004	178.5498	1.0253
21	0.036823528	0.014335781	0.014230711	70.1946	21.2715	1.009
22	0.037594201	0.018160279	-0.004569489	96.246	25.7834	0.9932
23	0.02783995	0.030768616	0.00649841	81.0992	47.8607	1.0324
24	0.022619641	0.033418436	-0.011643021	106.094	55.9075	1.0151
25	0.012380765	0.036849368	0.015900971	67.7533	71.4285	0.954
26	0.00837538	0.041137444	-0.001250487	91.7061	78.4921	1.0079
27	0.010644033	-0.024809284	0.032174585	39.9985	293.221	1.0029
28	0.013754457	-0.036654662	0.01520693	68.7726	290.5683	1.0024
29	0.027326723	-0.024488403	0.020434489	60.8869	318.1354	1.0268
30	0.037408815	-0.018242135	0.00563958	82.2833	334.0042	0.9463
31	0.037068295	-0.005194603	0.019051444	63.0247	352.0227	1.012
32	0.041999728	0	0.000150987	89.794	0	1.0253
33	-0.028212155	0.002948551	-0.030973868	137.5166	174.0335	0.954
34	-0.022826123	-0.014665642	-0.032060679	139.7604	212.7205	0.9738
35	-0.009183428	-0.028126445	-0.029808853	135.2133	251.9179	1.0029
36	-0.012302547	0.006918823	-0.039557265	160.3628	150.6471	1.0426
37	-0.006130139	-0.010980567	-0.040073042	162.577	240.8266	1.0426

38	0.010113875	-0.023754736	-0.03312736	142.0685	293.0625	1.0024
39	0.011840051	-0.006560409	-0.039758953	161.1987	331.0098	0.954
40	0.006130139	0.010980566	-0.040073039	162.577	60.8266	0.9738
41	-0.025887652	-0.027677207	-0.018105291	115.536	226.9135	1.0268
42	-0.024678452	-0.03387427	0.002740042	86.2594	233.9255	1.0151
43	-0.036679849	-0.008899681	-0.018422388	116.0164	193.6382	1.012
44	-0.036335686	-0.020699981	-0.003902395	95.3313	209.6696	0.9463
45	-0.041935747	-0.002321812	-4.67155E-05	90.0637	183.169	1.0253
46	-0.037520459	0.010964305	-0.015362267	111.4549	163.7105	1.009
47	-0.038547052	0.016400072	0.003026973	85.8671	156.9524	0.9932
48	-0.024137073	0.02066478	-0.02746577	130.8398	139.4318	1.0024
49	-0.029473784	0.028489432	-0.00914595	102.5775	135.9729	1.0324
50	-0.00544157	0.024900315	-0.033382085	142.6375	102.3273	1.0029
51	-0.01434762	0.034551056	-0.019088482	117.032	112.5511	0.954
52	0.0044431	0.036967039	-0.019434429	117.5631	83.1464	1.0024
53	0.023110697	-0.029893349	-0.018338031	115.8884	307.7078	1.0324
54	0.026510308	-0.032575329	0.000227251	89.69	309.1392	1.0151
55	0.005300179	-0.036546219	-0.020007047	118.4478	278.2519	0.954
56	0.009406824	-0.040831477	-0.002881338	93.9338	282.9735	1.0079
57	-0.011681846	-0.038563231	-0.011849547	106.3875	253.147	1.0024
58	-0.007155393	-0.040867068	0.006533236	81.0511	260.0688	1.0079
59	0.014708897	0.025211	-0.03020023	135.9764	59.7394	1.0268
60	0.024684172	0.006257108	-0.033399704	142.6771	14.2241	1.012
61	0.030475416	0.019407585	-0.021414822	120.6556	32.4901	0.9463
62	0.027225321	-0.013234437	-0.029114115	133.8834	334.0753	1.009
63	0.037608328	0.001368636	-0.0186478	116.3591	2.0842	1.0253
64	0.036330318	-0.016888911	-0.012604466	107.464	335.0677	0.9932

Table 1: em64 microphone locations and quadrature weights.