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# em32 Eigenmike® microphone array release notes (v15.0)

Notes for setting up and using the mh acoustics em32 Eigenmike® microphone array

## Introduction

The mh acoustics em32 Eigenmike® microphone array is a 32 channel spherical microphone array that uses 32 individually calibrated professional quality 14 mm electret microphones embedded in a rigid sphere baffle. The theory behind the Eigenmike® array has been published in many technical papers and is covered by both US and international patents awarded to mh acoustics LLC. This document explains how to setup the hardware and covers some details on how to configure the hardware and software.

## Hardware description and setup

The hardware for the em32 Eigenmike® microphone array consists of three parts: 1) the spherical microphone array which has all of the programmable microphone preamplifiers and A/D conversion hardware embedded within the spherical baffle, 2) the mh acoustics Eigenmike® Microphone Interface Box (EMIB) that converts the proprietary serial data from the em32 array to a standard Firewire interface and 3) a MacBook Pro Windows® laptop with an integrated FireWire interface and ASIO software drivers and control software.

To begin the setup of the Eigenmike® microphone array, mount the spherical array on an industry standard microphone stand (a stand is supplied with the em32 Eigenmike® system.) Note that the shock-mount supplied with each em32 system can support both 3.8" - 16 and 5/8" - 27 thread sizes. To use the 5/8" thread, please remove the adapter insert on the shock-mount housing. The RJ-45 jack at the bottom of the em32 array is used to connect a CAT-5 or CAT-6 cable to the EMIB via the RJ45 connector labeled "Mic 1". The total length of the CAT-5 cable should be less than 140 meters. (NOTE: never plug the em32 array or the EMIB into a standard Ethernet network, as doing so could destroy the em32, EMIB, or devices plugged into the network). The EMIB contains the power supply that supplies power to the em32 over the CAT-5 cable.

Next, plug the supplied FireWire cable into either FireWire port on the back of the EMIB. The other end of the FireWire cable should be plugged into the MacBook Pro FireWire 800 port. Power for the EMIB is through the labeled external power supply. The EMIB supply is capable of operating with input voltages from 100-240 V AC and frequencies of 50 or 60 Hz. If everything is fine, the "Power", "CPU", "FireWire", and "Mic 1" LED's on the front of the EMIB should be solidly lit. If any of these LED's is not solidly lit, try power cycling the EMIB. If the four LED's do not light, then contact mh acoustics for support. Do not change any of the "Systems Options" toggle switches in the back of the EMIB as these are for testing and other features. All switches should be off except switch 5 which should be on. Figure 1 shows a photograph of the factory default settings for the switch pack on the rear of the EMIB. The "Word Clock" input and output BNC jacks on the rear are functional, but one has to indicate the word clock source in the TCAT driver. For proper operation of the em32 array, mh acoustics suggests that the user use 44.1 kHz sampling. Operation at 48kHz is possible, but the Eigenbeamformer compensation filters are computed at 44.1kHz so there will be a slight deviation in the overall frequency response of the array if operated at 48 kHz.
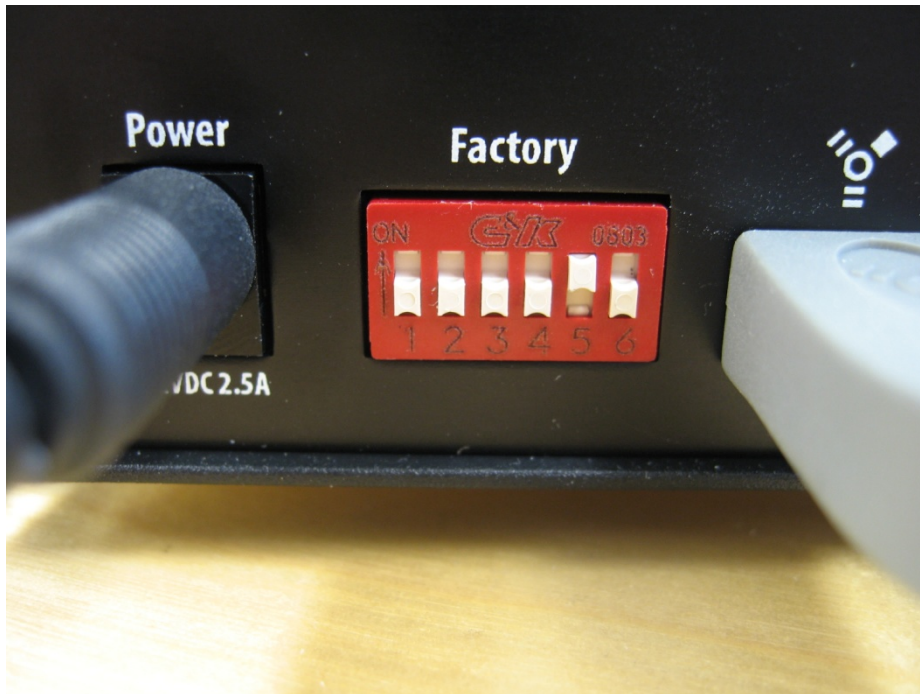
• Figure 1 Photograph of default switch positions on the EMIB

For em32 systems ordered with a MacBook Pro, the machine is a dual-boot machine that can run either Mac OS-X or Microsoft Windows® operating systems.  In the future, mh acoustics will support both operating systems, but the initial software will only operate under the Windows® XP Professional and Windows 7 operating systems as supplied with the em32 system.  By default mh has set the MacBook Pro to boot to Windows® 7.

For audio monitoring and playback, the EMIB has a stereo headphone output with level control. If the user would like more than stereo two-channel playback, the rear optical ADAT output port allows up to 8 simultaneous analog outputs (or the EMIB 8-LO Line-Out analog option card if it is installed inside the EMIB). Note that in the ASIO device driver that ADAT channel 1 corresponds to "Eigenmike-EMIB ADAT 1" which is the 9-th output channel in the ASIO output stream. Similarly, the EMIB8-LO first channel is the 17-th ASIO output channel.  The user has control over how these physical ASIO channels are mapped to mh acoustics EigenStudio™ logical output channels.

Consecutive individual channels on the ADAT interface have correspondingly increasing output channel numbers. For the ADAT digital output, users will need an external D/A that has an ADAT input (these are common devices and relatively inexpensive). mh acoustics offers the EMIB 8-LO daughter card that mounts inside the EMIB that offers 8 balanced analog outputs using a standard TASCAM multichannel audio cable. More outputs can be formed in software and recorded for playback on other multichannel playback hardware. mh acoustics has successfully used the our EigenStudio® GUI software (discussed in more detail later) to record all 32 input channels as well as the all defined output channels, while simultaneously playing back multichannel data from the em32 array.

The orientation of the em32 mounted in on the shaft is such that the negative Z-axis is in the direction of the CAT-6 cable at the bottom of the em32 array shaft. The positive Z-axis is in the direction of the LED on the top of the em32. The negative X-axis is towards the shock-mount holder that defines the equator seam as the X-Z plane. The side of the em32 shaft that has the mh and Eigenmike array logo designates the positive X-axis. The positive X-axis is towards microphone 1 and the Y-Z plane is defined as the plane that contains microphones 1,3,5,9,17,19,21,25. Figures 2 and 3 actually show the microphone channel numbering of the em32 for two views of the array: looking from the positive and negative X directions. Note, that the actual microphones that form the Y-Z plane are microphones 13,14,15,16,32,31,30, and 29.

It should be noted that the internal electronics within the em32 sphere dissipate a few watts of electrical power. Since this power is dissipated as heat energy, the external surface of the em32 becomes warm after reaching steady state temperature. It is normal for the surface of the em32 to reach temperatures around 40-45$^o$ C. Internal chip temperatures can be higher, but all internal chip temperatures are below their maximum rated operating specification. Do not operate the em32 at ambient temperatures where the surface exceeds 50$^o$ C. Covering the em32 with foam or other materials to act as a windscreen should be avoided so as not to allow the em32 surface to exceed the 50$^o$ C maximum temperature. mh acoustics is developing an effective windscreen that will allow air cooling of the em32 surface. mh acoustics is now shipping a cost effective windscreen that allows convective airflow to cool that em32 array while blocking most wind-induced noise from the recording. Please contact mh acoustics for more information or check the News page on the website for details.

Although each em32 is anodized giving it a tough scratch resistant surface, care should be used when handling the array. The microphone holders are mounted from the inside of the em32 so replacing any of the microphone capsules requires the array to be returned to mh acoustics for repair.  Please do not open the em32 or attempt to remove any microphones since doing so without the appropriate tools and assembly/disassembly knowledge could lead to damage of the em32.
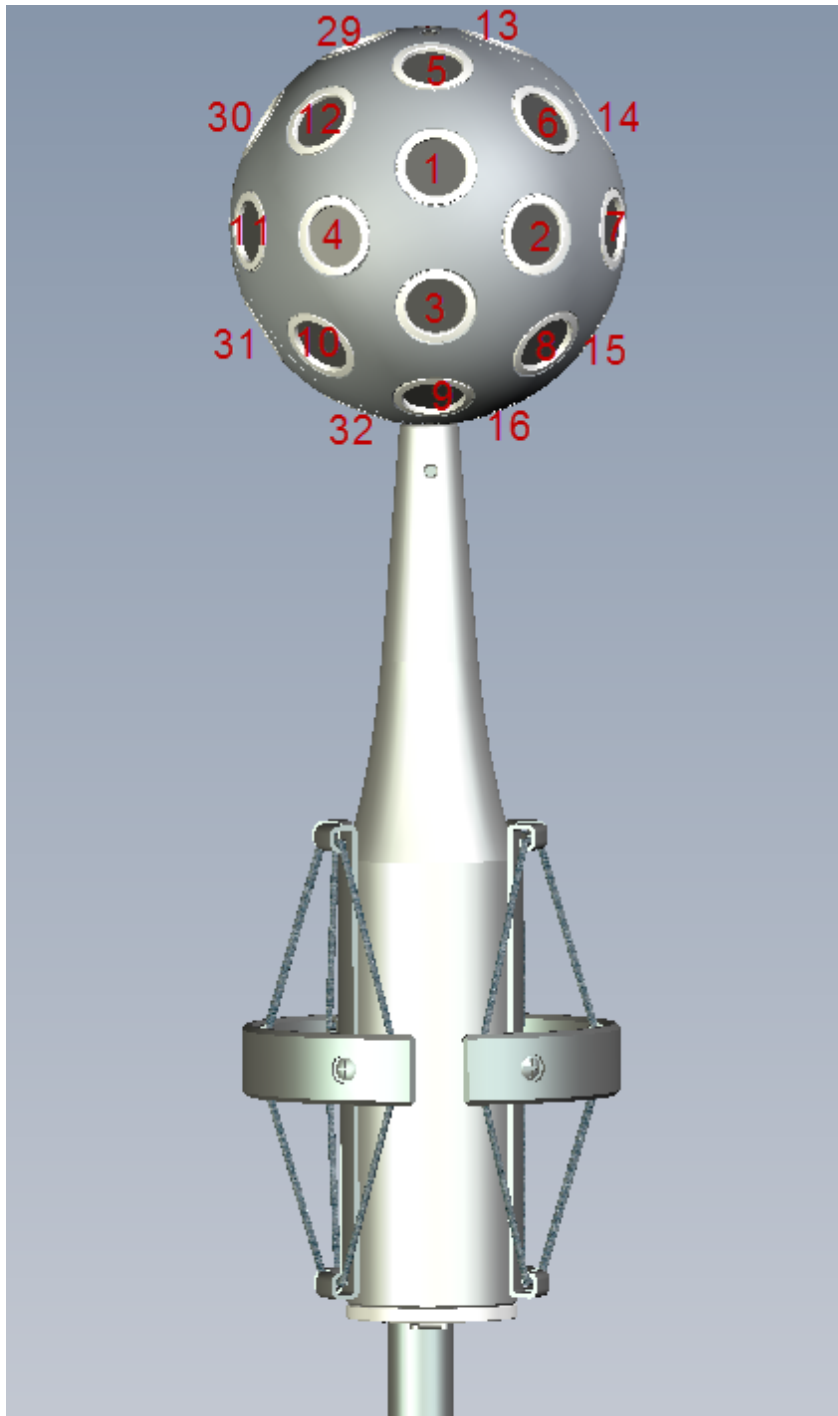
- Figure 2, em32 microphone channel assignment looking from the positive X-axis. The seam between the two halves of the spherical baffle of the array roughly defines the Y-Z plane. (Note that the engraved Eigenmike® and mh acoustics logos appear above and below microphones 1 and 3 respectively)
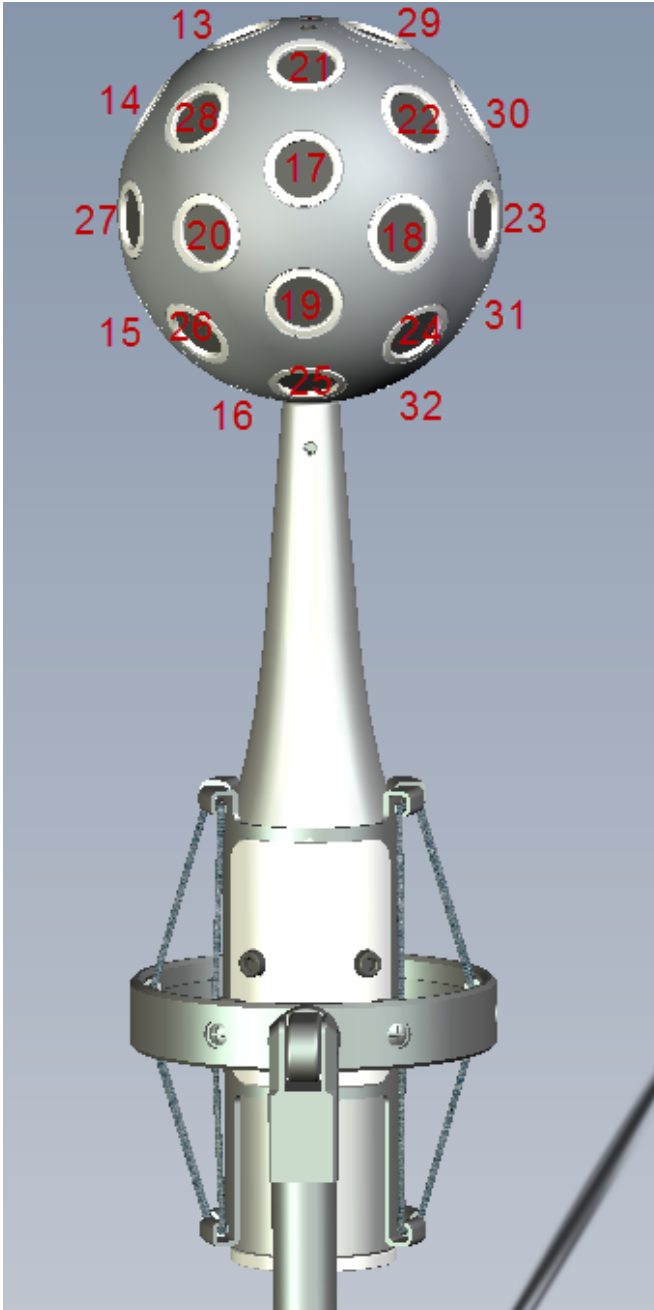
• Figure 3 em32 microphone array microphone channel assignment looking from close to the negative X-axis direction (from the microphone shock mount microphone stand mount).

The EMIB contains a Firewire chipset from TC Electronics. Therefore the Eigenmike® system uses the TCAT DICE ASIO audio driver is used by Windows to connect our software to the em32 hardware.  Early Windows XP systems showed an issue with the 802.11 wireless interface on MacBook Pro systems with BootCamp 1.0 and 2.0. However, with Windows 7 and Bootcamp 3.2-5.0 as well as MacBook Pros built after 2009 we have not seen this problem. However, for reliability of recording real-time audio, <span style="color:red">mh recommends that the user disable the wireless Ethernet when using the em32 system</span>. Wireless 802.11 is disabled by default on delivery of the system. A tool to test real-time performance of the Windows System is included in the TCAT driver. This tool measures the DPC (Deferred Procedure Call) latency. In order to run this tool in Windows 7 one must initiate the driver as a System Administrator by right clicking on the TCAT desktop icon and selecting "Run as Administrator". The DPC latency tool can be used to track down issues related to audio glitches or dropouts in the real-time audio stream due to processes or programs that are demanding high priority from the OS. As shipped, the DPC latency is low and in the "green" region on the DPC tool. The TCAT driver allows one to adjust the audio buffering of the driver by choosing higher levels of "Safe Mode" levels. Increasing the "Safe Mode" level adds more latency into the audio chain as it increases the internal buffer sizes within the ASIO driver.

The supplied ASIO audio drivers are labeled as "TCAT Dice EVM Platform" since we have chosen not to use the mh acoustics signed driver (this would only have been a name change to the driver). Future em32 systems might present the string "mh acoustics Eigenmike® microphone array" if we decide to use a signed driver.  The ASIO driver allows the em32 system to appear as a 32-channel in/out sound card. Thus, one can use any standard multichannel audio processing software to record and playback all or any of the raw Eigenmike® array microphone channels. Each em32 system is supplied with our EigenStudio® application for controlling the Eigenmike. The EigenStudio® application offers a simple GUI interface for control, record and playback.

mh's EigenStudio® application (mh icon on Desktop) defaults to forming two simultaneous output channels forming a stereo output (to start the user in a simple state). As discussed later, users can store defined setup states and recall those later.  For Podium, the setup has an Eigenmike® project defined that allows the user to have complete control over the 8 output channels. One can also direct any of the 8 output channels to the headphones by reassigning the output channel in Podium. A manual for Podium is included on the disk in the C:\EM32 Eigenmike directory and is also placed as an icon on the desktop.

Control of the em32 system is handled via a virtual midi connection to and from the em32. Data communication can be observed on the EMIB front panel "Midi" LED. One can also confirm command communication to the array by observing the multicolor LED on the top of the array (positive Z-axis). During midi communication (to set and get programmable settings of the array hardware), the LED goes solid green. If communication does not occur within a 5 second window, the LED appears orange in color.

In the first revision of the em32 EMIB hardware (Mark 1), for midi commands to properly control the em32 internal hardware such as PGA (programmable gain amplifiers) on the microphone inputs and access to other pertinent data from the em32 EEPROM and other registers, the sampling rate must be set to 44.1 kHz.  EMIB Mark II units were introduced in 2009 and used an updated chipset that removed this limitation and one can select either 44.1 or 48 kHz for the internal sampling rate. Figure 4 shows a screen snap-shot of the correct state setting of the TCAT device driver that as shipped in the first revision of the em32 product. The driver shows the current sampling rate. Initial em32 systems with Mark I EMIB's only allow a sampling rate of 44.1 kHz. However, systems EMIB Mark II interfaces allow one to select internal sampling rates of 44.1 and 48 kHz.. One can check

the suitability of real-time audio on the system by using the DPC latency check in the DPC tab. If one wants to use the external word clock as the master clock, (through the rear BNC or the ADAT interface), this can be accomplished by selecting the Sync Source in the TCAT driver settings window.

mh ships em32 systems with a Buffer Size of 1024 and "Safe Mode Level 1" in the Operation mode tab. We have not seen many operational differences between "safe" and "normal" modes in the "Operation Mode" tab within the ASIO device panel. However, we have seen some more robustness against occasional audio glitches if one uses the "safe" mode. If you experience any audio drop-outs, it is recommended to increase the Buffer Size and to increase the Operation mode to a higher "Safe Mode Level". It should be noted that increasing the buffer size and higher level "Safe Mode" levels increase the audio delay through the PC system. This would only be necessary if you have loaded other applications that run in the background and are demanding of real-time system resources.

In the bottom half of the Driver window (see Figure 4) one can see the general setting and the devices that are available to the driver (should see Eigenmike EMIB). There is also "Firmware Loader" button that allows users to update the EMIB firmware as new updates become available. The em32 hardware firmware is presently only upgradeable only by sending the em32 back to mh acoustics,

Also please note that the frequency response of the em32 has been bandlimited to an upper frequency of 20 kHz. This bandwidth is divided into two operational modes of the Eigenbeam beamformer processing. The lower region operates up to 9 kHz and gives full control of the Eigenbeamforming. Above, 9 kHz, the beamformer changes modes and uses a subset of elements to extend the beamformer to 20 kHz. The 9 kHz limit is due to the discrete spatial sampling of the sound field with 32 microphones distributed relatively uniformly on the 8.4cm diameter spherical baffle. The electret microphones used in the em32 have a frequency response upper limit that exceeds 20 kHz.
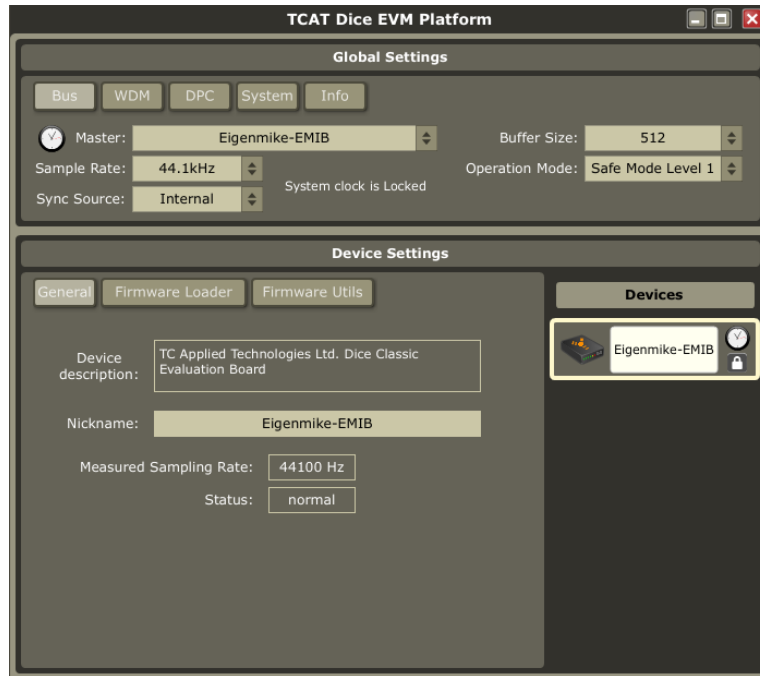
• Figure 4 Screen snapshot of ASIO driver setup for Eigenmike® EMIB

## em32 Eigenmike® recording, playback, and Control

Each em32 system is supplied with mh acoustics EigenStudio™ GUI application that is described in the sections below.

### EigenStudio® GUI

mh acoustics' EigenStudio® application is the main software control application written by mh acoustics for our em32 Eigenmike® microphone array. This front-end GUI application allows for control of the main beamforming capabilities of the em32 array. It allows for Eigenbeamformer rendering for up to 18 real-time output channels from signals from the em32 array or from previously recorded data. From recorded data, EigenStudio™ can output any number of desired output channels to file. The EigenStudio™ application can be run for processing and playback without an em32 and EMIB plugged into the PC. When an EMIB is not detected, the EigenStudio™ application allows the user to select any ASIO driver or device that are defined on the machine for audio output. To use the PC's sound card for output, select ASIO4ALL or equivalent ASIO driver for sound output rendering (systems shipped with a MacBook Pro include the ASIO4ALL driver).

The EigenStudio® application is stored in the C:\EM32 Eigenmike\EigenStudio directory. A desktop shortcut is placed on the main screen to allow for quick launch of the application. Figure 6 shows the default main GUI window. The GUI is split logically into 7 areas from the top left to bottom right: the main timeline bar, Input Levels and Volumes, high-pass filter settings, signal monitor selection, output levels and level control, input source (live from the em32 or from a saved file), beampattern(s), Beam Tracks, Recording channels (microphone inputs, processed output, or both), and the Stop, Pause, Play, and Record transport buttons. Each of these sections will now be described.

## 1.1 "Settings" Pull-down

The "Settings" pull-down menu item at the upper left of the main screen above the File Navigation panel allows the user to select how to route the output channels and monitor channels.  Figure 5 shows the pull-down panel to select the output signal routing. EigenStudio™ can form any number of output channels (note that Figure 5 shows the window that would appear with the EMIB-8LO option card installed in the EMIB). The pull-down allows one to "activate" the physical output channels on the EMIB as well as assign the order that the channels are assigned to the beam outputs from Eigenbeam processing by the EigenStudio™ application. The radio button under the "first group" label allows the user to assign the order of the output signals from EigenStudio when the EMIB is present.  By default the ADAT output is the default output from the EMIB. Therefore Beam 1 would be output on ADAT channel 1 and so on. If the user selects the 8-LO as the "first group" then Beam 1 would be output to channel 1 of the EMIB-8LO analog output option card and ADAT would be assigned to the second set of 8 output channels. By allowing one to also use the headphone output as a physical output signal up to 18 real-time output channels are available when an EMIB is plugged into the PC. One can assign the two-channel headphone as the "first group" but mh would advise against that since one could mistakenly output the "monitor" channel selection to headphones and therefore those channels would have the audio that was selected in the monitoring selection which has precedence over the Beam output selection when routing output audio.

Figure 6 shows the output channel selection when the EMIB is not plugged into the PC when starting EigenStudio. For this case, the user can select any number or hardware output channels depending on the number of output channels are present in any other multichannel ASIO hardware that is present on the PC. When EigenStudio™ is used in playback mode from a recorded file one might have ASIO audio hardware that can playback with many more channels than the 8 that are available on the Eigenmike® systems EMIB interface box. It is also possible that the user might want to offset the ASIO output channel on playback (this can be a very useful feature) and this pull-down panel allows the user to change the desired logical first output channel. If one is using the EMIB, then the user should select 8 channels which are available at the ADAT optical digital output on the back of the EMIB or the 8-LO if that output option is present in the EMIB.



• Figure 5 Settings pull-down for setting up the output channels
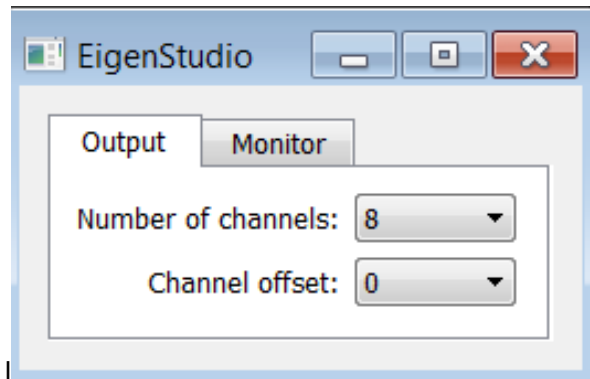
The other section of the "Settings" pull-down allows the user to select how the output monitor channels are defined.  Figure 7 shows the pull-down panel when the "Monitor" tap is selected and some non-EMIB hardware is connected.
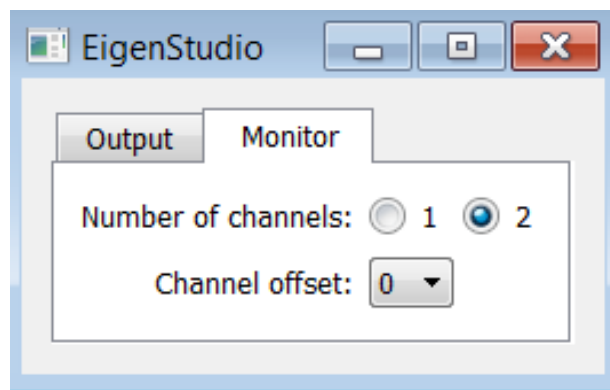


• Figure 7 Settings pull-down for setting the monitor channel(s)

For the monitor selection the user again has the ability to listen to any mono signal that can be any raw microphone input or any beam output signal and to assign that channel to any of the ASIO hardware output channels.  The selection of the "Number of channels" allows the user to play the monitor signal diotically (same audio signal on two channels). This option is useful when monitoring over headphones and it is desired to have audio on both the left and right headphone channels. Like the settings for the output channels, for convenience, one can define the monitor channel to be offset in the hardware ASIO channel output. Note that the monitor channel can overwrite the audio (beam) output channel. For example with the above setting (Figs. 6 and 7) beam output 1 and 2 would be overwritten by the monitor output (assuming that monitoring is active – see below).

## 1.2  File Navigation

The "File Navigation" panel (Figure 8) provides feedback about the current position in the audio file during playback. The pane graphically shows the current position within the file, allows the user to select audio sections, to move to a certain position and more. The following items describe the separate parts of the panel in detail:

- Timer: The numeric timer at the upper left of the panel displays the current position in the audio file in the format hours: minutes: seconds.

- Upper Timeline: The top timeline always corresponds to the full audio file. It displays a gray bar that reflects the selection that is covered by the lower timeline. The top timeline is only for display purposes, no active selections can be made within the bar.

- Bottom Timeline: The lower timeline enables the user to select certain parts of the audio file for listening or processing. A specific time selection is made by a click & drag operation and is marked by a light blue bar. For example to select the audio from 1:40 minutes to 1:55 minutes the user will have to left-click inside the timeline at position 1:40, then drag the mouse to position 1:55 and release the mouse button. The "Audio Transport" buttons at the bottom of the UI will now act only on the selected part of the audio, e.g. "play" will start the audio at 1:40 minutes.

Another feature is the selection of the current position. The dark blue bar indicates the current position in the file (if no dark blue bar is visible the audio has been stopped and the current position at the left end of the selected – light blue – bar). If the user left-clicks inside the lower timeline (and releases the mouse button) the current audio position will move to this location.

- Zoom In: The zoom-in "<>" button is located to the right of the lower timeline. Clicking this button will expand the current audio selection over the full length of the lower timeline. Note that the gray bar in the top timeline shrinks to reflect the update in the lower timeline. The zoom-in feature is useful in large files if the user wants to narrow in on a short section of audio.

- Zoom Out: The zoom-out button "><" is located to the right of the upper timeline. The zoom-out button will reset the lower timeline to the full file length.
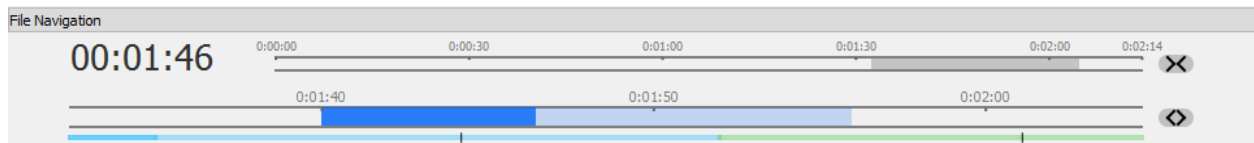


- Figure 8: File Navigation panel

The timeline bar and counter at the top of the application is used only during playback operation. During playback, the time counter indicates the time from the beginning of the playback file as well as a visual indication on the bar as to how far the user is into the total length of the playback file. Note that during recording, the amount of time recorded so far is NOT displayed in the File Navigation" panel but is shown next to the input or output file name that is selected (lower right hand side next to the file name).

One useful feature is that the user can "rubber-band" a time selection in the timeline bar by holding down the left mouse button and dragging it along the timeline bar. Playback will loop over this defined time range until either the user stops playback or selects another time region of interest. After selecting a time region (or just letting the file play out in its entirety), one can select to record the output and save the processed audio to a file while playing the recorded audio. Figure 9 shows an example of a prerecorded file being streamed, processed, and recorded to the selected output file (one could also record the input again which could be useful if one would like to retain only a segment of the file that was selected in the time bar earlier.)
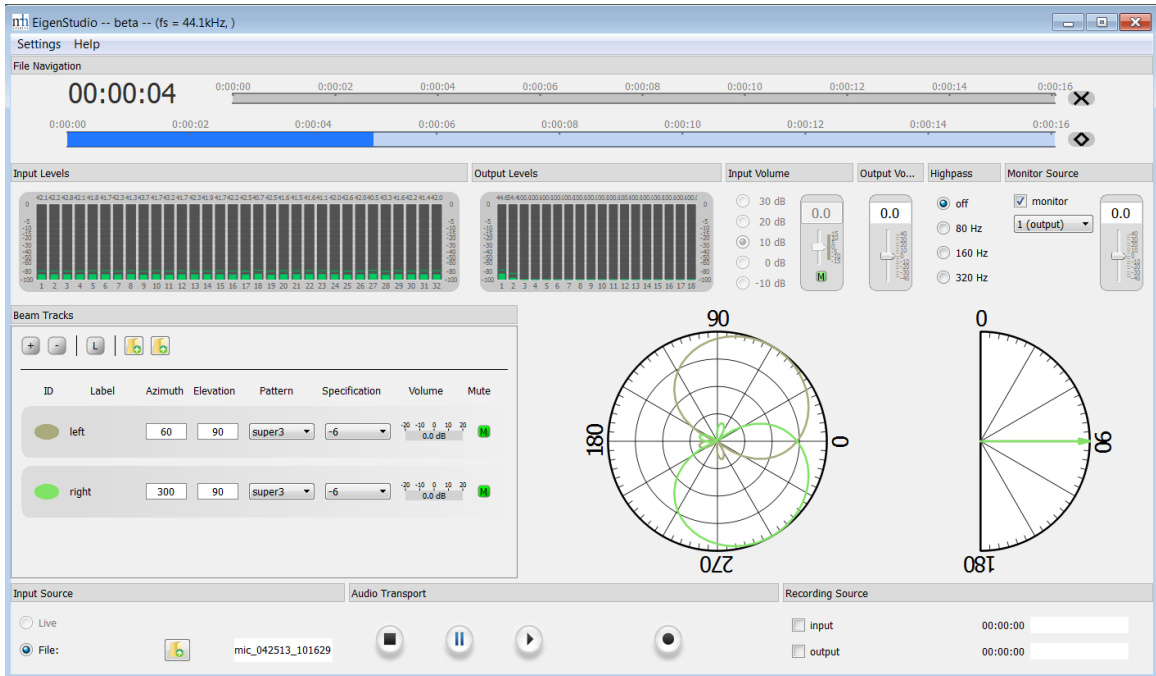
- Figure 9 Main screen for EigenStudio® application

## Input and Output Levels, Input Volume and High-pass filters

In Figure 9, the Input and Output Levels (green VU meters) are shown below the timeline navigation bars. These levels indicate the overall audio input and output levels either live from the microphone or from the recorded file. The VU Levels allow one to verify that all inputs are functional and within the operational level of the A/D and D/A converters. Next to the input VU meters on the right, is the Input Volume that sets the microphone programmable gain amplifier (PGA) that is applied to the microphones before the A/D converter digitizes them. The discrete PGA control range is 40 dB which should be adequate for most cases. The em32 microphone elements are rated to 130 dB in SPL with a sensitivity of 30 mV/Pa. The input gain has to be reduced to -10 dB to prevent clipping of the A/D converters when the input level reaches the maximum SPL rating of the microphone elements. When the input audio signals are low in sound level, applying PGA gain to the microphone signals before the A/D converter is appropriate to fully exercise the A/D's input signal range. The volume slider next to the PGA radio button allows for an additional +- 14 dB of digital gain. Please note however that this digital post gain and does not change the input microphone PGA inside the A/D converter. To the right of the digital gain slider is the ability to insert a high-pass filter at 80, 180, and 360 Hz to remove unwanted low frequency signal components in the recording or playback.

## Monitor Source

On the far right panel below the timeline bar is the Monitor Source selection. This allows one to audition any of the input or output signals and verify that everything is behaving correctly. The pull-down box allows the user to listen to any input or output signal through the headphones or other hardware output channels. Note that the monitor output can overwrite an audio (beam) output. Refer to section 1.1 for selecting the monitor output channel.

## Beam Tracks and Beampattern

Below the Input and Output VU Levels is the Beam Tracks section. The Beam Tracks section is the main user control and setting section for the Eigenmike® Eigenbeam processing. In this section, the total number of output beams can be selected (by adding or deleting beams), the beampatterns can be changed and steered and the output gain of each beam set. One can also "Link" all of the beampatterns by selecting the "L" radio button along the top of the Beam Tracks section. Linking the beampatterns allows the user to rotate one beampattern and have all other beampatterns rotate the same amount is both azimuth and elevation. ). Next to the "L" button are two file icons, mouse clicking on the left icon allows saving the current Beam Track setup to a file. Click on the right icon to load a saved Beam Track setup.

Currently there are 10 selectable beampatterns that can be chosen: omnidirectional, cardioid, hypercardioid, and supercardioid. The cardioid patterns can be with first, second, or third-order patterns. (More technical details on the selectable beampatterns are given in Appendix A). Since it em32 Eigenmike® array is capable of any general beampattern up to and including third-order, future updates to EigenStudio will allow the user to design and select whatever beampattern they like that can be realized with spherical harmonics up to and including third-order).

At the top of the Beam Tracks section pressing the "+" adds a new output channel. Similarly pressing the "-" deletes an output channel. Pressing the "L" link box allows one to keep the relative steering directions the same when steering one of the output beams steering direction. (Note that one can also place the mouse cursor on top of one of the beampatterns in the Beampattern panel and while pressing the left mouse button, can drag the beampattern steering direction to any desired steering direction. One can also just click the mouse at the desired steering angle anywhere along the outer radius of the polar plot (just inside the angle numbers).

Clicking on a Beam Track label highlights that beam track in the Beam Track window as well as the Beampattern window (by increasing the line width in the graphic). Right clicking on the Beam Track label allows the user to change the Beam Track label. Double-clicking inside the oval color icon in the Beam Track allows the user to change the Beam Track color. The "Specification" parameter (which is actually the White-Noise-Gain constraint limit) can be set in the pull-down menu. More negative values of the "Specification" allow the beamformer to increase directionality at lower frequencies with the tradeoff that there can be SNR loss through the beamformer (relative to that of a single microphone element of the array).

To the right of the Beam Tracks control area is a section that shows two graphic facsimiles of the output beampatterns and steering direction of the outputs that have been defined in the Beam Tracks section. As mentioned above, placing the mouse on the highlighted Beam Track while holding the left mouse button allows one to "drag" the beampattern azimuth and elevation to other angles. Again, clicking on the outer radius of each plot also steers the beam to that angle.

**Input and Recording Source**

Below the Beam Tracks section on the left hand side of the EigenStudio® application screen is the selection of either live signals from the em32 array or from a recorded file. Clicking on the file open icon brings up a dialog box to select the recorded file to load.

 The Recording Source on the far right allows one to record the input, the output, or both. The elapsed amount of time for the recording is shown next to the recording source radio button as well as in file name dialog box to select the names for the recorded files.

**Transport Control**

Between the Input Source and Record Source sections is the Audio Transport control region. Icons in this region are standard in many audio tools. From left to right the controls are: Stop, Pause, Play/Start, and Record. To start audio processing from startup, press the Start icon. To Record, first Arm the recording by pressing the Record icon. Record is armed when the Record icon turns red in color. Once armed, begin recording by pressing the Play/Start button. Pressing the Pause button while recording will stop the recording. Pressing the Pause button again (toggling the Pause state) will start the recording again. Note that pressing Pause and then resuming saves the data to a new filename that also contains the recording date and a unique identifier generated by the PC system clock.

## Microphone PGA control GUI

The C:\EM32 Eigenmike directory contains a GUI program em32gain.exe that allows for general control of the em32 microphone PGA setting program outside of the EigenStudio™ application. This program brings up a simple GUI control for setting the em32 microphone preamplifier gains in 1 dB steps. Technically the discrete set of PGA gain steps in the EigenStudio application should be sufficient, but some users may want more precise control or are using another DAW tool with our VST plugin which does not support PGA control. Figure 10 shows the GUI look that pops up onto the Windows display when the program is run. The program first checks to make sure it can establish a connection to the EMIB and the em32 array. If it can make a connection, there will be a green check mark next to each device at the top of the GUI. The slider allows the user to change the input gain of the microphone before A/D conversion over a range or 40 dB. The green "M" at the bottom of the slider is a mute button. Pressing this button will mute the inputs to the A/D converters inside the em32 array.

The GUI program has a slider that allows the user to set the gain over a range of -10dB to +30dB as well as mute the microphone signals. Note that em32Gain.exe first reads all of the individual microphone element calibration values stored within the em32 and then applies the appropriate gain to each channel by adjusting the PGA gain settings to get each channel within 0.5 dB of each other.

As mentioned in the introduction, each em32 microphone capsule is calibrated for magnitude and phase response relative to a standard reference. mh acoustics measures the frequency response of each microphone for every em32 and stores this data for future reference. The electret microphones used in the em32 have an excellent magnitude and phase match from unit to unit. This is due to the fact that the manufacturer we have chosen builds these microphones for professional applications and rigorously tests these microphone cartridges. With such excellent phase match of the electret elements it was determined that a simple gain match to was all that is required for proper operation of the em32. Therefore, calibration is done simply by using the stored gain deviation that mh acoustics measures during the calibration of the elements and then using the measured sensitivity of each microphone to match microphone sensitivities. Finally, it should be noted that the electret microphones used in the em32 were designed to minimize self-noise in the microphone element and each element has an Equivalent Input Noise (EIN) of 15 dBA. (Note that if one intends to record
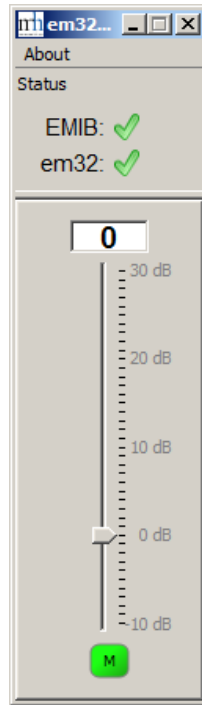
● Figure 10 em32gain.exe GUI

As a result of minimizing the EIN in the microphone element, the individual microphone response is not flat to 20 kHz. mh acoustics compensates for the frequency response of the microphone elements in the Eigenmike® VST plugin.

mh acoustics thanks-you for your purchase of the em32 Eigenmike® microphone array system. Your new Eigenmike® array has been carefully hand assembled, calibrated and tested, by mh acoustics personnel.  We hope that your new em32 Eigenmike® array allows you to do some new and interesting things with regards to recording and analyzing spatial audio. Please feel free to contact us and tell us what interesting things you are doing with your em32. We would really like to get your feedback on how to further improve the software and hardware.

## Appendix A – Performance metrics: em32 beampattern responses, Directivity Index and White-Noise-Gain

The VST release of the em32 VST plugin supports 12 user selectable beampatterns. Appendix A shows the design predictions of the main performance metrics associated with directional microphone systems. For each selectable pattern (which can be steered to any spherical angle) beampattern, the directional polar response is shown at a few select frequencies (300 Hz, 1 kHz, 4 kHz, and 8 kHz). The polar patterns are shown with a 40 dB dynamic range, where each concentric circle represents a 10 dB difference in output level. Note that beampatterns are three-dimensional functions.  Thus the overall 3D beampatterns for the plots shown below can be easily visualized since these patterns are rotationally symmetric around the abscissa (the line defined by the radial line at 0 degrees).

A more comprehensive view of the directional response of the beamformer is shown to the right of the polar pattern as a surface "waterfall" plot. This plot shows the directional response over all

frequencies. The visual dynamic range shown in the waterfall plots is 30 dB whose response level is shown as a different color.

Next, the directivity index (DI) is shown for each of the selectable beampatterns. DI is useful if one wants to predict the amount of directional gain of the array in a diffuse sound field (since DI is defined as the ratio of on-axis received power to the integrated power over all possible angles of incidence).

Finally, a plot of the White-Noise-Gain (WNG) is shown for each of the selectable beampatterns. WNG is an important metric in beamformer performance as it shows the amount of gain over white noise of the beamformer WNG can be used to predict the output SNR of the beamformer from knowledge of the noise of the microphone and electronics used in the beamformer. The em32 microphone and electronic input-referred self-noise for each input channel is 17 dBA. With a WNG of 0 dB, the beamformer output will have the same SNR as a single element of the beamformer. Thus, positive WNG mean that the output SNR (where the noise here is uncorrelated self noise between the channels and not spatial noise) will be lower than a single element. Similarly, an negative WNG implies that the output SNR will be lower than that of a single microphone SNR.
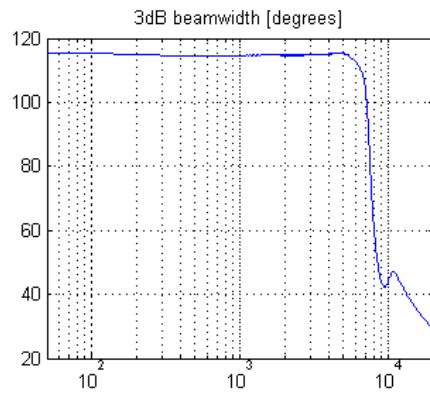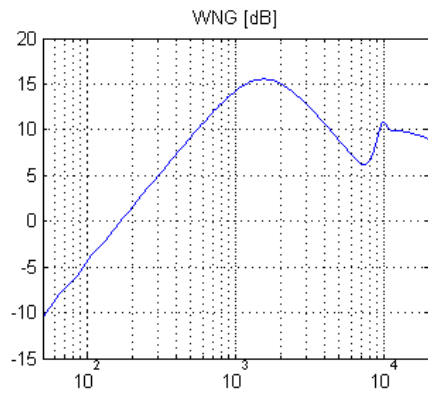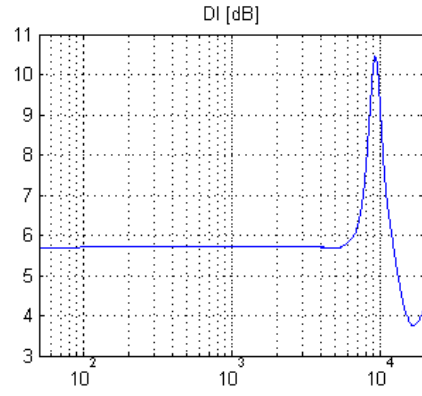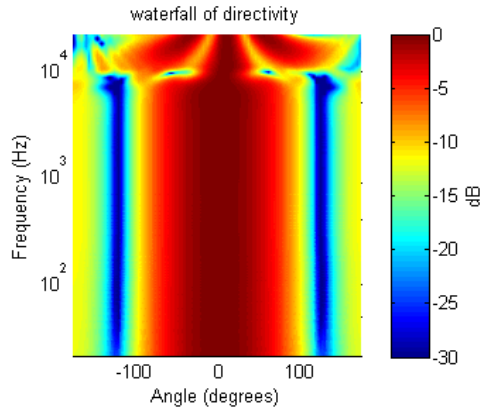
# First-order cardioid

## Second-order Cardioid

waterfall of directivity

DI [dB]

WNG [dB]

3dB beamwidth [degrees]

directivity pattern
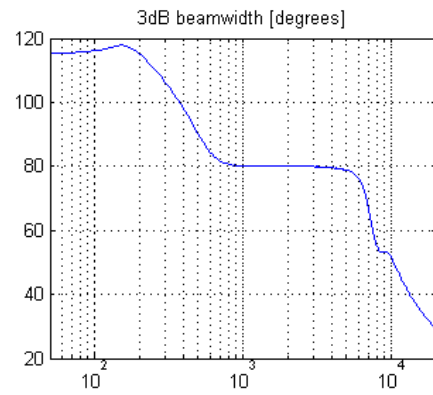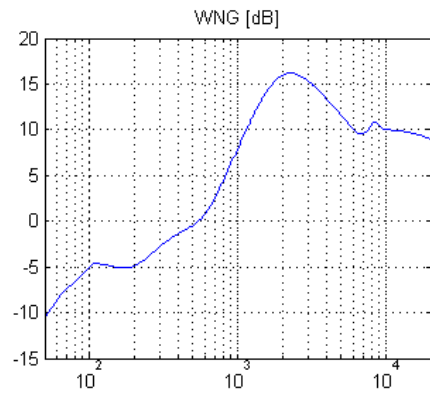
| | |
|---|---|
| —— | 100Hz |
| —— | 500Hz |
| —— | 1000Hz |
| —— | 4000Hz |
| —— | 8000Hz |
| —— | 16000Hz |

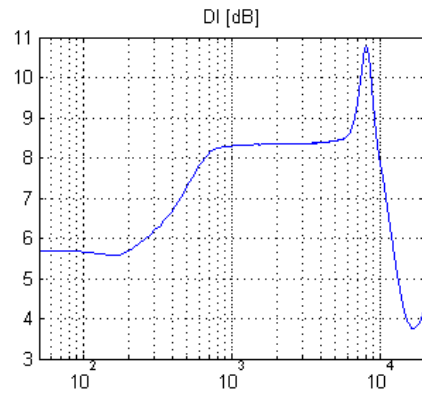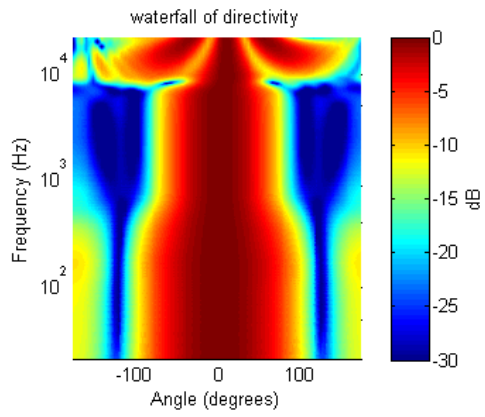# Third-order Cardioid



waterfall of directivity



DI [dB]



WNG [dB]



3dB beamwidth [degrees]



directivity pattern

| | |
|---|---|
| —— | 100Hz |
| —— | 500Hz |
| —— | 1000Hz |
| —— | 4000Hz |
| —— | 8000Hz |
| —— | 16000Hz |
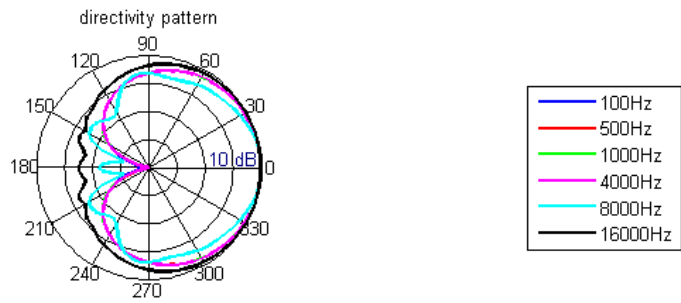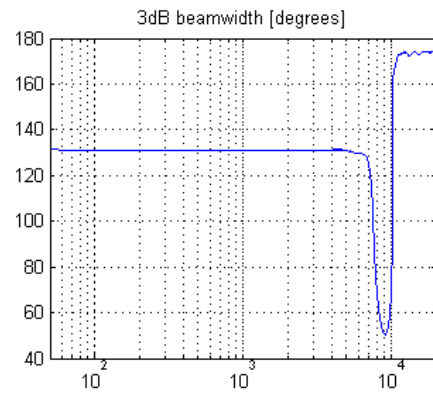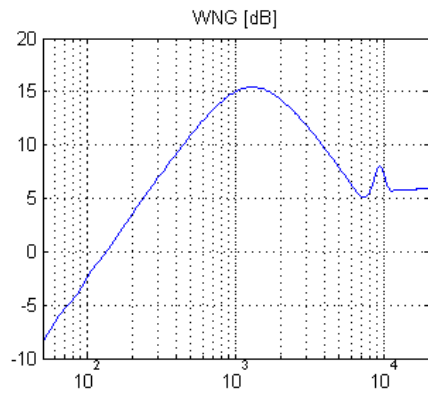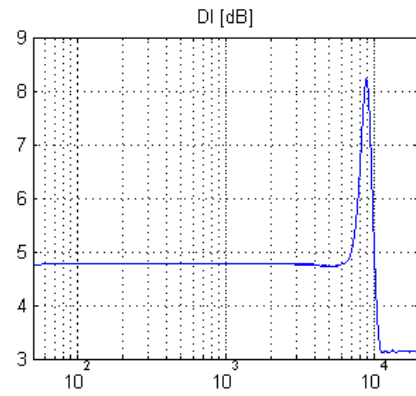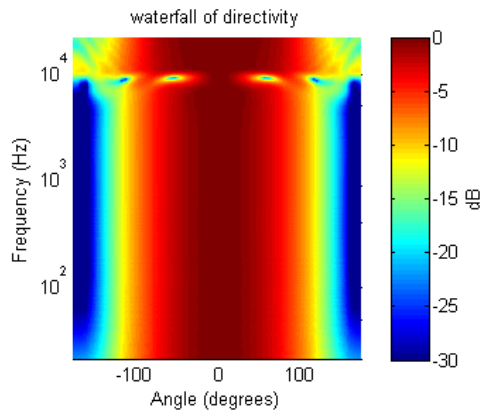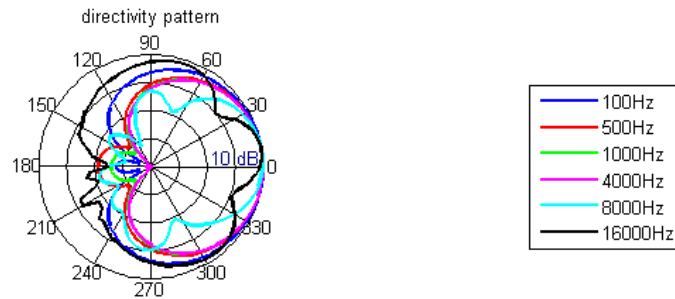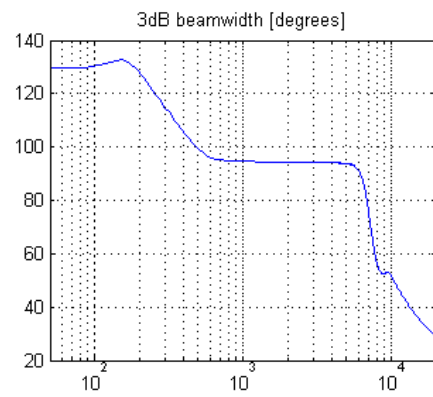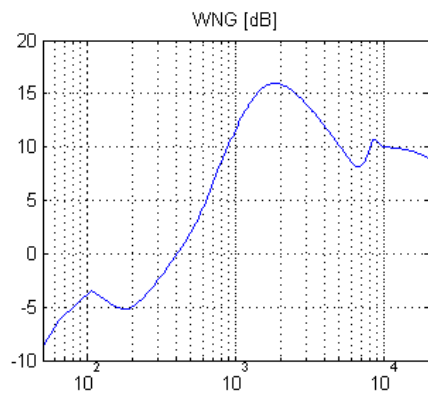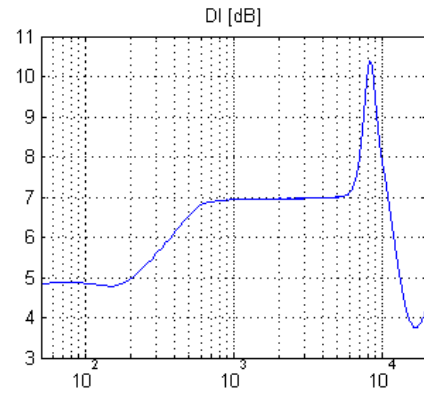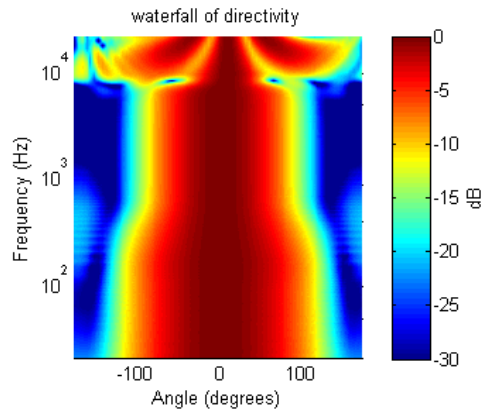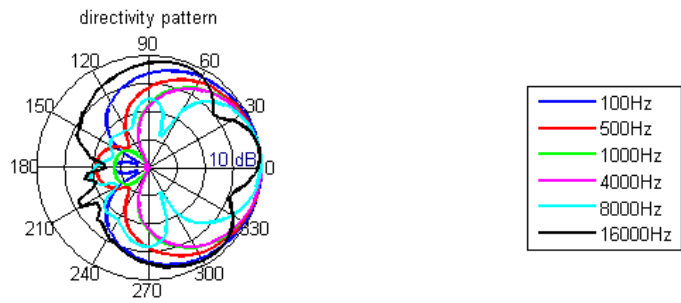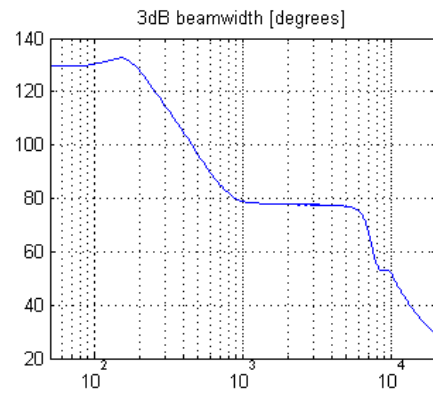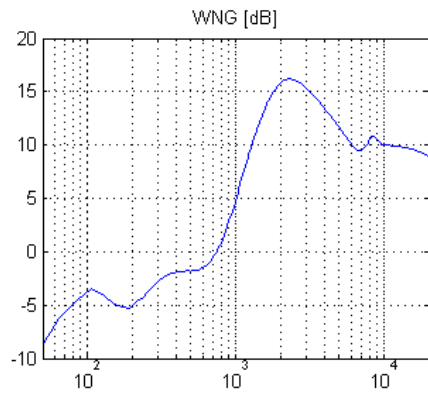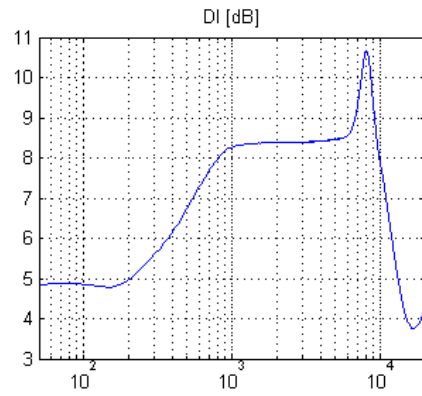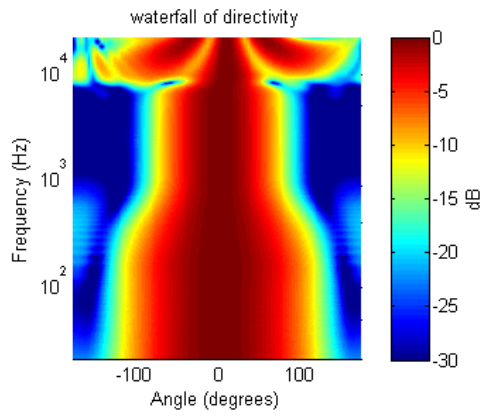
# First-order Hypercardioid

waterfall of directivity

DI [dB]

WNG [dB]

3dB beamwidth [degrees]

directivity pattern

100Hz
500Hz
1000Hz
4000Hz
8000Hz
16000Hz
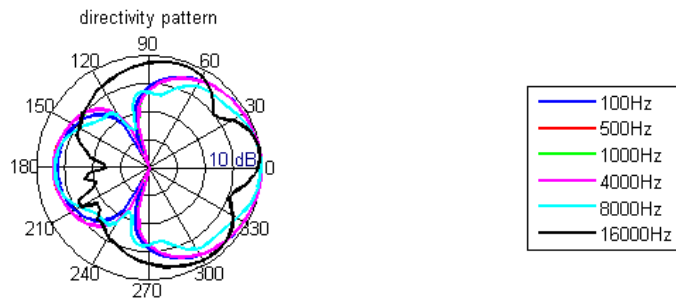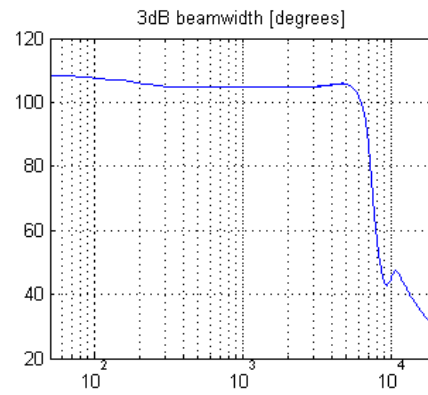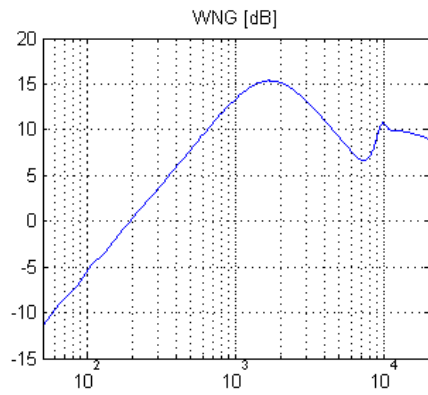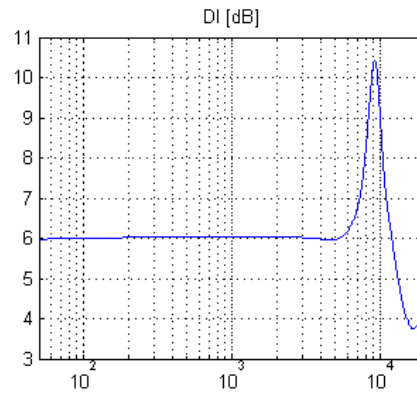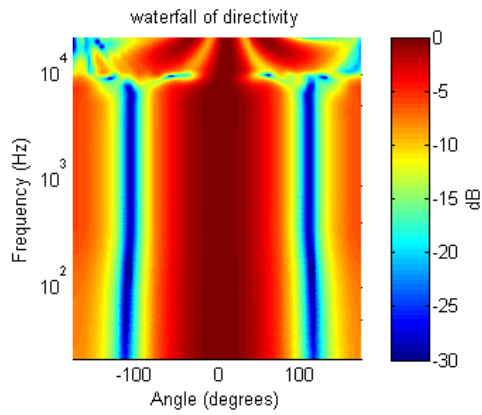
# Second-order hypercardoid

# Third-order Hypercardiod


waterfall of directivity


DI [dB]


WNG [dB]


3dB beamwidth [degrees]


directivity pattern

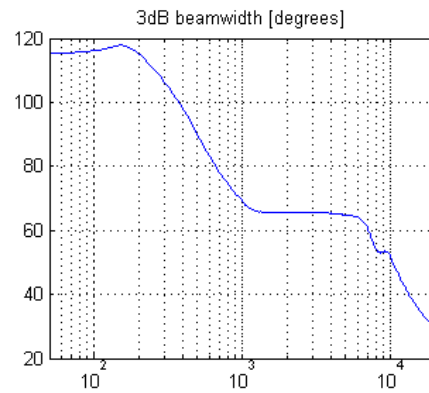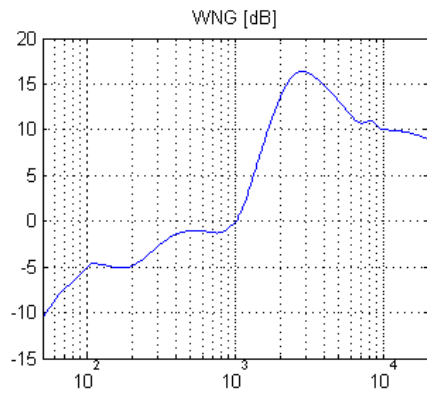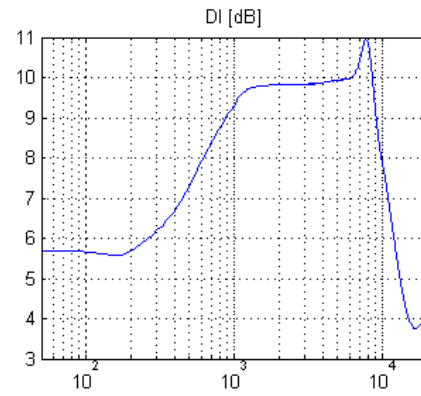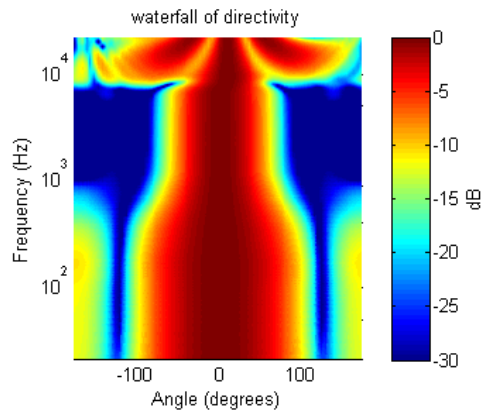| | |
|---|---|
| — | 100Hz |
| — | 500Hz |
| — | 1000Hz |
| — | 4000Hz |
| — | 8000Hz |
| — | 16000Hz |

# First-order Supercardioid

waterfall of directivity

DI [dB]

WNG [dB]

3dB beamwidth [degrees]

directivity pattern

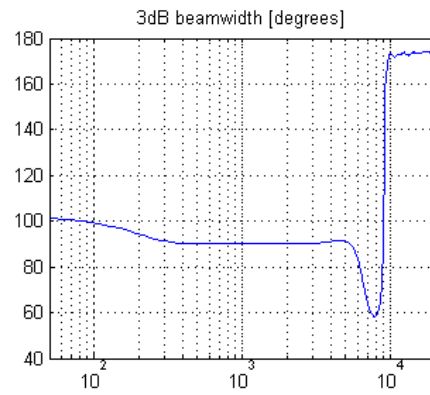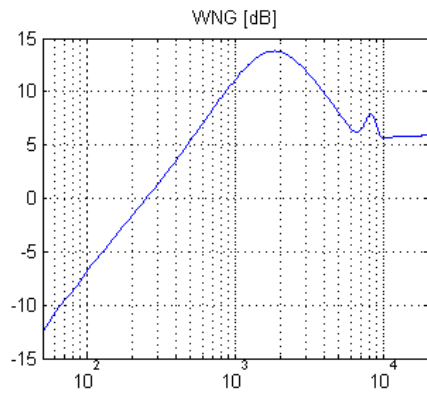| | |
|---|---|
| 100Hz | |
| 500Hz | |
| 1000Hz | |
| 4000Hz | |
| 8000Hz | |
| 16000Hz | |
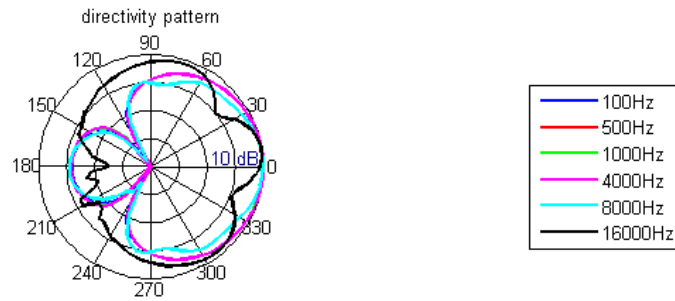
# Second-order Supercardioid

# Third-order supercardioid



waterfall of directivity



DI [dB]



WNG [dB]



3dB beamwidth [degrees]



directivity pattern

| | |
|---|---|
| —— | 100Hz |
| —— | 500Hz |
| —— | 1000Hz |
| —— | 4000Hz |
| —— | 8000Hz |
| —— | 16000Hz |

# First-order dipole



waterfall of directivity

DI [dB]

WNG [dB]

3dB beamwidth [degrees]

directivity pattern

| | |
|---|---|
| | 100Hz |
| | 500Hz |
| | 1000Hz |
| | 4000Hz |
| | 8000Hz |
| | 16000Hz |
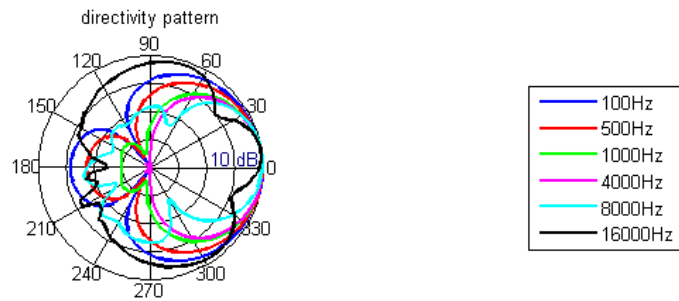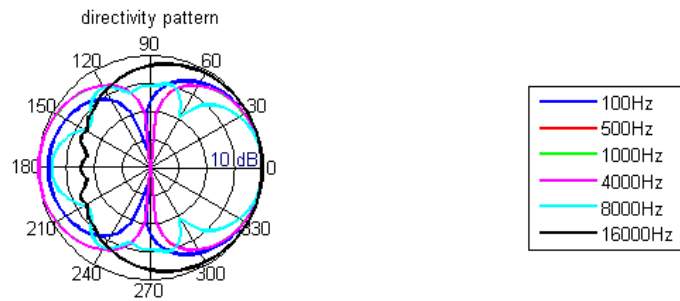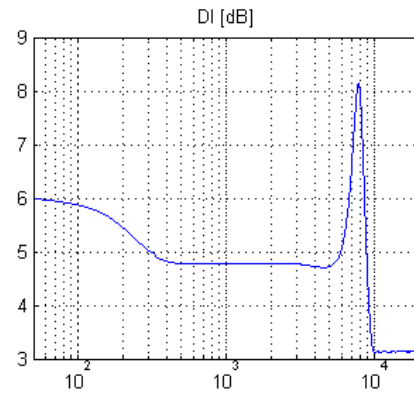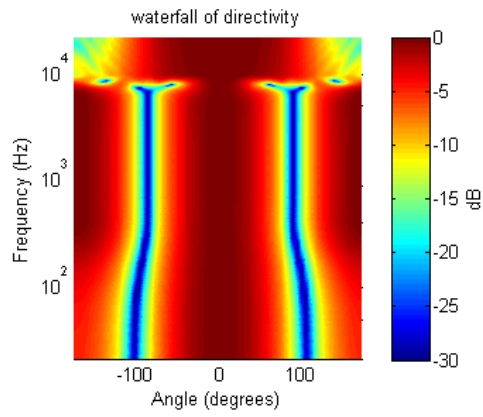
## Torus (second-order)[1]
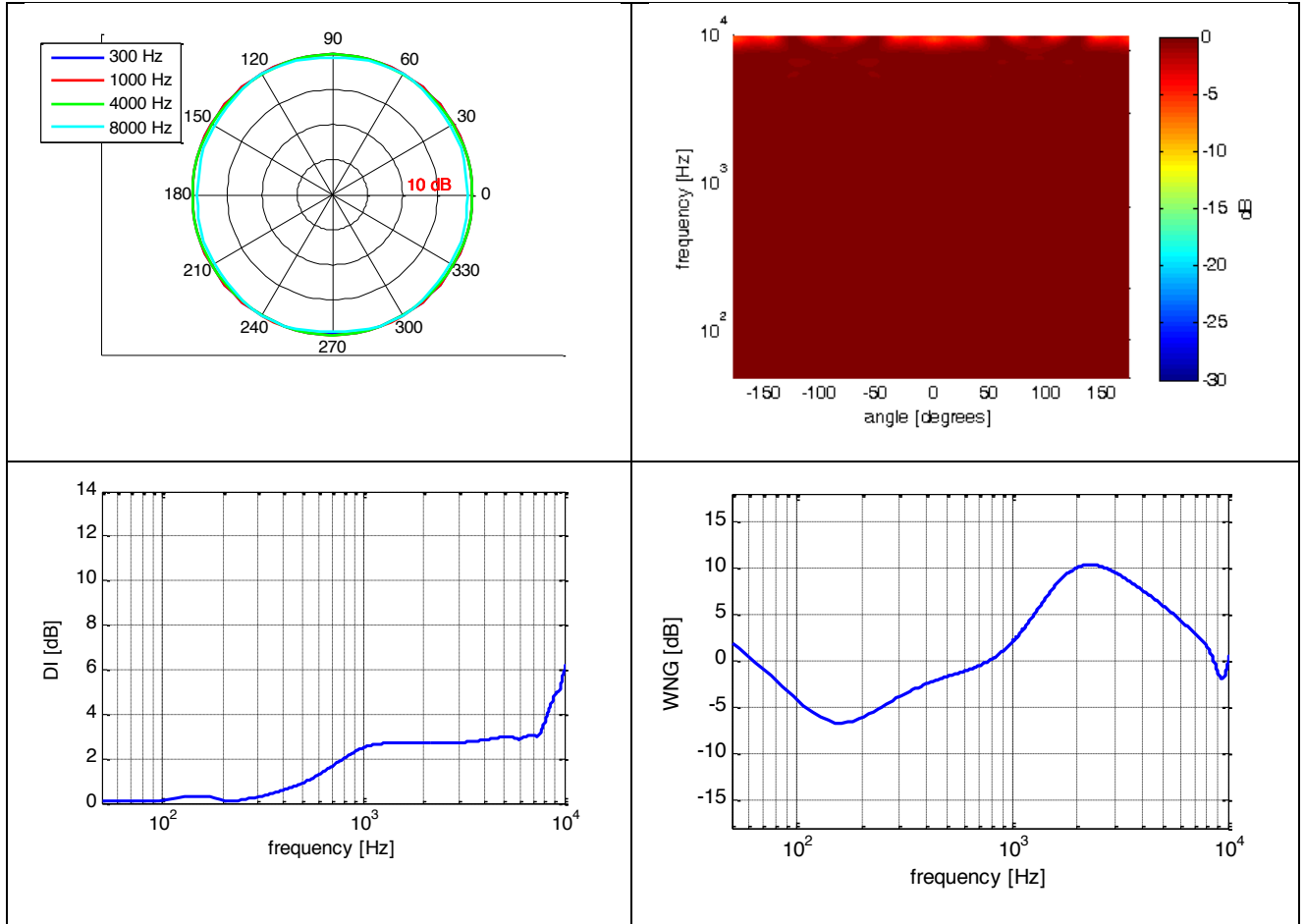
[1] Steering angle: $\vartheta = 0°$, $\varphi = 0°$; look-direction: $\vartheta = 90°$, $\varphi = 0°$.

## Appendix – em32 microphone element positions

The microphones are numbered in a pattern that is convenient for assembly of the em32 array. Table 1 shows the spherical angles for each microphone channel. The radius of the em32 is 4.2 cm.

| Microphone | Theta | Phi |
|---|---|---|
| 1 | 69 | 0 |
| 2 | 90 | 32 |
| 3 | 111 | 0 |
| 4 | 90 | 328 |
| 5 | 32 | 0 |
| 6 | 55 | 45 |
| 7 | 90 | 69 |
| 8 | 125 | 45 |
| 9 | 148 | 0 |
| 10 | 125 | 315 |
| 11 | 90 | 291 |
| 12 | 55 | 315 |
| 13 | 21 | 91 |
| 14 | 58 | 90 |
| 15 | 121 | 90 |
| 16 | 159 | 89 |
| 17 | 69 | 180 |
| 18 | 90 | 212 |
| 19 | 111 | 180 |
| 20 | 90 | 148 |
| 21 | 32 | 180 |
| 22 | 55 | 225 |
| 23 | 90 | 249 |
| 24 | 125 | 225 |
| 25 | 148 | 180 |
| 26 | 125 | 135 |
| 27 | 90 | 111 |
| 28 | 55 | 135 |
| 29 | 21 | 269 |
| 30 | 58 | 270 |
| 31 | 122 | 270 |
| 32 | 159 | 271 |

● Table 1 Spherical angles of the microphone channels