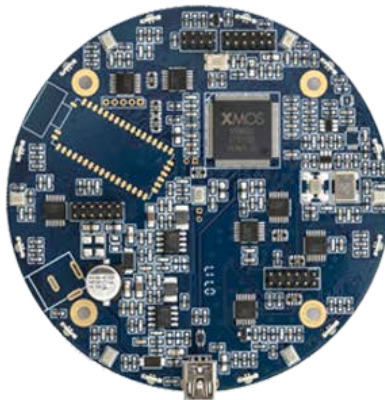


# UMA-8

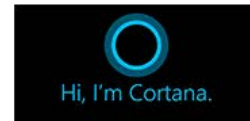
USB MICROPHONE ARRAY WITH EMBEDDED DSP

## User Manual

amazon alexa



UMA-8 Far Field  
USB Mic Array





## Revision history

Revision	Description	Date
1.0	First public release	22 May 2017
1.1	Adding Raw mode details	06 June 2017
1.2	Documenting API command	19 October 2017

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## IMPORTANT INFORMATION

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Please read the following information before use. In case of any questions, please contact miniDSP via the support portal at [minidsp.desk.com](http://minidsp.desk.com).

### SYSTEM REQUIREMENTS – WINDOWS

- 1GHz or higher processor clock speed recommended / Intel® Pentium®/Celeron® family, or AMD K6®/AMD Athlon®/AMD Duron® family, or compatible processor recommended.
- 512 megabytes (MB) of RAM or higher recommended
- One free USB 2.0 port
- Microsoft® Windows® Win7/Win8/Win10

### SYSTEM REQUIREMENTS – MAC OS X

- Intel Core Duo processor or greater
- 256 megabytes (MB) of RAM or higher recommended
- One free USB 2.0 port

### DISCLAIMER/WARNING

miniDSP cannot be held responsible for any damage that may result from the improper use or incorrect configuration of this product. Please read this manual carefully to ensure that you fully understand how to operate and use this product, as incorrect use or use beyond the parameters and ways recommended in this manual have the potential to cause damage to your audio system.

Please also note that many of the questions we receive at the technical support department are already answered in this User Manual and in the online [application notes](#) on the miniDSP.com website. So please take the time to carefully read this user manual and the online technical documentation. Thank you for your understanding!

### WARRANTY TERMS

miniDSP Ltd warrants this product to be free from defects in materials and workmanship for a period of one year from the invoice date. Our warranty does not cover failure of the product due to incorrect connection or installation, improper or undocumented use, unauthorized servicing, modification or alteration of the unit in any way, or any usage outside of that recommended in this manual. If in doubt, contact miniDSP prior to use.

### FCC CLASS B STATEMENT

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

- This device may not cause harmful interference.



- This device must accept any interference received, including interference that may cause undesired operation.

**Warning:** This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

**Notice:** Shielded interface cable must be used in order to comply with emission limits.

**Notice:** Changes or modification not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

## CE MARK STATEMENT

The UMA-8 has passed the test performed according to European Standard EN 55022 Class B.

## PACKAGE CONTENTS

Your *UMA-8* package includes:

- One *UMA-8* USB far field microphone array module
- One USB cable for computer connectivity (1.5m)

## A NOTE ON THIS MANUAL

This User Manual is designed for reading in both print and on the computer. If printing the manual, please print double-sided. The embedded page size is 8 ½" x 11". Printing on A4 paper will result in a slightly reduced size.

For reading on the computer, we have included hyperlinked cross-references throughout the manual. In addition, a table of contents is embedded in the PDF file. Displaying this table of contents will make navigation much easier:

- In Adobe Reader on Windows, click on the "bookmarks" icon at the left. The table of contents will appear on the left and can be unfolded at each level by clicking on the "+" icons.
- In Preview on the Mac, click on the **View** menu and select **Table of Contents**. The table of contents will appear on the left and can be unfolded at each level by clicking on the triangle icons.

# 1 PRODUCT OVERVIEW

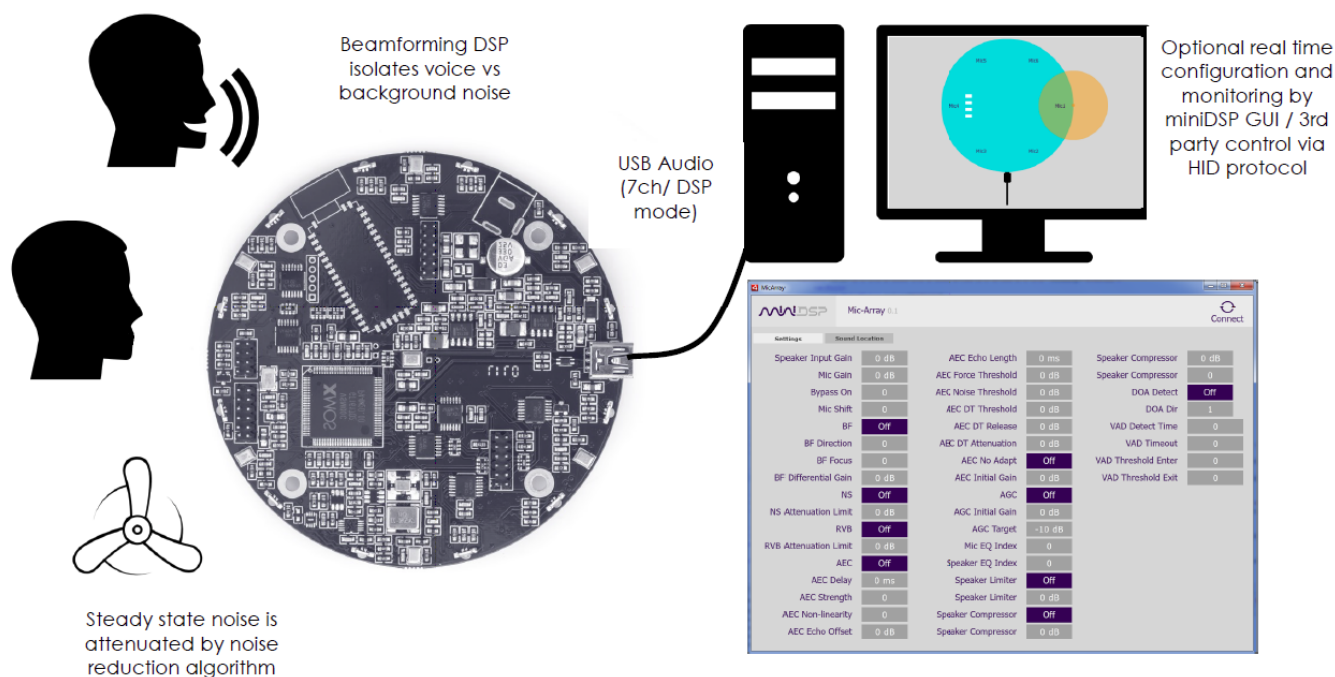
Thank you for purchasing a miniDSP *UMA-8* USB microphone array. The **UMA-8** is a high-performance yet low cost multichannel USB microphone array built around XMOS multicore technology. Seven high-performance MEMS microphones are configured in a circular arrangement to provide high-quality voice capture for a wide range of applications.

Leveraging the onboard DSP processing, the UMA-8 supports voice algorithms including beamforming, noise reduction, acoustic echo cancellation and de-reverb. Non-technical users can enjoy a plug&play experience, while advanced users can fine-tune all DSP parameters with a realtime Win/Mac GUI for optimum performance. The UMA-8 is a fully compliant UAC2 audio interface with driverless support for Mac/Linux and ASIO drivers for Windows.

Depending on the loaded firmware, the UMA-8 has different modes of operation.

For the current release, the interface is running as a 2xIN (2ch Beamforming), 2 x OUT (Stereo out on I2S\_OUT\_0 / Pin J3.1) configuration in the DSP mode. In the RAW mode, the unit is advertising as a 8xIN (7 x MEMS + 1 x PDM input), 2 x OUT (Stereo out on I2S\_OUT\_0 / Pin J3.1).

The below sections will hopefully clarify how to install and configure the UMA-8. If any doubt/questions, feel free to contact our technical support team.



## 2 HARDWARE CONNECTIVITY

### 2.1 BOARD OVERVIEW

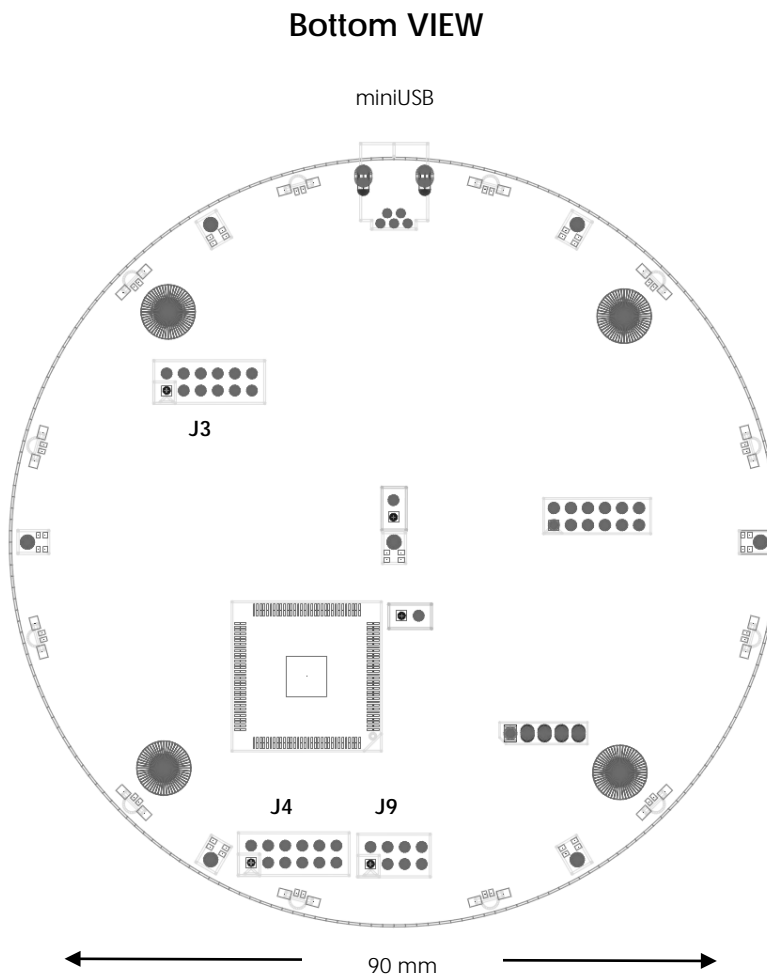
The UMA-8 has few exposed headers to help for customization of the product. Note that MEMS are up-firing( through the board) such that SW1/2/3/4 are on top.

#### J3 / Audio data & clocks

J3.1 - I2S_OUT_0	J3.2 - I2S_IN_0
J3.3 - I2S_OUT_1	J3.4 - I2S_IN_1
J3.5 - I2S_OUT_2	J3.6 - I2S_IN_2
J3.7 - I2S_OUT_3	J3.8 - I2S_OUT_4
J3.9 - MCLK	J3.10 - I2S_BCLK
J3.11 - GND	J3.12 - I2S_LRCLK

#### J2 / Control IO pins

J2.1 - GND	J2.2 - 3.3V
J2.3 - GND	J2.4 - 3.3V
J2.5 - N/A	J2.6 - UART_TX
J2.7 - UART_RX	J2.8 - XMOS_RST
J2.9 - I2C_SDATA	J2.10 - I2C_SCLK
J2.11 - N/A	J2.12 - N/A





## 2.2 USB (AUDIO + DC POWER)

The UMA-8 is USB self powered. A single USB connection to your PC/Mac will suffice to provide power to the unit and stream bidirectional audio simultaneously.

To record audio from the *UMA-8*, connect the USB port of the *UMA-8* to a USB 2.0 port on your computer using the supplied cable. The *UMA-8* should also be connected for initial driver installation under Windows.

## 2.3 I2S EXPANSION HEADER PINOUTS

Headers J3 is provided for connection of I/O circuitry via I2S. The pinouts are shown in Table 1.

In the current firmware, only J3.1 / I2S\_OUT0 is enabled as “**SPEAKER OUTPUTS**”

*Table 1. J3 expansion header pinout*

J3.1 - I2S_OUT_0	J3.2 - I2S_IN_0
J3.3 - I2S_OUT_1	J3.4 - I2S_IN_1
J3.5 - I2S_OUT_2	J3.6 - I2S_IN_2
J3.7 - I2S_OUT_3	J3.8 - I2S_OUT_4
J3.9 - MCLK	J3.10 - I2S_BCLK
J3.11 - GND	J3.12 - I2S_LRCLK

## 2.4 I2S OVERVIEW

I2S, or Inter IC Sound, is an electrical serial bus used to interface digital audio devices at the chip and circuit board level. An I2S interface consists of up to three clocks, and a data line for each pair of channels. There are three types of clock:

- MCLK**      The master clock that the *UMA-8* uses internally. This clock is always provided as an output by the *UMA-8*, and connected circuitry can choose whether or not to use it.
- LRCLK**    The frame synchronization clock, also known as the word clock. This clock is equal to the sampling frequency ( $F_s$ ) of the audio signal.
- BCLK**      The bit clock (also known as shift clock or system clock). This is always equal to  $64 \times F_s$ .

*Table 2. I2S clock ratios*

Firmware	Sample Rate	Master clock (MCLK)	Bit clock (BCLK)	MCLK/LRCLK
2ch DSP Mode	16 kHz	24.576 MHz	1.024 MHz	1536
8ch RAW Mode	11.2/16/32/44.1/48kHz	24.576 MHz	1.024 MHz	

The timing of data lines is determined by the bit clock and the word clock, as illustrated in the following diagram:

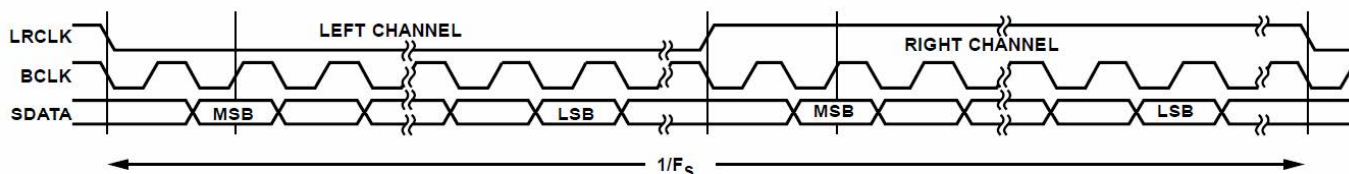


Figure 31. I2S Mode—16 Bits to 24 Bits per Channel

The UMA-8 board has four I2S input data lines and four I2S output data lines, each carrying two channels of audio. Note, however, that current firmware do not make use of all I2S in/out lines. The below table will be updated over time as we add new features to the firmware.

Table 3. Input and output mapping

FW	Recording (PC in)	Playback (PC Out)
2ch DSP mode	8ch beamformed to 2ch	I2S_OUT0
8ch RAW mode	8ch RAW channels	I2S_OUT0

The playback output, also called “SPEAKER OUTPUTS” through out this section is to be connected to an external I2S amplifier or I2S DAC. Feel free to contact miniDSP tech support if you have some specific questions.

## 2.5 ADDITIONAL I2S USAGE NOTES

Note that I2S is not a “plug and play” protocol. It requires attention to technical details such as clocking and wire layout. It is a solution for OEMs and *advanced* DIYers (or professionals) with suitable knowledge, skills and measurement equipment.

Be sure to take the following precautions when designing your I2S interface and wiring:

### General I2S usage notes

- Unbuffered I2S lines must be kept short to ensure clock and data integrity.
- If driving longer lines, buffers may be required for the clock signals (MCLK, LRCLK, and BCLK).
- Observe correct grounding and shielding, and keep analog and digital grounds separated.
- Ensure that the clock ratios (as listed in Table 2) are compatible with connected circuits.

### 3.3V logic level

All lines use a 3.3V logic level. Ensure that connected circuits use a compatible level (1.8V, for example, will not work).

## 3 DRIVER INSTALLATION AND CONFIGURATION – WINDOWS

---

### 3.1 INSTALLATION

The UMA-8 is a USB Audio Class 2.0 device. For use with Microsoft Windows, driver installation is required.

#### 3.1.1 Download the latest driver

When you receive notification that your order has shipped, your installation software download will be available at the *User Downloads* section of the miniDSP website:

<http://www.minidsp.com/userdownloads>

(If you are unable to access this section of the website, please login first.)

Download the installation zip file under the **USB Microphone array series** heading and unzip the folder on your PC.

#### 3.1.2 Connect your UMA-8

In order to automatically detect the UMA-8 during driver installation, connect the UMA-8 to your PC and power it on before going to the following step.

#### 3.1.3 USB Driver installation

1. Connect the UMA-8 to the computer using the supplied USB cable, and power it on.
2. Navigate to the **WinDrivers** folder of the software download and double-click on the appropriate installer:
  - **miniDSP\_UAC2\_v3.34.0\_ForWin7\_8\_10.exe** for Windows 7, 8, and 10

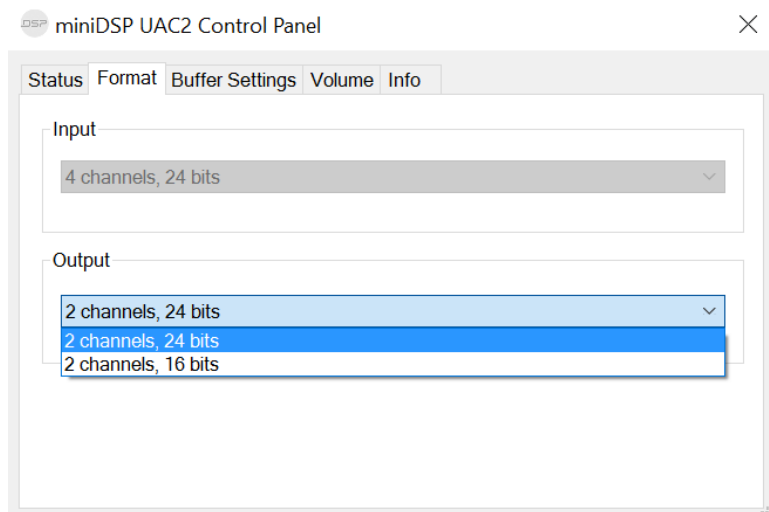
We recommend accepting the default installation location. Once the driver installation completes, click the **Finish** button.

## 3.2 CONFIGURATION

To configure the UMA-8, open the **USB Audio Control Panel** (from Start Menu -> miniDSP Ltd -> UAC2 Control Panel). It has several panes, described below.

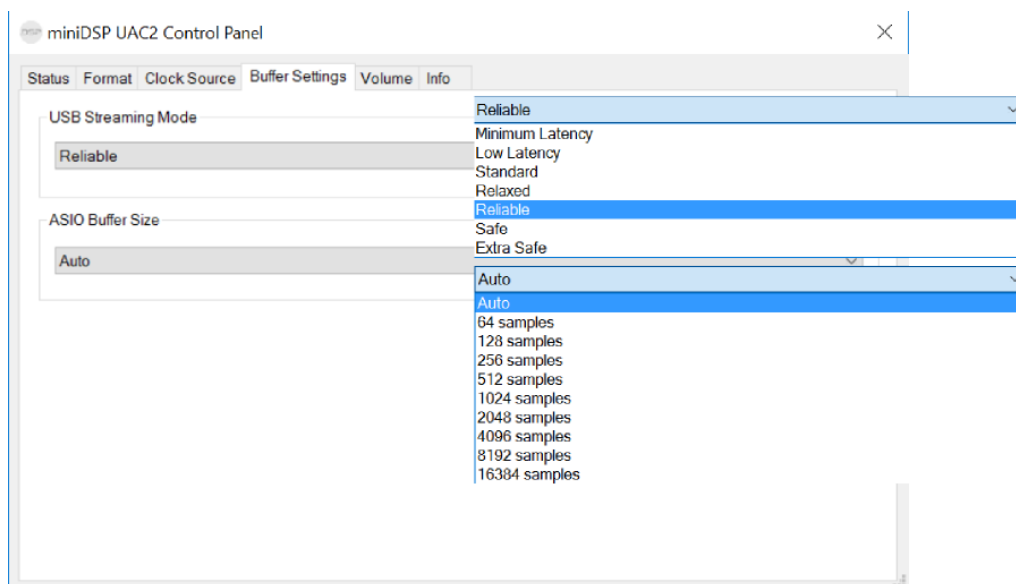
### 3.2.1 Format

This panel selects the input and output data format. The screenshot here shows the DSP mode.



### 3.2.2 Buffer settings

The buffer settings are for those looking to optimize buffering and latency settings. Note that changing these settings may result in unstable operation. For example, the lowest latency settings require high amounts of CPU and memory, and may not work on some machines. If you do not require lowest latency, we recommend that you do not depart from the default safe settings.



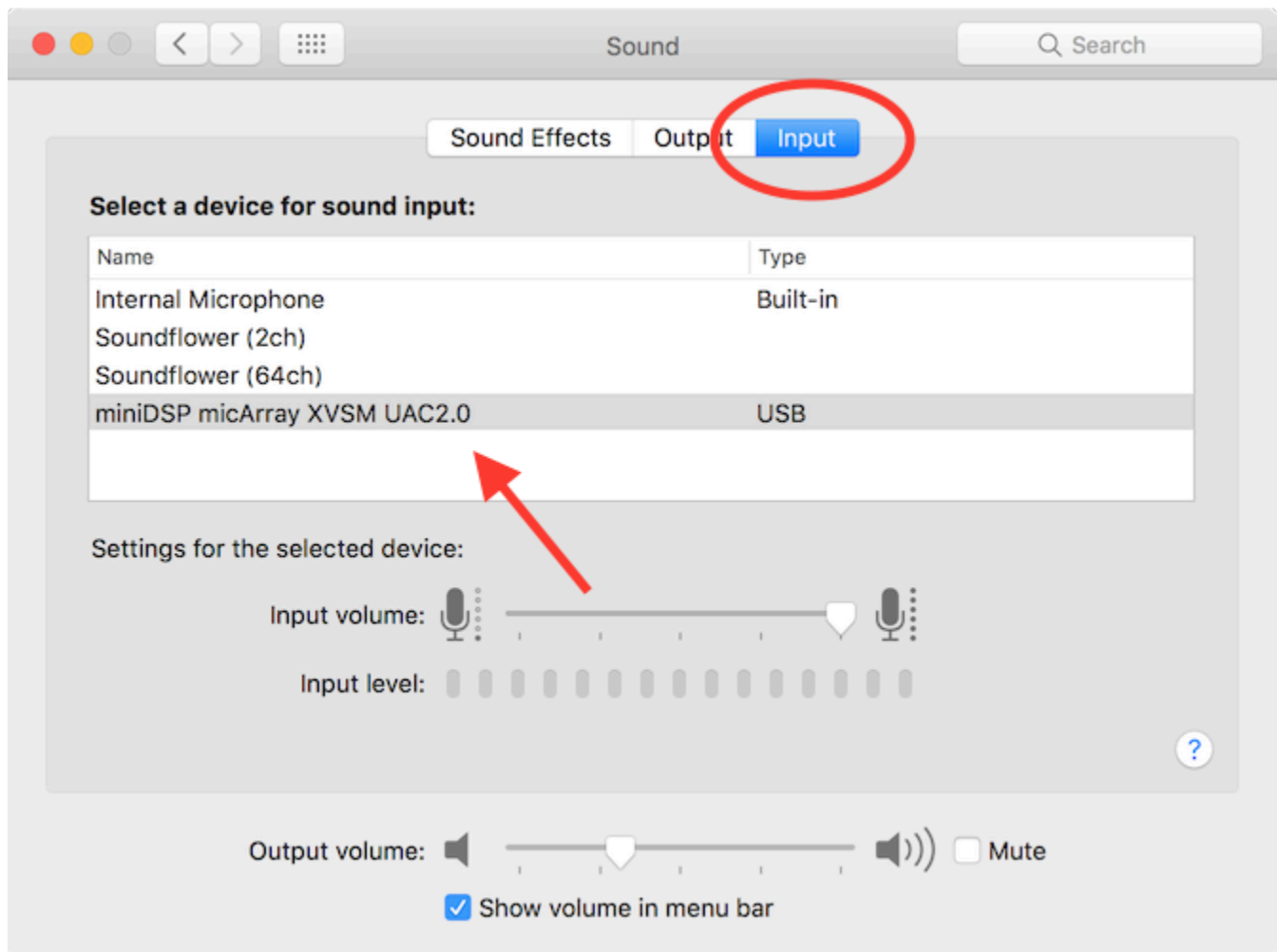
## 4 DRIVER INSTALLATION AND CONFIGURATION – MAC OS X

### 4.1 INSTALLATION

Mac OS X has native support for USB Audio class 2.0 devices, so no driver installation is required. The UMA-8 will automatically be detected by Mac OS X as a compliant multichannel USB audio interface.

### 4.2 CONFIGURATION

Open the program **Audio MIDI Setup** (in **Applications->Utilities**). The UMA-8 will appear automatically in the list on the left hand side, as a **miniDSP micArray XVSM UAC2.0**



## 5 DRIVER INSTALLATION AND CONFIGURATION – LINUX

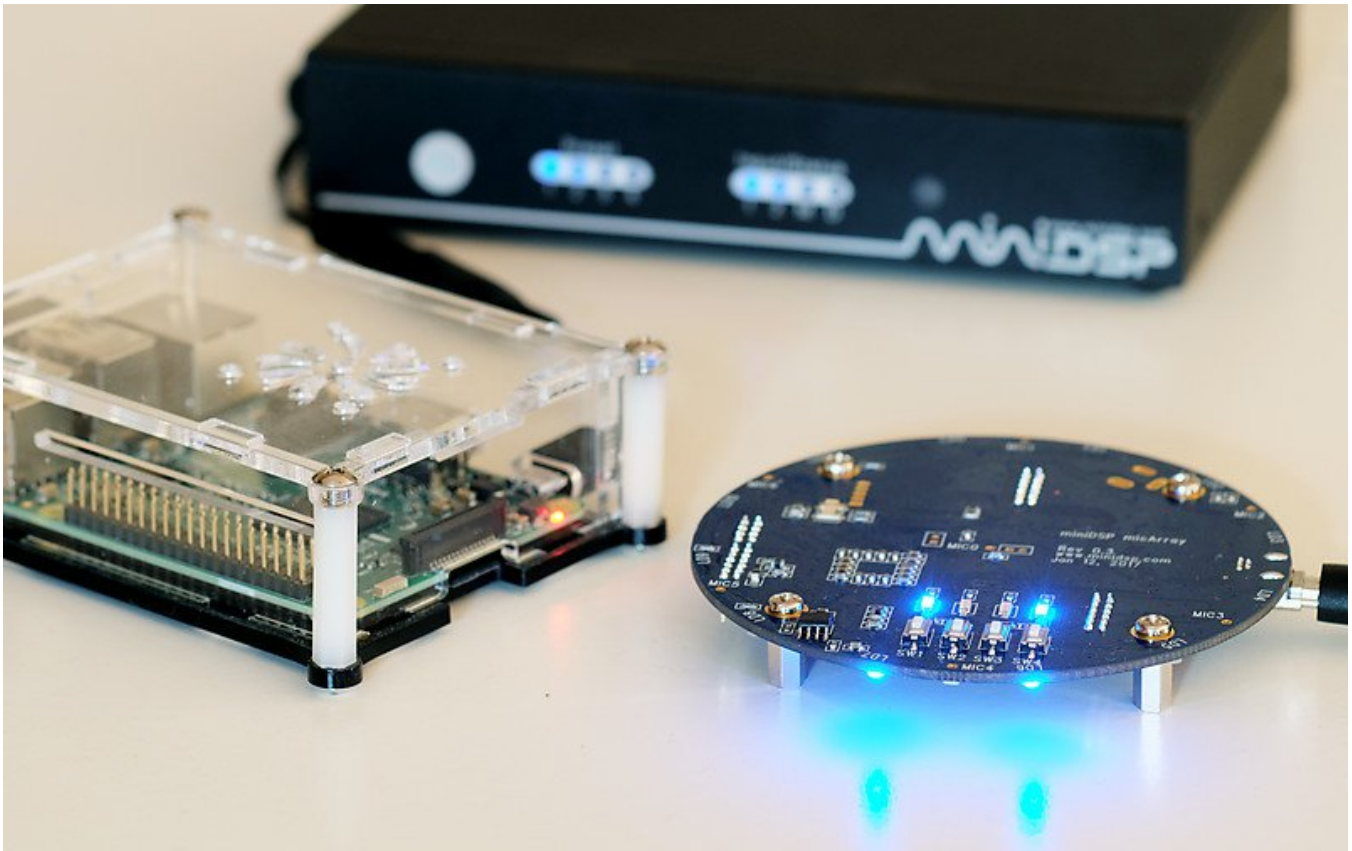
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### 5.1 ALSA INSTALLATION

The UMA-8 is a fully compliant UAC 2 (USB Audio Class) device. While miniDSP's technical support team unfortunately can't provide email/phone support for Linux due to the large number of distributions, we know from experience that the UMA-8 is Plug&Play with Alsa 2.0 drivers. For more information/support on ALSA driver, please consult [ALSA project](#).

### 5.2 RASPBERRY PI

The UMA-8 was tested to be plug&play with AlexaPi. A complete app note with step by step instructions to build your DIY Alexa speaker is provided at the [following link](#).



## 6 SOFTWARE INSTALLATION

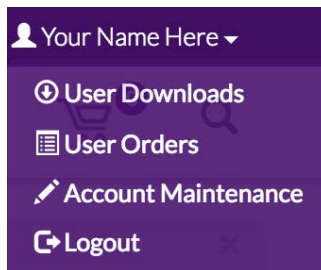
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If you purchased your product directly from miniDSP, your software will be available from the [User Downloads](#) section of the miniDSP website when your order ships.

If you purchased your product from a miniDSP dealer, you will receive a coupon together with the product. Redeem this coupon and select the Plugin Group “UMA-8” at the link below:

- <https://www.minidsp.com/support/redeem-coupon>

To access the download, you will need to be logged into the miniDSP.com website with the account you created when purchasing. The User Downloads link is visible from the dropdown menu at the top right of the website page:



Navigate to the **UMA-8** section and download the zip file. Unzip the downloaded file: on Windows, right-click and select “Extract All...”; on Mac, double-click.

### 6.1 WINDOWS INSTALLATION

The miniDSP software requires that other frameworks are installed for it to work. For Windows 7 and later, these packages should be installed automatically by the miniDSP installer.

For Windows XP and Vista, please download and install the following frameworks before attempting to install any miniDSP software:

- [Microsoft .NET framework](#) (version 3.5 or later)
- Latest version of [Adobe Air](#)
- Microsoft Visual C++ 2010 Redistributable Package: for [x86](#) (32-bit operating system) or [x64](#) (64-bit operating system)

To install the plugin:

3. Navigate to the **Windows** folder of the software download.
4. Double-click on the **Mic-Array.exe** installer program to run it. We recommend that you accept the default installation settings.
5. The plugin will start automatically if you accepted the default installation settings. To make it quicker to run in future, right-click on its icon in the taskbar and select “Pin to taskbar.”



**Note 1:** The Adobe Air framework may need to connect to the Internet the first time you run the plugin.

**Note 2:** The first time you run the plugin, you may see a warning from Windows Firewall asking whether the software should be allowed network access. If you do, ensure that “Private networks...” is checked and “Public networks...” is not checked. Then click on “Allow access.”

## 6.2 MAC OSX INSTALLATION

To install the plugin:

6. Navigate to the **Mac** folder of the software download.
7. The installer program is named **MicArray.pkg**. To run it, double-click on it, or right-click and open as described below. We recommend that you accept the default installation settings.
8. To run the plugin, locate **Mic-Array.app** in the Applications -> miniDSP folder and double-click on it. To make it easier to run in future, right-click on its dock icon and select Options -> Keep in Dock.



## 7 CONFIGURING DSP SETTINGS

The UMA-8 is a versatile microphone array which can operate in 2 modes:

- **DSP mode:** The 7 MEMS are beamformed to a 2ch signal for the PC. Advanced settings of the GUI are available for control. Audio output is 16kHz. A stereo output is being sent to the I2S\_OUT0 (J3.1header). A 3<sup>rd</sup> party external board is therefore required to output analog audio(e.g. I2S amplifier module or I2S DAC module)
- **RAW mode:** 8ch of audio (7ch coming from the MEMS microphone + 1 ch from spare PDM input) are available as raw audio (non processed). Sample rates are available as 11.2/16/32/44.1/48kHz. Note that in RAW mode, the UMA-8 will output the RAW signal from the MEMS without any digital gain (typically applied in 2ch DSP mode). You may require your own application to boost the signal.

The below information will highlight settings for the DSP mode. In the RAW mode, no processing is enabled and no configuration is available.

**NOTE:** The UMA-8 MicArray tool targets advanced users looking for customization (E.g. DSP features) and some settings might render the UMA-8 unusable. In the event that you'd like to "Restore to default" settings, follow this operation:

- 1) Unplug USB 2) Press Switch 2&3 (SW2&3 on top of board) 3) Plug USB again.

The below GUI is a real time interface which will sync onboard setting when you CONNECT to the board.

Any changes/modifications are real time and will be stored on the onboard flash for subsequent reboots.


Mic-Array 1.0
Connect

Settings

Sound Location

Speaker Input Gain	0 dB	AEC Echo Length	0 ms	Compressor Gain	0 dB
Mic Gain	0 dB	AEC Force Threshold	0 dB	Compressor Ratio	0
Bypass On	0	AEC Noise Threshold	0 dB	DOA Detect	Off
Mic Shift	0	AEC DT Threshold	0 dB	DOA Dir	1
BF	Off	AEC DT Release	0 dB	VAD Detect Time	0
BF Direction	0	AEC DT Attenuation	0 dB	VAD Timeout	0
BF Focus	0	AEC No Adapt	Off	VAD Threshold Enter	0
BF Differential Gain	0 dB	AEC Initial Gain	0 dB	VAD Threshold Exit	0
NS	Off	AGC	Off		
NS Attenuation Limit	0 dB	AGC Initial Gain	0 dB		
RVB	Off	AGC Target	-10 dB		
RVB Attenuation Limit	0 dB	Mic EQ Index	0		
AEC	Off	Speaker EQ Index	0		
AEC Delay	0 ms	Limiter	Off		
AEC Strength	0	Limiter Threshold	0 dB		
AEC Non-linearity	0	Compressor	Off		
AEC Echo Offset	0 dB	Compressor Threshold	0 dB		

### 7.1.1 UMA-8 DSP MODE

Bypass On	0
Mic Shift	0

Bypass\_on [0,1] / Modes for DSP processing

0: normal operation / Both microphone processing and speaker processing is enabled

1: bypass microphone processing, speaker processing remains

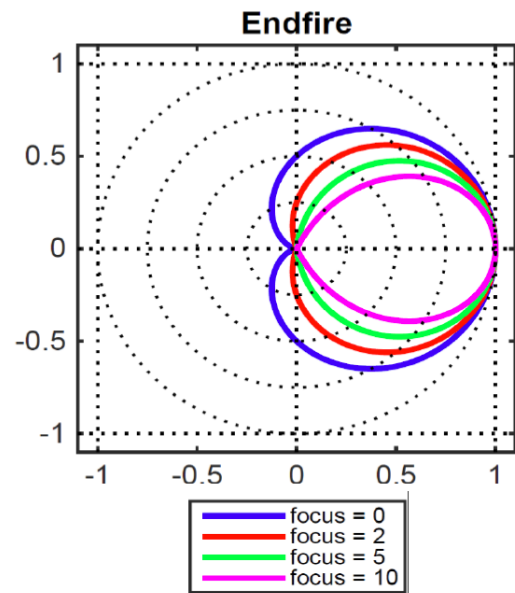
2: pure bypass where both the speaker processing AND microphone processing is disabled

Mic\_shift [0..4]: left shift of microphone input signal can be used to boost the gain of the MEMS. Use this caution as it could create internal DSP saturation\*/

### 7.1.2 BEAMFORMER

The UMA-8 is using a 2ch highly directive end-fire beamformer to isolate voice from noise. The beamformer uses a time-frequency adaptive LMS algorithm with differential microphone pre-processing. The beamformer is suitable for non-stationary sound stations, and integrates seamlessly with AEC and NS algorithms.

BF	Off
BF Direction	0
BF Focus	0
BF Differential Gain	0 dB



BF: [0, 1]: 0: Beamformer off, Beamformer 1: on

BF\_direction: [0..3]: 0: 360 degree, 1: end-fire 1, 2: end-fire 2, 3: line array

BF\_focus [0..10]: Beamformer focus controls the endfire polar pattern of the microphone. See below for chart

BF\_diffgain\_dB; /\* [-20..0]: diffuse sound gain [dB] controls the de-reverb processing.

### 7.1.3 NOISE SUPPRESSION

This setting controls the Noise Suppression algorithm for up to 20dB of Noise Reduction.

NS	Off
NS Attenuation Limit	0 dB

NS: [0, 1]: 0: off, 1: on \*/

NS\_attlimit\_dB: [-20..0]: noise attenuation in dB

### 7.1.4 DE\_REVERB

The De-reverb technology removes room reverberation effects .It gives the feeling the talker is closer to the microphone and can remove up to 20dB of non –stationary reverb.

RVB	Off
RVB Attenuation Limit	0 dB

rvb\_on; /\* [0, 1]: 0: off, 1: on \*/

rvb\_attlimit\_dB; /\* [-20..0]: reverb attenuation in dB \*/

### 7.1.5 ACOUSTIC ECHO CANCELLATION (AEC)

The onboard AEC block can be used to remove echo during.

AEC	Off
AEC Delay	0 ms
AEC Strength	0
AEC Non-linearity	0
AEC Echo Offset	0 dB

aec\_on; /\* [0, 1]: 0: off, 1: on \*/

aec\_delay\_ms; /\* [-1, 0..30]: -1: auto, 0..30 AEC delay [ms] \*/

aec\_strength; /\* [0..10]: Echo suppression strength \*/  
 aec\_nonlin; /\* [0..10]: non-linearity modeling \*/  
 aec\_lecho\_offset\_dB; /\* [-80..0]: late echo estimation offset [dB] \*/

AEC Echo Length	0 ms
AEC Force Threshold	0 dB
AEC Noise Threshold	0 dB
AEC DT Threshold	0 dB
AEC DT Release	0 dB
AEC DT Attenuation	0 dB
AEC No Adapt	Off
AEC Initial Gain	0 dB

aec\_lecho\_len\_ms; /\* [0..400]: late echo length time constant [ms] \*/  
 aec\_force\_thr\_dB; /\* [-80..0]: threshold below which echo removal is put to maximum \*/  
 aec\_noise\_thr\_dB; /\* [0..40]: when ERLE < aec\_noise\_thr, then noise echo is not cancelled \*/  
 aec\_dt\_thr\_dB; /\* [0..40]: when ERLE > aec\_dt\_thr in dB, then doubletalk release will be disabled \*/  
 aec\_dt\_release\_dB; /\* [0..24]: during near-end and doubletalk echo removal is released [dB] \*/  
 aec\_dt\_att\_limit\_dB; /\* [-80..0]: maximum echo attenuation when ERLE < aec\_dt\_thr [dB] \*/  
 aec\_no\_adapt; /\* [0, 1]: 0: aec echo path estimation on, 1: set echo path to aec\_gain \*/  
 aec\_init\_gain\_dB; /\* [-60..20]: echo path initial gain (if aec\_noadapt then update aec gains to aec\_gain) [dB] \*/

### 7.1.6 AUTO GAIN CONTROL (AGC)

This setting adjusts the microphone level to maintain the desired target. It's handy setting to insure that a constant level is achieved. Note that the AGC takes a bit of time to adapt to environment and train its settings.

AGC	Off
AGC Initial Gain	0 dB
AGC Target	-10 dB

```
agc_on; /* [0, 1]: 0: off, 1: on */
agc_init_gain_dB; /* [0..36 dB]: default gain when AGC starts */
agc_target_dB; /* [-30..-10]: agc target [dB] when UMA-8 is operational */
```

### 7.1.7 EQUALIZER SELECTION / FUTURE NOT ENABLED AT THIS TIME.

The following control enable up to 5 bands of basic EQ to be applied on the SPEAKER output for smart speakers.

Mic EQ Index	0
Speaker EQ Index	0

**NOTE** : This feature isn't enabled at this point of time. Will be enabled in future firmware revisions.

```
mic_eq_index; /* [0..2]: mic EQ preset index */
spk_eq_index; /* [0..2]: spk EQ preset index */
```

### 7.1.8 LOUDSPEAKER COMPRESSOR/LIMITER

The UMA-8 has embedded loudspeaker compressor/limiter to protect the speaker output (2ch playback).

Limiter	Off
Limiter Threshold	0 dB
Compressor	Off
Compressor Threshold	0 dB

```
spk_limiter_on; /* [0, 1]: 0: audio gain control off, 1: audio gain control on */
spk_limiter_thr_dB; /* [-20..0]: limiter threshold [dB] */
spk_compr_on; /* [0, 1]: 0: audio gain control off, 1: audio gain control on */
spk_compr_thr_dB; /* [-30..0]: compressor threshold [dB] */
```

Compressor Gain	0 dB
Compressor Ratio	0

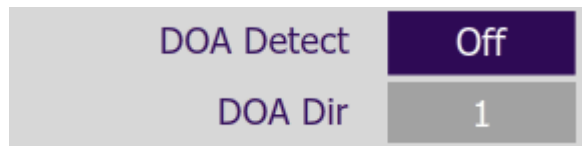
```
spk_compr_gain_dB; /* [0..12]: compressor makeup gain [dB] */
```



```
spk_compr_ratio; /* compressor ratio [index] /\nindex 0 1 2 3 4 5 6 7 8 9\nratio 1.5 2 3 4 5 6 7 8 9 10 */
```

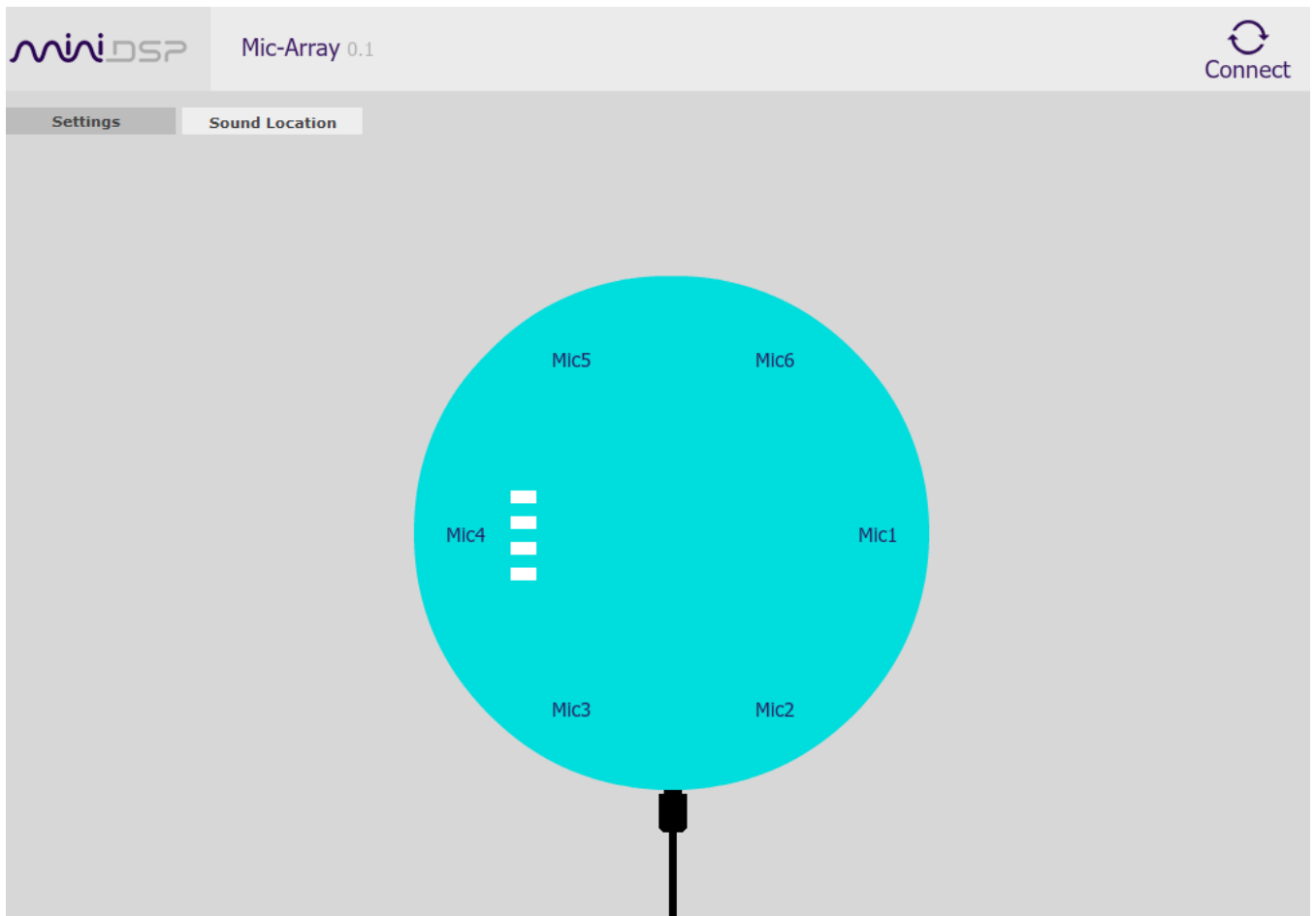
### 7.1.9 Direction of Arrival (DOA)

On the second tab of the MicArray GUI, one can monitor real time the Direction Of Arrival (DOA) as detected by the algorithm. The DOA also controls the RGB led illumination (i.e.2 active microphones) End users can easily perform the same DOA detection inside their own software thanks to the 3<sup>rd</sup> party USB library documented in “USB API/CUSTOMIZATION” section.



DOA Detect / ON = enabled, OFF = No detection and LED control.

DOA Dir = Manual control of the active microphones when DOA is turned OFF



### 7.1.10 Voice Activated Detection (VAD)

The Voice Detection algorithm can be used to detect voice vs noise.

- VAD Detect time (samples) / Default 2: Equivalent to the attack time before VAD is enabled.
- VAD TimeOUT (samples) / Default 50 (3ms): Release time

VAD Detect Time	0
VAD Timeout	0
VAD Threshold Enter	0
VAD Threshold Exit	0

### 7.1.11 Control Switches

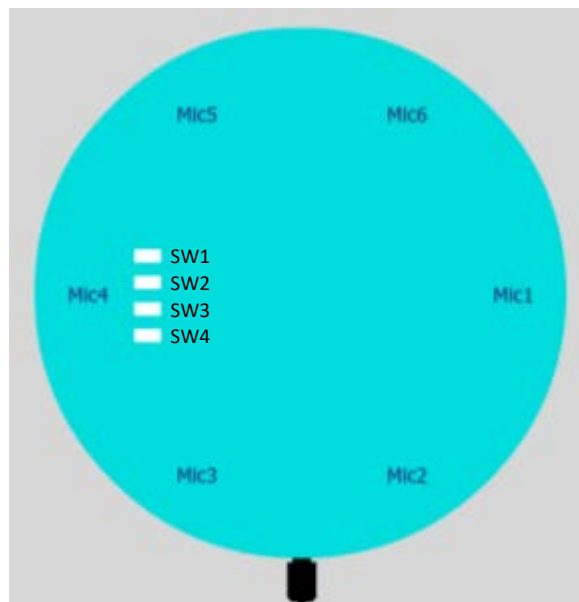
The UMA-8 is fitted with 4 control switches on the top side of the module with the following use:

**SW1:** Bypass/activates all microphone processing. The processing is active when the application starts.

**SW2:** Mic monitoring enable/disable. Enabling the monitor allows the microphone signal to be directly monitored on the I2S data 1&2 out; Disabling allows audio playback from the host i.e. normal USB DAC. The monitor is disabled when the application starts.

**SW3:** Manually rotate beam direction to one of the 6 microphone direction. The beam direction is shown by the LEDs.

**SW4:** DOA mode enable/disable



## 8 USB API / CUSTOMIZATION

For advanced DIYers, the UMA-8 may be controlled directly using a basic USB API. Note that the below sections assumes you have basic knowledge of USB HID libraries. miniDSP unfortunately won't be able to provide support to debug/test your code or provide support on basics of HID control. The good news is that there isn't shortage of USB API libraries online. ☺

### 8.1.1 Discovery of UMA-8

On power up, Mic-Array will be recognized by the computer as an HID device with Vendor ID=0x2752. Product ID=0x1C, some USB library can be used to connect to this device and send / receive commands to / from it.

### 8.1.2 Voice Activated Detection (VAD) + Direction Of Arrival (DOA)

When voice is detected (VAD active) or the direction of arrival (DOA) changed, the microphone array will send a command to the USB host through USB interrupt. The command format is shown below:

Byte 1	Byte 2	Byte 3	Byte 4	Byte 5	Byte 6
0x06	0x36	VAD	Angle.HB	Angle.LB	Dir

VAD = 1 / Voice is detected, the Dir byte has to be read to determine the Direction of Arrival.

VAD = 0 / No more voice signal detected

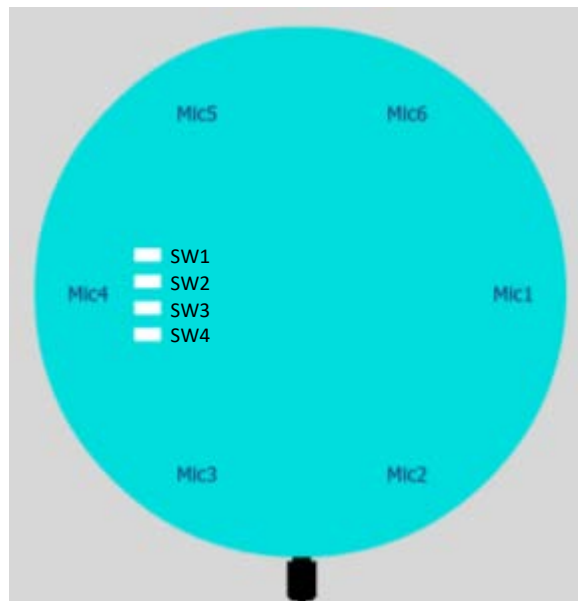
Dir = Direction of Arrival, represented by a number correspond to mic position. Can range from 1 ~ 6 for Mic1~6 as per below diagram. .

Angle = angle of the direction number as specified by Dir, in degrees. Range from 0 ~ 360

Angle.HB = high byte of the angle

Angle.LB = low byte of the angle

The mic position, labelled by the mic number is shown in the figure below:





### 8.1.3 Turn ON/OFF Direction of Arrival (DOA)

To turn on / off DOA detect, send the command through USB to

Byte 1	Byte 2	Byte 3	Byte 4	Byte 5
0x09	0x13	Action	0x00	0x24

Byte 6	Byte 7	Byte 8	Byte 9	Byte 10
On / Off	0x00	0x00	0x00	CRC

Action = 0xA0 , load values to DSP but not load to memory

Action = 0x80 , load values to DSP and memory

On / Off = 0x00 (to turn off DOA detect)

On / Off = 0x01 (to turn on DOA detect)

CRC = sum of all bytes except the CRC byte itself (i.e. Byte 1 +  
Byte 2 + Byte 3 + ... + Byte 9)

For example to turn on DOA detect (and to keep it turned on even after power-cycled, use Action = 0x80), send this command: 09 13 80 00 24 01 00 00 00 C1

### 8.1.4 Set Direction of Arrival (DOA)

To set DOA direction, send the command through USB to MicArray:

Byte 1	Byte 2	Byte 3	Byte 4	Byte 5
0x09	0x13	Action	0x00	0x25

Byte 6	Byte 7	Byte 8	Byte 9	Byte 10
Dir	0x00	0x00	0x00	CRC

Action = 0xA0 , load values to DSP but not load to memory

Action = 0x80 , load values to DSP and memory

Dir = an integer number representing a direction

(valid range is 1 ~ 6 for six directions)

CRC = sum of all bytes except the CRC byte itself (i.e. Byte 1 + Byte 2 + Byte 3 + ... + Byte 9)

For example to set DOA direction to number 1 (and to keep this setting even after power-cycled, use Action = 0x80), send this command: 09 13 80 00 25 01 00 00 00 C2

## 9 ADDITIONAL INFORMATION

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### 9.1 SPECIFICATIONS

Item	Description
USB streaming engine	XMOS XSVM 2000 - Multicore USB audio processor with embedded DSP
USB audio capabilities	USB audio recording in 2 possible modes depending on firmware: - 8-channel mode (7 x MEMS installed + 1 x spare PDM port in the center) - Stereo recording with DSP processing enabled USB audio playback: Stereo I2S channel for I2S out (e.g. external amplifier/DAC board.)
DSP processing (prebuilt firmware)	Beamforming with configurable beam width (up to 20dB attenuation)  - Perceptual acoustic echo cancellation (up to 80dB attenuation)  - Noise suppression (up to 20dB attenuation)  - De-reverb ( up to 20dB attenuation)  - Manual mode for control of beam forming
UAC2.0 drivers	Driverless interface for Mac OS X v10.6.4 and up Thesycon Windows ASIO driver (All versions) Linux Alsa 2.0 compliant Control via HID interface for advanced settings and active microphone
Resolution / Sample rate	24bit @ 11/16/32/44.1/48 kHz
I2S port	Output port for PDM to I2S conversion (upcoming firmware update required)
MEMS microphones	7 x Knowles SPH1668LM4H with low noise buffer and high performance modulator - Low distortion: 1.6% @ 120 dB SPL - High SNR: 65 dB and flat frequency response - RF shielded against mobile interference - Ominidirectional pick-up pattern
LED	12 x RGB LED / Bottom mounted - Circular light guide included
Expansion connector	2 x 12-pin, 2 mm pitch expansion connector for connectivity to hardware.  XMOS JTAG connector for custom code.
Power supply	USB powered
Dimensions (diameter) mm	90 mm diameter / 20mm height with LED ring, 14mm height without LED ring

## 9.2 FIRMWARE UPGRADE

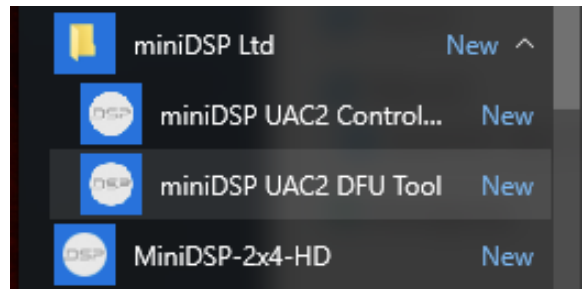
miniDSP may occasionally provide an upgrade to the UMA-8 MCU firmware to enable new features. To upgrade the MCU firmware, first download and install the latest version of the Mic Array plugin from the **User Downloads** section of the miniDSP website.



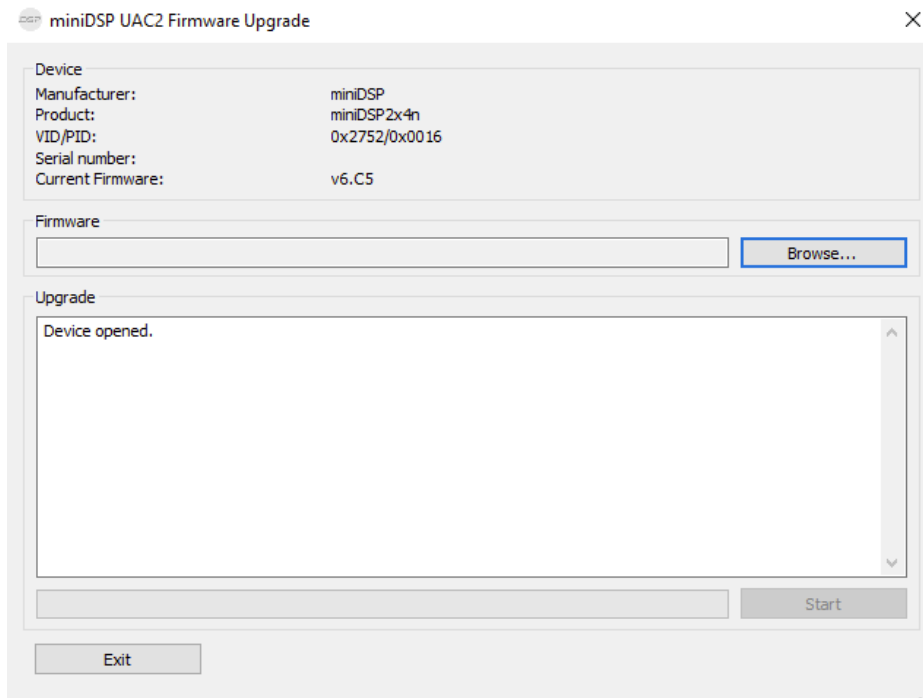
DO NOT DISCONNECT THE USB CABLE OR POWER FROM THE *UMA-8* WHILE FIRMWARE UPGRADE IS IN PROGRESS. DOING SO MAY “BRICK” YOUR *UMA-8*

### 9.2.1 Windows

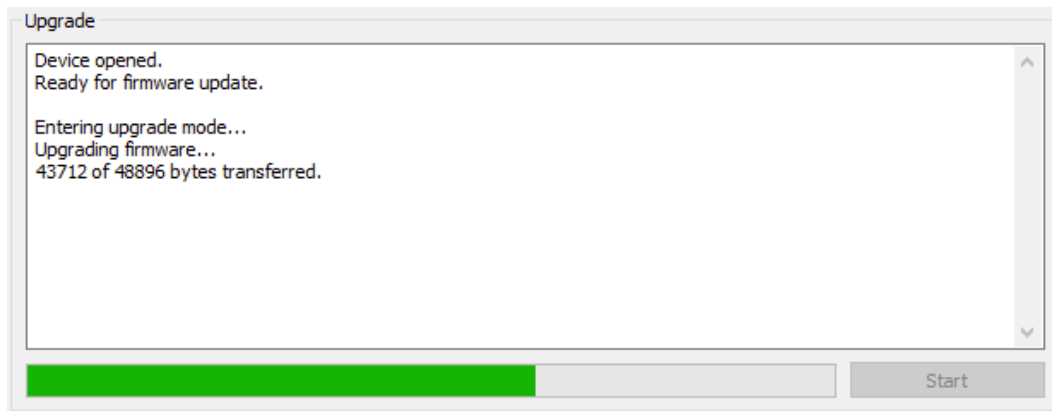
1. Connect the UMA-8 to your computer via USB (if not already connected) and power it on.
2. Start the **miniDSP UAC2 DFU Tool**.



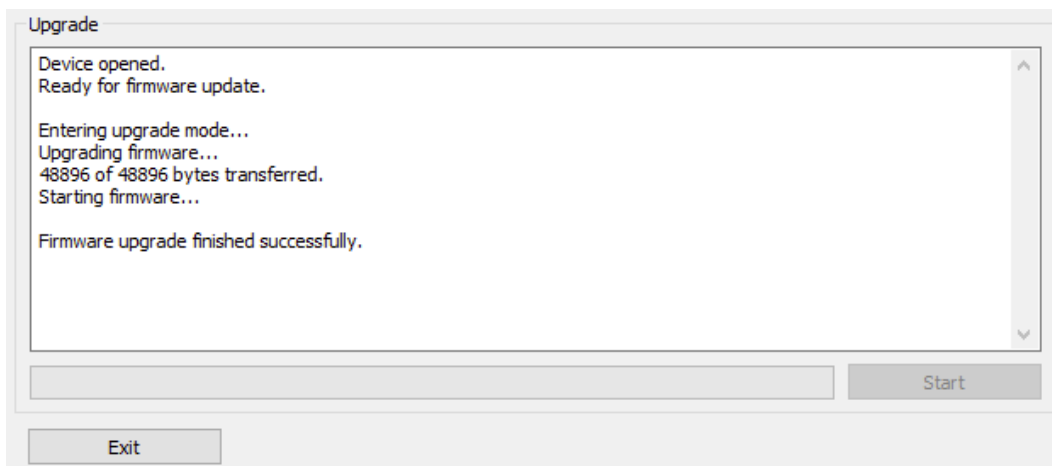
3. The upgrade program will start:



4. Click on the **Browse** button, navigate to the folder **XMOS\_Firmware** in the plugin download folder, and select the firmware file.
5. Click on the **Start** button.
6. You will get a progress bar as upgrade proceeds:



7. Once the firmware upgrade completes, you will see a message that the upgrade completed successfully:



8. Click on **Exit**.
9. That's it! You're done. You can now use your 2x4 HD with the new functionality.

### 9.3 OBTAINING SUPPORT

1. Check the forums on [minidsp.com](http://minidsp.com) to see if this issue has already been raised and a solution or solutions provided.
2. Contact miniDSP via the support portal at [support.minidsp.com](http://support.minidsp.com) with:
  - a. The product information including OS version and version of driver installed (for Windows).
  - b. A clear explanation of the symptoms you are seeing.
  - c. A description of the troubleshooting steps you performed and the results obtained.