NANO SHARC

2-IN 8-OUT I²S AUDIO PROCESSOR BOARD

WITH FIR PROCESSING CAPABILITY

User Manual
### Revision history

<table>
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<th>Revision</th>
<th>Description</th>
<th>Date</th>
</tr>
</thead>
<tbody>
<tr>
<td>V1.0</td>
<td>First version</td>
<td>22 Sept 2016</td>
</tr>
<tr>
<td>V1.1</td>
<td>Updated for dedicated plugin. New firmware update tools.</td>
<td>31 Aug 2018</td>
</tr>
<tr>
<td>V1.2</td>
<td>Link to nanoSHARC 2x8 plugin, updated screenshots.</td>
<td>31 Aug 2018</td>
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miniDSP Ltd, Hong Kong / www.minidsp.com / Features and specifications subject to change without prior notice
IMPORTANT INFORMATION

Please read the following information before use. In case of any questions, please contact miniDSP via the support portal at minidsp.desk.com.

SYSTEM REQUIREMENTS

To configure the miniDSP audio processor, you will require a Windows PC or Apple Mac OS X computer with the following minimum specification:

Windows
- PC with 1GHz or higher processor clock speed. Intel® Pentium®/Celeron® family, or AMD K6®/AMD Athlon®/AMD Duron® family, or compatible processor recommended.
- 512 megabytes (MB) of RAM or higher
- Keyboard and mouse or compatible pointing device
- USB 2.0 port
- Microsoft® Windows® Vista® SP1/ XP pro SP2/Win7/Win8.1/Win10
- Microsoft® .NET framework v3.5 or later
- Adobe AIR environment (latest version)
- Adobe Flash player (latest version)

Mac OS X
- Intel-based Mac with 1 GHz or higher processor clock speed
- 512 megabytes (MB) of RAM or higher
- Keyboard and mouse or compatible pointing device
- USB 2.0 port
- OS X 10.8 or higher, macOS 10.12 or higher
- Adobe AIR environment (latest version)
- Adobe Flash player (latest version)

DISCLAIMER/WARNING

miniDSP cannot be held responsible for any damage that may result from the improper use of this product or incorrect configuration of its settings. As with any other product, we recommend that you carefully read this manual and other technical notes to ensure that you fully understand how to operate this product. The miniDSP audio processor is a powerful tool, and misuse or misconfiguration, such as incorrectly set gains or excessive boost, can produce signals that may damage your audio system.

As a general guideline, you should perform the initial configuration of the miniDSP audio processor before enabling audio through any connected output device or amplification. Doing so will help ensure that the software is correctly configured.
Finally, note that the miniDSP audio processor is a very flexible device, and many of the questions we receive at the tech 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 support. Thanks 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 nanoSHARC has passed the test performed according to European Standard EN 55022 Class B.

**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.
1 PRODUCT OVERVIEW

Thank you for choosing the nanoSHARC audio processor board. The nanoSHARC is a tiny yet extremely powerful and flexible digital audio processing circuit board intended for OEM applications and advanced DIY enthusiasts. It includes an onboard optical digital input and asynchronous USB audio input. Audio output and additional audio inputs are to be provided by the system integrator via I2S interfaces.

The audio processing functionality of the board is implemented via the use of a software plugin. For more information on supported plugins, see Software plugins on page 8.

The nanoSHARC is an evolution of miniDSP’s existing audio processing solutions. For those seeking a ready-made “in the box” solution, please see our 2x4 HD, DDRC-24, and other products.

1.1 THE MINIDSP CONCEPT

The miniDSP concept is “one hardware unit + one software plugin = audio processing solution.” This concept leverages the inherent flexibility of DSP (digital signal processing) to deliver a range of flexible but cost-effective solutions.

![Diagram of miniDSP concept]

**Hardware unit**

In this case, the hardware unit is the miniDSP nanoSHARC board together with your own I2S interface hardware.

**Software plugin**

The software plugin is installed on your PC or Mac, and determines the processing that the DSP will perform. It provides a friendly user interface, and downloads instructions into the miniDSP hardware unit that tell it how to process the audio signal.
1.2 SOFTWARE PLUGIN

A dedicated plugin, **nanoSHARC-2x8-96k**, is available for the nanoSHARC board. Its key features are summarized below. In addition, the miniDSP 2x4 HD1 plugin can be used with the nanoSHARC board.

<table>
<thead>
<tr>
<th>Feature</th>
<th>nanoSHARC-2x8-96k</th>
<th>2x4 HD1</th>
</tr>
</thead>
<tbody>
<tr>
<td>Internal sample rate</td>
<td>96 kHz</td>
<td>96 kHz</td>
</tr>
<tr>
<td>Number of input channels</td>
<td>2</td>
<td>2</td>
</tr>
<tr>
<td>Number of output channels</td>
<td>8</td>
<td>4</td>
</tr>
<tr>
<td>Access to I2S input</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Access to I2S output</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Number of FIR taps (total)</td>
<td>3400</td>
<td>4096</td>
</tr>
<tr>
<td>Minimum FIR taps per channel</td>
<td>6</td>
<td>6</td>
</tr>
<tr>
<td>Maximum FIR taps per channel</td>
<td>2048</td>
<td>2048</td>
</tr>
<tr>
<td>PEQ bands per input channel</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>PEQ bands per output channel</td>
<td>10</td>
<td>10</td>
</tr>
<tr>
<td>Compressor block</td>
<td>✓</td>
<td>✓</td>
</tr>
<tr>
<td>Max per-channel time delay</td>
<td>35 ms</td>
<td>80 ms</td>
</tr>
</tbody>
</table>

1. Requires suitable hardware/interface circuitry.

1.3 ORDERING PLUGINS

Plugins must be ordered in addition to the hardware:

- **nanoSHARC 2x8 96k plugin**
- **2x4 HD1 plugin**
2 BOARD LAYOUT AND CONNECTIVITY

This section describes the board and its interfaces.

2.1 BOARD LAYOUT

This diagram shows the layout of the nanoSHARC board.

2.2 DC POWER

Provide DC power between pins 9 and 10 (ground) and pins 11 and 12 (12 V DC) of header J2. The supply voltage must be 12V DC, or the board may malfunction or be damaged. The power supply must be capable of supplying at least 300 mA at 12V DC.

2.3 USB CONNECTIVITY

Connect the mini USB port (mini type B) to a computer to configure the board. The USB port also provides asynchronous USB Audio (Class 2) streaming at all standard sample rates from 44.1 to 192 kHz. USB Audio input is selected from within the plugin or via remote control.

2.4 OPTICAL DIGITAL AUDIO INPUT

Connect the optical port to a digital source. Thanks to the onboard ASRC (asynchronous sample rate converter), sample rates from 20 to 216 kHz are accepted. Optical digital audio input is selected from within the plugin or via remote control.
2.5 I2S Expansion Header Pinouts

Headers J2 and J5 are provided for connection of I/O circuitry via I2S. The pinouts are shown in Table 2 and Table 3. Note that I2C_SCL and I2C_SDA are intended for miniDSP use only – they are not documented or supported for other use.

<table>
<thead>
<tr>
<th>Pin</th>
<th>Function</th>
<th>Pin</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>I2S_LRCLK</td>
<td>2</td>
<td>I2S_BCLK</td>
</tr>
<tr>
<td>3</td>
<td>GND</td>
<td>4</td>
<td>MCLK</td>
</tr>
<tr>
<td>5</td>
<td>I2S_OUT0</td>
<td>6</td>
<td>I2S_OUT1</td>
</tr>
<tr>
<td>7</td>
<td>I2C_SCL</td>
<td>8</td>
<td>I2C_SDA</td>
</tr>
<tr>
<td>9</td>
<td>GND</td>
<td>10</td>
<td>GND</td>
</tr>
<tr>
<td>11</td>
<td>+12V</td>
<td>12</td>
<td>+12V</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Pin</th>
<th>Function</th>
<th>Pin</th>
<th>Function</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>I2S_LRCLK</td>
<td>2</td>
<td>I2S_BCLK</td>
</tr>
<tr>
<td>3</td>
<td>GND</td>
<td>4</td>
<td>MCLK</td>
</tr>
<tr>
<td>5</td>
<td>I2S_OUT0</td>
<td>6</td>
<td>I2S_OUT1</td>
</tr>
<tr>
<td>7</td>
<td>I2S_OUT2</td>
<td>6</td>
<td>I2S_OUT3</td>
</tr>
<tr>
<td>9</td>
<td>GND</td>
<td>10</td>
<td>GND</td>
</tr>
<tr>
<td>11</td>
<td>I2S_IN0</td>
<td>12</td>
<td>I2S_IN1*</td>
</tr>
<tr>
<td>13</td>
<td>I2S_IN2*</td>
<td>14</td>
<td>I2S_IN3*</td>
</tr>
</tbody>
</table>

* Not accessed by current plugins

2.6 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 nanoSHARC uses internally. This clock is always provided as an output by the nanoSHARC, 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 (Fs) of the audio signal.
- **BCLK** The bit clock (also known as shift clock or system clock). This is always equal to 64 x Fs.
Table 4 summarizes the relation between the clocks for the 96 kHz plugin.

**Table 4. I2S clock ratios**

<table>
<thead>
<tr>
<th>Plugin sample rate (LRCLK)</th>
<th>Master clock (MCLK)</th>
<th>Bit clock (BCLK)</th>
<th>MCLK/LRCLK</th>
<th>BCLK/LRCLK</th>
</tr>
</thead>
<tbody>
<tr>
<td>96 kHz</td>
<td>24.576 MHz</td>
<td>6.144 MHz</td>
<td>256</td>
<td>64</td>
</tr>
</tbody>
</table>

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

![Figure 31. PS Mode—16 Bits to 24 Bits per Channel](image)

The nanoSHARC board has four I2S input data lines and four I2S output data lines, each carrying two channels of audio. Note, however, that the input data lines for channels 3 through 8 (I2S_IN1, I2S_IN2, and I2S_IN3) are not used by current plugins.

### 2.7 I2S DETAILS

The nanoSHARC acts in **master** mode with respect to I2S clocking.

That is, the nanoSHARC provides the I2S clocks and the connected devices are expected to transmit and receive data using those clocks. The clocks will match the sample rate of the plugin loaded onto the nanoSHARC. (See the [miniSHARC User Manual](http://www.minids.com) for more information about I2S master and slave modes). A typical connection scenario is shown in Figure 1. Note that:

1. The connected devices may or may not use MCLK. This is dependent on the specifics of the devices.
2. The digital optical and asynchronous USB audio inputs use an onboard asynchronous sample rate convertor to convert the incoming sample rate to the nanoSHARC’s clock domain.
3. External digital input circuitry will need to use an asynchronous sample rate convertor to convert the incoming sample rate to the nanoSHARC’s clock domain.
2.8 ADDITIONAL I2S USAGE NOTES

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. For off-the-shelf completed units with similar capabilities, please see our 2x4 HD, DDRC-24, and other products.

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 BLCK).
- Observe correct grounding and shielding, and keep analog and digital grounds separated.
- Ensure that the clock ratios (as listed in Table 4) 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 SOFTWARE INSTALLATION

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. To access the download, you will need to be logged into the website with the account you created when purchasing.

Note: this installation procedure describes the nanoSHARC-2x8-96k plugin. If you intend to use the 2x4 HD1 plugin with the nanoSHARC board, please follow the procedure in the 2x4 HD User Manual.

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 “nanoSHARC” at the link below:

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

The User Downloads link is visible from the dropdown menu at the top right of the website page:

Navigate to the OpenDRC/miniSHARC/nanoSHARC plug-ins section and download the zip file under the heading nanoSHARC 2x8 96k. Unzip the downloaded file (on Windows, right-click and select “Extract All...”; on Mac, double-click).

Note: the Adobe Air framework may need a network connection the first time the plugin is used. If the plugin does not start properly, see Troubleshooting.

3.1 WINDOWS

3.1.1 Possible Windows installation issues

The miniDSP software requires that a number of other frameworks be installed for it to work. For Windows 7 and later, these packages should be installed automatically. For earlier versions of Windows, please download and install the following frameworks before attempting to install any miniDSP software. You can also manually install these if you receive an error message that required software is missing.

- 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).
3.1.2 Plugin installation

1. Navigate to the Plugins folder of the software download and then to the Windows folder.
2. Double-click on the nanoSHARC_2x8_96k.exe installer program to run it. We recommend that you accept the default installation settings.
3. 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.”

3.1.3 USB Driver installation

4. Connect the nanoSHARC to the computer using the supplied USB cable, and power it on.
5. Navigate to the WinDrivers folder of the software download and double-click on the appropriate installer:
   - miniDSP_UAC2_v2.29.3_ForWinXP_Vista.exe for Windows XP and Vista
   - miniDSP_UAC2_v4.47.0_ForWin7_8_10.exe for Windows 7, 8, and 10

(The version number embedded in the filename may be different.)
We recommend accepting the default installation location. Once the driver installation completes, click the Finish button.

⚠️ The Windows PC will not be able to communicate properly with the nanoSHARC if you did not have it connected by USB and powered on when you installed the driver. If that is the case, you will need to uninstall the driver, connect the nanoSHARC, power it on, and run the installer again.

**Note:** the first time you run the plugin, you may see a warning from Windows Firewall as shown below. If so, ensure that “Private networks...” is checked and “Public networks...” is not checked. Then click on “Allow access.”
3.2  Mac

3.2.1  Possible Mac installation issues

If double-clicking on an installer brings up a message that the installer cannot run, use this alternate method (note that the name of the plugin will be `nanoSHARC-2x8-96k.pkg`, not `MiniDSP_Plugin.pkg` as shown in the screenshots):

1. Right-click on the installer (or click while holding the Control key).
2. On the menu that pops up, move the mouse over the “Open With” item and then click on “Installer (default).”
3. The following window will appear. Click on “Open.”

![Screenshot showing right-click options]

3.2.2  Plugin installation

6. Navigate to the Plugins folder of the software download and then to the Mac folder.
7. The installer program is named `nanoSHARC-2x8-96k.pkg`. To run it, double-click on it, or right-click and open as described above. We recommend that you accept the default installation settings.
8. To run the nanoSHARC-2x8-96k plugin, locate it 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.

Note: No USB driver installation is required for Mac OS X.
4 CONFIGURING THE NANOSSHARC

The nanoSSHARC can be configured with either the nanoSSHARC-2x8-96k or 2x4 HD1 plugin / user interface program. This section describes the nanoSSHARC-2x8-96k plugin. If you intend to use the 2x4 HD1 plugin with the nanoSSHARC board, please refer to the 2x4 HD User Manual.

Once the nanoSSHARC is fully configured, the computer is no longer required, as source and preset selection can be done with a remote control. If desired, however, the plugin can remain connected during use for real-time (“live”) control of all audio processing.

This screenshot shows the nanoSSHARC-2x8-96k plugin with the key areas highlighted:

⚠️ During initial configuration of the processor, it is strongly recommended that any connected amplification be powered off.

⚠️ If switching plugins, be sure to use the Refresh DSP Program option from the Restore menu.
4.1 Synchronizing with the Processor

Communication with the processor takes place over a USB connection. Ensure that the computer is connected to the nanoSHARC by a USB 2.0 port. Then click on the Connect button:

![Connect](connect.png)

The first time you connect, or if you have made any changes to any data in the user interface, the following dialog box will appear:

![Dialog Box](dialog_box.png)

The options are:

**Synchronize Config**

Download the currently selected configuration into the corresponding configuration preset of the processor. After downloading the configuration data, the plugin is in online mode and any changes to processing parameters will be downloaded immediately in real time. That is, the user interface is now “live.”

**Synchronize and Upgrade**

This is similar to Synchronize Config, but also upgrades the internal data of the processor. This option may appear after downloading and installing an updated version of the plugin.

**Restore Config**

Restore the data in the currently selected configuration to the factory defaults. When using this option, any connected equipment should be muted or powered off until you have set the configuration to a working state. Note that the configuration data will be lost, so if needed, ensure that you have saved the configuration to a file prior to using this option.

**Help**

This option brings up a help screen explaining the options.

**Cancel**

This option cancels the attempt to connect to the processor. The plugin will remain in offline mode.
4.2 **MASTER CONTROL**

Once the plugin is online, the items in the Master Control area are active. The **Mute** button disables all audio output:

The **Master Volume** display shows the current volume setting. The master volume can be set directly by clicking here and typing a new value. It can also be set with the remote control:

The **IP Address** and **Auto** fields are for use with networked control of the plugin using the miniDSP **Wi-DG Wifi/Ethernet to USB bridge**. See the **Wi-DG User Manual** for details.

4.3 **CONFIGURATION/PRESET SELECTION**

The set of data that controls the back-end processing is called a *configuration*. This includes crossovers, parametric EQ and the routing matrix. It does not include the master volume or mute status.

Four configurations are stored onboard. The currently selected preset is indicated by a dark background:

To switch to a different preset, just click on the desired button:
4.4 Inputs & Routing Tab

The Inputs & Routing tab displays two input channel control strips and the 2x8 routing matrix.

4.4.1 Input selection

When the plugin is connected to the nanoSHARC, the currently selected input appears next to the “Inputs” label. Click on the current input name to drop down a selector menu, from which you can select a different input. (You can also select the input with a remote control.)
4.4.2 Input channel strips

Channel label

Each input channel has a customizable label, which is shown at the top of the channel strip. This label also appears on the Routing matrix. To change the label, click on it, type a new label (up to eight characters), and press the Return key.

Level meter, Current RMS level

Displays the current signal level in real time. (The plugin must be in online mode to display signal levels.)

Gain adjustment

The gain of each channel can be adjusted by moving the Gain Adjustment slider, or by typing the desired gain into the Current Gain text box. The maximum gain setting is 12 dB, and the minimum gain setting is –72 dB. (0 dB, the default, is unity gain or no change in level.)

PEQ settings

Click on this button to open the parametric EQ settings window for that channel. There are ten parametric EQ filters on each input channel. For more details, see Parametric EQ on page 23.

Mute

Press this button to mute that input channel. The button color and label changes to show that the channel is muted.
4.4.3 Routing

The **Routing** matrix mixer is used to direct input channels (along the left) to output channels (along the top). To turn on routing for a crosspoint, click on that crosspoint. The display will change from “Off” to “On”.

The matrix mixer makes the nanoSHARC adaptable to many audio processing situations. The default routing is suitable for a stereo 4-way loudspeaker:

<table>
<thead>
<tr>
<th>Input1</th>
<th>Output1</th>
<th>Output2</th>
<th>Output3</th>
<th>Output4</th>
</tr>
</thead>
<tbody>
<tr>
<td>On</td>
<td>On</td>
<td>On</td>
<td>On</td>
<td>On</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Input2</th>
<th>Output5</th>
<th>Output6</th>
<th>Output7</th>
<th>Output8</th>
</tr>
</thead>
<tbody>
<tr>
<td>Off</td>
<td>Off</td>
<td>Off</td>
<td>Off</td>
<td>Off</td>
</tr>
</tbody>
</table>

However, any other routing from inputs to outputs can be used. Changing the names of the input and output channels helps to visualize the system configuration. This example is for a stereo two-way loudspeaker and four subwoofers driven by the sum of left and right channels:

<table>
<thead>
<tr>
<th>Left</th>
<th>Tweet L</th>
<th>Woof L</th>
<th>Tweet R</th>
<th>Woof R</th>
</tr>
</thead>
<tbody>
<tr>
<td>On</td>
<td>On</td>
<td>Off</td>
<td>Off</td>
<td>Off</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Right</th>
<th>Tweet L</th>
<th>Woof L</th>
<th>Tweet R</th>
<th>Woof R</th>
</tr>
</thead>
<tbody>
<tr>
<td>Off</td>
<td>Off</td>
<td>On</td>
<td>On</td>
<td>On</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Left</th>
<th>Sub 1</th>
<th>Sub 2</th>
<th>Sub 3</th>
<th>Sub 4</th>
</tr>
</thead>
<tbody>
<tr>
<td>On</td>
<td>On</td>
<td>On</td>
<td>On</td>
<td>On</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Right</th>
<th>Sub 1</th>
<th>Sub 2</th>
<th>Sub 3</th>
<th>Sub 4</th>
</tr>
</thead>
<tbody>
<tr>
<td>On</td>
<td>On</td>
<td>On</td>
<td>On</td>
<td>On</td>
</tr>
</tbody>
</table>
4.5 Output Tab

The Output tab displays a row of eight output channel control strips. All output channels are identical.

4.5.1 Channel strip layout

Each output channel has a complete "strip" of controls.

4.5.2 Channel label

Each output channel has a customizable label, which is shown at the top of the channel strip. This label also appears on the Routing matrix. To change the label, click on it, type a new label (up to eight characters), and press the Return key.

4.5.3 Level meter and gain control

Level meter, Current RMS level

Displays the current signal level in real time. (The plugin must be in online mode to display signal levels.)

Gain adjustment

The gain of each channel can be adjusted by moving the Gain Adjustment slider, or by typing the desired gain into the Current Gain text box. The maximum gain setting is 12 dB, and the minimum gain setting is –72 dB. (0 dB, the default, is unity gain or no change in level.)
4.5.4  Parametric EQ

Parametric equalization (PEQ) is a flexible type of equalization filter. It can be used to correct for errors in loudspeaker output, to compensate for acoustic room effects, and to tailor the overall system response for best sound. Click on the PEQ button to open the parametric equalizer settings window:

There are ten parametric EQ filters on each input and output channel. The window displays a frequency response graph showing the combined response of all enabled parametric filters on that channel. For example, the screenshot above shows a response curve created with a low-shelf boost filter at 100 Hz, a dip at 500 Hz, and a high-shelf cut filter at 5000 Hz.

Hovering the mouse over the curve brings up an overlay showing the frequency and the gain at that frequency.

Each channel can be linked to one other channel. When a channel is linked to another, the PEQ settings of that channel are mirrored to the other. Typically, the corresponding drivers on the left and right channels are linked: left and right tweeter, left and right woofer, and so on. To link a channel, select the other channel from the drop-down menu at the top left of the PEQ screen, and click the Link checkbox.
EQ band selection

Click on the tabs **EQ1, EQ2**, etc. to display the parameters for that filter.

Basic/Advanced

By default, each filter is in basic mode, and shows the controls described below. Advanced mode enables custom biquad programming for almost infinite flexibility in filter implementation. This is described in **Custom biquad programming** on page 29.

Filter type

Selects the type of filter:

- **PEAK** Create a dip or a peak in the frequency response.
- **LOW_SHELF** Reduce or increase part of the frequency spectrum below a given frequency.
- **HIGH_SHELF** Reduce or increase part of the frequency spectrum above a given frequency.
- **ALL_PASS** Create a phase shift across the frequency band. This can be useful for correcting phase issues and for simulating analog crossovers.

Frequency

For the **PEAK** filter type, this is the center frequency of the peak or dip. For the **HIGH_SHELF** and **LOW_SHELF** filter types, this is the frequency at which the gain is half of the set value. For the **ALL_PASS** filter type, this is the center frequency of the phase shift.

Gain

For the **PEAK** filter type, this is the gain in dB at the center frequency. For the **HIGH_SHELF** and **LOW_SHELF** filter types, this is the gain in dB reached at high or low frequencies respectively. A filter has no effect if its gain is set to 0 dB. Gain can be adjusted in increments of 0.1 dB up to +/-16 dB. This item is not present for the **ALL_PASS** filter type.

Q

**Q** controls the “sharpness” of the filter. For the **PEAK** filter type, lower Q gives a broader peak or dip, while higher Q gives a narrower peak or dip. For the **HIGH_SHELF** and **LOW_SHELF** filter types, Q controls how quickly the filter transitions from no gain to maximum gain. For the **ALL_PASS** filter type, higher Q gives a steeper phase transition.

Bypass

The **Bypass** button enables or disables a filter. The filter is enabled if the button says “BYPASS” and disabled if the button says “BYPASSED” (see screenshot below). Note that this button only enables and bypasses a single filter; all other filters must be bypassed or enabled individually.
4.5.5 Crossover

Each output channel has independent high pass and low pass filters. Click on the Xover button to open the crossover settings window:

Crossovers “split” the frequency band to send to different drivers. In a two-way loudspeaker, a low pass filter is used to remove high frequencies from the signal sent to the woofer, and a high pass filter is used to remove low frequencies from the signal sent to the tweeter. In a three-way speaker, the midrange driver will use both the high pass and low pass filters. Crossover filters can also be used to limit low frequency content delivered to a speaker or subwoofer, to help protect it from over-excursion.

Unlike conventional analog crossovers, the flexibility of DSP allows a completely arbitrary mix of different filter slopes and types. Filters can be set at any frequency, or disabled completely. This allows maximum flexibility in matching your crossover to the acoustic characteristics of the loudspeaker drivers.

The current channel is displayed in orange, with the others displayed in grey. Hovering the mouse over the curve brings up an overlay showing the frequency and the attenuation at that frequency.
Basic/Advanced

By default, the crossover is in basic mode, and shows the controls described below. Advanced mode enables custom biquad programming for almost infinite flexibility in crossover filter implementation. This is described in Custom biquad programming on page 29.

Cutoff Frequency

Sets the nominal cutoff frequency of the crossover. In actual fact, the crossover has a more or less gradual transition from “full on” to “full off,” as determined by the filter slope.

Filter type

Selects the type and slope of the filter. The steeper the slope, the more quickly frequencies above or below the cutoff frequency are attenuated. There are three types of filter:

**Butterworth (BW)**

Available in 6, 12, 18, 24, 30, 36, 42, and 48 dB/octave, Butterworth crossover filters are 3 dB down at the cutoff frequency.

**Linkwitz-Riley (LR)**

Available in 12, 24, 36, and 48 dB/octave, Linkwitz-Riley crossover filters are 6 dB down at the cutoff frequency.

**Bessel**

Available in 12 dB/octave only, a Bessel filter gives a more gradual roll-off through the crossover region.

Bypass

Clicking on the Bypass button disables or enables that high pass or low pass filter.

Each channel can be linked to one other channel. When a channel is linked to another, the crossover settings of that channel are mirrored to the other. Typically, the corresponding drivers on the left and right channels are linked: left and right tweeter, left and right woofer, and so on. To link a channel, select the other channel from the drop-down menu at the top left of the Xover screen, and click the Link checkbox.
4.5.6 Compressor

The compressor reduces the gain of an output channel when the audio signal reaches a certain level as specified by the **Threshold** parameter. The gain of the channel will be progressively reduced as the signal increases above the threshold, according to the **Ratio** parameter. This can be used to limit the power delivered to speakers and thus reduce the risk of damage from overdriving.

This screenshot shows an example Compressor setting:

![Compressor Setting](image)

(Note that the compressor algorithm is bypassed by default, so click on the **Bypass** button to see the curve as shown here.)

In this example, the threshold is set to -20 dB, so the compressor will activate when the signal on that channel reaches -20 dB (relative to full output). The ratio is set to 2, so if the input signal level to the compressor then increases by 10 dB, the output level will increase by only 5 dB. If the input signal level to the compressor is at full scale (0 dB), then the output level will be limited to -10 dB.

Two additional parameters control the action of the compressor: the attack time and the release time. These two parameters govern how quickly the compressor activates when the signal level exceeds the threshold, and how quickly it deactivates when the signal level reduces. The optimum settings may need to be tuned by ear. For more information, see the Wikipedia article **Dynamic range compression**.)
4.5.7  FIR

Each output channel has an FIR filter bank with a variable number of taps. Click on the FIR button to open the FIR filter settings window:

FIR filtering is a powerful feature that allows very complex filters to be constructed (with the aid of suitable design software). These filters can correct for amplitude only ("linear phase filters"), phase only, or a combination or both. FIR filtering is described more in FIR filtering and design, starting on page 32.

4.5.8  Invert and mute

Each channel can be inverted in polarity, and individually muted. When either of these options is selected, the visual indicator on the button is "lit":

4.5.9  Time delay

A delay of up to 35 ms can be applied to each output channel. To set the delay, click in the delay entry box for a channel. The delay value can be entered numerically, and the up and down arrows can be used to change the delay in small (0.01 ms) increments. The maximum time delay of 35 ms corresponds to a distance of approximately 12 meters (about 40 feet).

The time delay corresponds to a distance. This distance is shown in centimeters below the entry box.
4.6 CUSTOM Biquad PROGRAMMING

Custom biquad programming is available in the PEQ and Crossover blocks. Its purpose is to allow you to directly provide the low-level parameters aka biquad coefficients that control the digital filters of the processor, thus providing an almost infinite degree of flexibility.

For example, you can create hybrid crossovers with staggered cutoff frequencies, create parametric EQ filters beyond those provided in the easy-to-use “basic” interface, implement a Linkwitz transform, or mix crossover and EQ biquads in the same block.

4.6.1 What’s a “biquad?”

A biquad is the basic unit of processing that is used to create digital filters. It can be described either with an equation or with a signal flow diagram, as shown here:

\[ H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}} \]

A single biquad like this can perform a great many functions, including all of the functions of a single parametric EQ filter, one 6 or 12 dB/octave high pass or low pass filter, and more. Biquads are combined in series (cascaded) to create more complex filters. The function that each biquad performs is determined by just five numbers: \(a_1, a_2, b_0, b_1,\) and \(b_2\). These numbers are called the coefficients.

4.6.2 Using custom biquad programming

Each crossover block and PEQ filter has a selector that switches it to advanced mode:

In advanced mode, the biquad coefficients can be pasted directly into the user interface. These coefficients must be calculated using a design program – see Biquad design software below for suggestions.
**Parametric EQ advanced mode**

In the parametric EQ blocks, advanced mode allows each individual filter to be specified by its biquad coefficients. After pasting in the coefficients, click on the **Process** button for them to take effect.

![Parametric EQ blocks with advanced mode](image)

**Parametric EQ file import (REW integration)**

Multiple biquads in the parametric EQ block can be set at once by importing a coefficient file. This file can be generated by Room EQ Wizard (REW) or by other programs. The design program must be set for a **96 kHz** sample rate. The number of filters is limited to a maximum of ten.

This example illustrates the correct file format:

```
biquad1,
b0=0.998191200483864,
b1=1.9950521500467384,
b2=0.996920046761057,
a1=-1.9950521500467384,
a2=-0.9951112472449212,
biquad2,
b0=0.999640139948623,
b1=-1.9981670485581222,
b2=0.9985489719847982,
a1=1.9981670485581222,
a2=-0.998189119334211,
biquad3,
...
biquad4,
...
biquad10,
b0=1.0010192374642126,
b1=-1.9950555192569264,
b2=0.9940580112181501,
a1=1.995060938714333,
a2=-0.9950718292249559
```

Note that the last line must not have a comma at the end. If the file has less than ten biquads, then only that number of biquads will be imported. For example, if importing a file with six biquads, the first six filters will be set, and the last four will not be changed. (Note: if the last line ends with a comma, that counts as an extra biquad.)

If the file contains more than ten biquads, then an error will be reported and no filters will be changed.
Crossover advanced mode

The Crossover blocks have eight biquads for each output channel. In Advanced mode, all eight biquads need to be specified. After pasting in the coefficients, click on the Process button for them to take effect.

4.6.3 Biquad design software

Following are programs that can be used to design your biquad coefficients.

4.6.3.1 Biquad calculation spreadsheet

The community-developed biquad calculation spreadsheet allows many filter types to be calculated, including notch filters, Linkwitz transforms, and filters with arbitrary Q-factor. Access this spreadsheet here (requires Microsoft Excel):


4.6.3.2 Room EQ Wizard (REW)

Room EQ Wizard is a free acoustic measurement and analysis tool, available for Windows, Mac and Linux platforms. It includes the ability to automatically generate a bank of parametric EQ biquads based on a measurement. These coefficients can be saved to a file from REW and loaded directly into a PEQ bank in a miniDSP plugin. Room EQ Wizard can be downloaded here:

- [http://www.roomeqwizard.com/#downloads](http://www.roomeqwizard.com/#downloads)

For guidance on using this feature, please refer to the app note Auto EQ with REW.
4.7 FIR FILTERING AND DESIGN

FIR filtering is a powerful and advanced feature of the nanoSHARC. It allows construction of complex arbitrary equalization and crossover filters with independent control of amplitude and phase. The parameters of each FIR filter are set in the FIR settings window:

Browse Opens a file browser to select a file containing FIR filter coefficients. (See Filter file format below).

Unload FIR Deletes the currently loaded filter from the display and from the DSP memory.

Send to DSP Writes the currently loaded filter into the DSP memory.

BYPASS Disables the FIR filter. The filter is disabled when the button is "lit."

File Mode / Manual Mode

In File Mode, the window displays the Browse and Unload FIR buttons as shown above. In Manual Mode, the display changes to allow direct text entry of the FIR filter coefficients, as shown below. The coefficients can be pasted into the window from a text editor.
4.7.1 FIR filtering overview

FIR ("finite impulse response") filtering differs from the IIR ("infinite impulse response") filters used in the PEQ and crossover blocks. Technically speaking, IIR filters are recursive, meaning that each output value is partially calculated from earlier output values as well as from input values. An FIR filter is specified by a large array of numbers, whereas an IIR filter requires only a fairly small of values to be specified. These numbers are conventionally referred to as "taps."

The nanoSHARC can compute a total of 3400 taps at 96 kHz. These taps can be distributed as you wish across the four output channels, with the limitation that each output channel must have 6 or more taps and can have no more than 2048 taps. The decision on how many taps to allocate to each channel is up to you, and should be determined after working with an FIR filter design program (see below). The number of taps is set in the lower right corner (click on the text entry box and type the desired number of taps, then press Tab or Return):

4.7.2 FIR filter design software

The filter coefficients must be created with the aid of filter design software. miniDSP does not provide any such software, instead referring you to the many freeware and commercial software packages available for this purpose. Please see the FIR filter tools page on our website.

4.7.3 Filter file format

The filter coefficient file loaded in File Mode uses IEEE 754 single-precision binary floating-point format. The number of entries in the file must not exceed the allocated number of taps.

In Manual Mode, the coefficients must be plain text in this format:

$$
b_0 = 1,
b_1 = -1,
b_2 = 0.5,
b_3 = -0.5,
b_4 = 0.2,
b_5 = 1,
$$

Taps Used : 1024
Taps Available : 2036
4.7.4  Loading filter coefficients

In File Mode:

9.  Click Browse, navigate to the file containing the filter coefficients, and open it. A dialog will appear confirming the number of coefficients loaded.
10. Confirm that the response curve is as you expect.
11. Press Send to DSP. This will write the coefficients into the DSP's memory.
12. To clear the filter coefficients, click Unload FIR and then Send to DSP.

In Manual Mode:

13. Cut and paste the coefficients from the text output of the design program.
14. Press the Process button.
15. Confirm that the frequency response graph is as you expect.
16. Press Send to DSP. This will write the coefficients into the DSP's memory.
17. To clear the filter coefficients, click Clear Taps and then Send to DSP.

If, after selecting a filter file or setting coefficients, the frequency response graph does not change as expected, make sure that the Bypass button is turned off.

If you clear the filter taps, make sure that you also bypass the filter, otherwise there will be no audio through that channel.
4.8 WORKING WITH CONFIGURATIONS

The data that controls the audio processing is called a *configuration*. The processor stores four configuration presets in its internal memory, which can be selected from the plugin or via remote control.

4.8.1 Online and offline mode

Initially, the plugin is in *offline* mode. When the **Connect&Synchronize** button is used, the plugin downloads configuration data into the processor and goes into *online* mode. Changes made in the plugin user interface therefore fall into two categories:

**The plugin is in online mode**

The plugin user interface is “live” – that is, any changes made to the audio processing parameters in the user interface are immediately downloaded to the processor. The effect of these changes will thus be audible as the changes are made.

**The plugin is in offline mode**

Changes made to audio processing parameters in the plugin user interface will be made locally only. The next time the plugin is synchronized to the processor, the parameters will be downloaded to the processor (as long as the **Synchronize Config** button is selected).

The configuration contained in the miniDSP hardware unit cannot be uploaded back to the computer. Therefore, you **must** save your configuration to a file if you wish to recover from any changes you make while in offline mode.

4.8.2 Selecting a configuration

The active configuration is selected by one of the four Configuration Selection buttons:

![Configuration Selection: Config 1 Config 2 Config 3 Config 4]

To switch to a different configuration, click on a different button. There are two cases:

**The plugin is in online mode**

Audio processing will switch to the parameters contained in the selected configuration. If, however, parameters of the newly selected configuration have been changed since the last that particular configuration was synchronized to the processor, then a dialog will appear asking you if you want to synchronize the configuration.

**The plugin is in offline mode**

The user interface will update to show the parameters of the newly selected configuration. If this configuration is changed in the user interface, it will be downloaded to the processor the next time it is synchronized.
4.8.3 Saving and loading configurations

Configurations can be saved to and loaded from files. Each configuration is stored in a separate file. It is very strongly recommended that each configuration programmed into the processor be saved to a file, to ensure that the configuration is not lost if the processor is inadvertently reset.

To save the currently selected configuration to a file, drop down the File menu, then select Save and then Save current configuration. In the file box, select a location and name of the file, and save it.

To load a configuration, first select the configuration preset that you wish to load into. Then drop down the File menu, select Load, and then Load configuration to current slot.

If the plugin is in online mode, the new configuration data will be downloaded to the processor immediately. If the plugin is in offline mode, the configuration will be loaded into the user interface only, and will be downloaded to the processor the next time it is synchronized.

To copy a configuration from one preset to another, save the configuration to a file, then select a different configuration preset and load the file.

4.8.4 Restoring to defaults

Configurations can be reset to the factory defaults from the Restore menu. There are two options:

Factory Default

Reset all four configuration presets to the factory default settings.

Current Configuration Only

Reset only the currently selected configuration preset to the factory default settings.

If the plugin is in online mode, the configuration data on the processor (all or just one configuration, as selected) will also be reset to factory defaults. Otherwise, the reset will take place in the user interface only, and the new configuration data will be downloaded to the processor next time it is synchronized.
4.9 Keyboard Shortcuts

The nanoSHARC-2x8-96k plugin supports the use of the keyboard for many operations.

Tab

The Tab key moves the focus from the current user interface element to the next. A blue-grey surrounding box usually indicates the user interface element with the focus. Shift-Tab moves the focus in the opposite direction.

Up/down arrows

The up/down arrow keys (and in some cases, the left/right arrow keys) adjust the value of many parameters, if they have the focus:

- Gain adjustment
- Crossover frequency and filter type
- PEQ filter frequency, gain, and Q

Space

The Space bar toggles buttons that have two states, such as Bypass, Invert, and Mute, if they have the focus.
5 USB Audio

The miniDSP nanoSHARC accepts stereo PCM audio at sample rates of 44.1, 48, 88.2, 96, 176.4, and 192 kHz on its USB audio input. The same USB connector is used both for streaming audio and configuration.

5.1 MacOS X

Open the program Audio MIDI Setup (in Applications->Utilities). The nanoSHARC will in the list on the left hand side. Clicking on it will show the input and output channels. Sample rate and number of bits (16 or 24) can be selected in the dropdown menus:

To set the nanoSHARC as the default audio output device, right-click and select “Use this device for sound output”.

![Audio MIDI Setup](image)
5.2 Windows

Open the UAC Control Panel from the Windows Start menu. This control panel allows you to view current settings. In addition, it can be used to set buffer size, although we recommend that you leave this setting at the default.

![miniDSP UAC2 Control Panel](image)

If you are having an issue with inadequate output volume over USB playback, check the Volume tab.

To set the nanoSHARC as the default output device, open the Windows Control Panel and navigate to the Audio Devices section. On the Output tab, select the nanoSHARC (currently labeled “miniDSP 2x4n”) and click on the “Set Default” button. Individual audio playback programs may allow the nanoSHARC to be selected independently of the system default and/or to use the miniDSP ASIO Driver.
6 INFRARED REMOTE CONTROL

Once configuration is complete, the computer is not required and can be disconnected. An infrared remote can be used to control volume, mute, preset selection, and input selection. The optional miniDSP remote can be used with the nanoSHARC:

**Source**

Cycles through the input sources.

**1, 2, 3, 4**

Switches to the selected preset.

**[Bell]**

Has no function with the nanoSHARC.

**Vol**

Reduce or increase the volume. Each press changes the volume in 0.5 dB. Holding down a button will accelerate volume change to 3 dB steps.

**Mute**

Mutes and unmutes audio output.

The nanoSHARC can also “learn” the control codes of your current remote if it supports one of the following remote control codes:

- Apple
- NEC
- Sony
- Philips RC6

To initiate the learning process, drop down the IR Remote menu and select **IR learning**. Click on the **Learn** button for an operation, and then press the desired button on the remote control. If the code is accepted, the status will change to show a tick, as shown at right.

To "unlearn" a command, press the **Learn** button and wait for the plugin to time out. Note that you cannot “learn” the miniDSP remote – it will always work, even if you learn another remote’s codes.
## 7 ADDITIONAL INFORMATION

### 7.1 SPECIFICATIONS

<table>
<thead>
<tr>
<th>Specification</th>
<th>Details</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Computer connectivity</strong></td>
<td>Driverless USB 2.0 control interface for Windows and Mac OS X</td>
</tr>
<tr>
<td><strong>USB audio input</strong></td>
<td>XMOS asynchronous USB audio, 44.1 to 192 kHz, USB Audio Class 2 compliant. ASIO driver for Windows, driverless for Mac OS X.</td>
</tr>
<tr>
<td><strong>Digital audio input</strong></td>
<td>TOSLINK optical. A high quality onboard Asynchronous Sample Rate Converter ensures compatibility with most sample rates, from 20−216kHz.</td>
</tr>
<tr>
<td><strong>Audio resolution</strong></td>
<td>24-bit input and output</td>
</tr>
<tr>
<td></td>
<td><strong>nanoSHARC</strong> plugin: 96 kHz internal sample rate</td>
</tr>
<tr>
<td><strong>Audio processing</strong></td>
<td>Analog Devices SHARC 32-bit floating-point processor</td>
</tr>
<tr>
<td></td>
<td>Specific processing functionality depends on loaded plugin.</td>
</tr>
<tr>
<td><strong>Storage/presets</strong></td>
<td>All settings controllable in real time from software user interface.</td>
</tr>
<tr>
<td></td>
<td>Up to 4 presets stored in local flash memory.</td>
</tr>
<tr>
<td><strong>Infrared remote</strong></td>
<td>“Learning remote” capabilities (NEC, Philips, Sony, Apple)</td>
</tr>
<tr>
<td></td>
<td>Controls master volume, mute, digital input selection, preset selection.</td>
</tr>
<tr>
<td><strong>Power supply</strong></td>
<td>12 V DC</td>
</tr>
<tr>
<td><strong>Dimensions (board only, H x W x D)</strong></td>
<td>62 x 65 x 15 mm</td>
</tr>
</tbody>
</table>
7.2 **FIRMWARE UPGRADE**

miniDSP may occasionally provide an upgrade to the nanoSHARC MCU firmware to enable new features. To upgrade the MCU firmware, first download the latest version of the nanoSHARC software package from the User Downloads section of the miniDSP website, then extract it on your computer (on Windows, right-click and select “Extract All...”; on Mac, double-click).

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

7.2.1 **Windows**

1. Connect the nanoSHARC (or Kit) to your computer via USB (if not already connected) and power it on.
2. Navigate to the `XMOS_Firmware\Firmware_Upgrade_Tools\Windows\miniDSPUAC2Dfu` folder of the software download.
3. Double-click on the `miniDSPUAC2Dfu.exe` program to run it:
4. Click on the **Browse** button and select the firmware file from the *XMOS_Firmware* folder of the software download. It will have a name like “nanoSHARC_XMOS_v1.7.bin.” (The version number may be different.)

5. Click on the **Start** button.

6. You will get a progress bar as upgrade proceeds:

![Upgrade Progress](image)

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

![Upgrade Complete](image)

8. Click on **Exit**.

9. That’s it! You’re done. You can now use your nanoSHARC with the new functionality.
7.2.2  macOS / OS X

1. Connect the SHD Series processor to your computer via USB (if not already connected) and power it on.
2. Navigate to the XMOS_Firmware/Firmware_Upgrade_Tools/Mac folder of the software download.
3. Double-click on the DFU Utility.app program to run it:
4. Click on the **Browse** button and select the firmware file from the **XMOS_Firmware** folder of the software download. It will have a name like “nanoSHARC_XMOS_v1.7.bin.” (The version number “v1.7” may change.)

5. Click on the **Start** button.

6. You will get a progress bar as the upgrade proceeds:

   ![Upgrade](image1.png)

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

   ![Upgrade](image2.png)

8. Click on **Exit**.

9. That’s it! You’re done. You can now use your nanoSHARC with the new functionality.
7.3 Troubleshooting

The following table lists the most common causes of issues. If following this table does not provide a solution, see Obtaining Support below.

<table>
<thead>
<tr>
<th>Item#</th>
<th>Symptoms</th>
<th>Troubleshooting recommendation</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Cannot install software</td>
<td>a. Confirm that you downloaded and installed the required frameworks first (see Software Installation).</td>
</tr>
</tbody>
</table>
| 2     | Software running in background but not showing | a. The Adobe Air environment may need a network connection the first time you run a plugin. Close the plugin program, ensure that your computer has a network connection, and restart the plugin.  
| 3     | Cannot connect to the board by USB | a. Reset the processor by power-cycling the unit.  
   b. Make sure the processor is seen in the device manager as a HID device. |
| 4     | Cannot reload a configuration | a. Confirm the file format of your file (.xml).  
   b. Confirm the version of the file. |

7.4 Obtaining Support

1. Check the forums on miniDSP.com to see if this issue has already been raised and a solution provided.

2. Contact miniDSP via the support portal at minidsp.desk.com with:
   a. The specific product you are having an issue with (in this case, nanoSHARC board).
   b. A clear explanation of the symptoms you are seeing.
   c. A description of troubleshooting steps (see Troubleshooting above) performed and your results.

Please note that miniDSP is only able to provide support for the hardware and functions documented in this manual, and only for problems specifically related to the miniDSP hardware and software functions. Any other items, such as designing or debugging your I2S interface circuitry or layout or interfaces to third-party hardware, are specifically excluded from the scope of miniDSP support.