

## 2x4 HD

COMPACT 2-IN 4-OUT  
HIGH-RESOLUTION AUDIO  
PROCESSOR



## 2x4 HD KIT

2-IN 4-OUT HIGH-  
RESOLUTION AUDIO  
PROCESSOR CIRCUIT BOARD



# User Manual



## Revision history

Revision	Description	Date
V1.0	First preliminary release version	15 January 2016
V1.1	Added kit board, installers	31 January 2016
V1.2	Updated installation procedure	22 February 2016
V1.3	Updated installation procedure, added USB Audio	21 March 2016

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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

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
- Mac OS X 10.8 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 2x4 HD 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 miniDSP 2x4 HD or 2x4 HD Kit audio processor. The miniDSP 2x4 HD is an extremely compact yet powerful and flexible digital audio processor for active crossover and other audio processing applications. It features:

- Asynchronous XMOS USB input for streaming audio (up to 192 kHz)
- Stereo digital optical TOSLINK input
- Two analog input channels and four analog output channels
- Parametric EQ on all input and output channels, Butterworth and Linkwitz-Riley crossovers up to 48 dB/octave, and advanced biquad programming for an unlimited range of filters and crossover types
- FIR filtering on every output channel (total 4096 taps)
- Matrix mixer for flexible channel assignment and configuration
- Real-time configuration of all processing parameters
- Volume control by infrared remote
- Four on-board presets, selectable by infrared remote

## 1.1 TYPICAL APPLICATIONS

The miniDSP 2x4 HD has a wide range of applications in home and professional audio. In this example, the 2x4 HD acts as both preamp and two-way crossover:

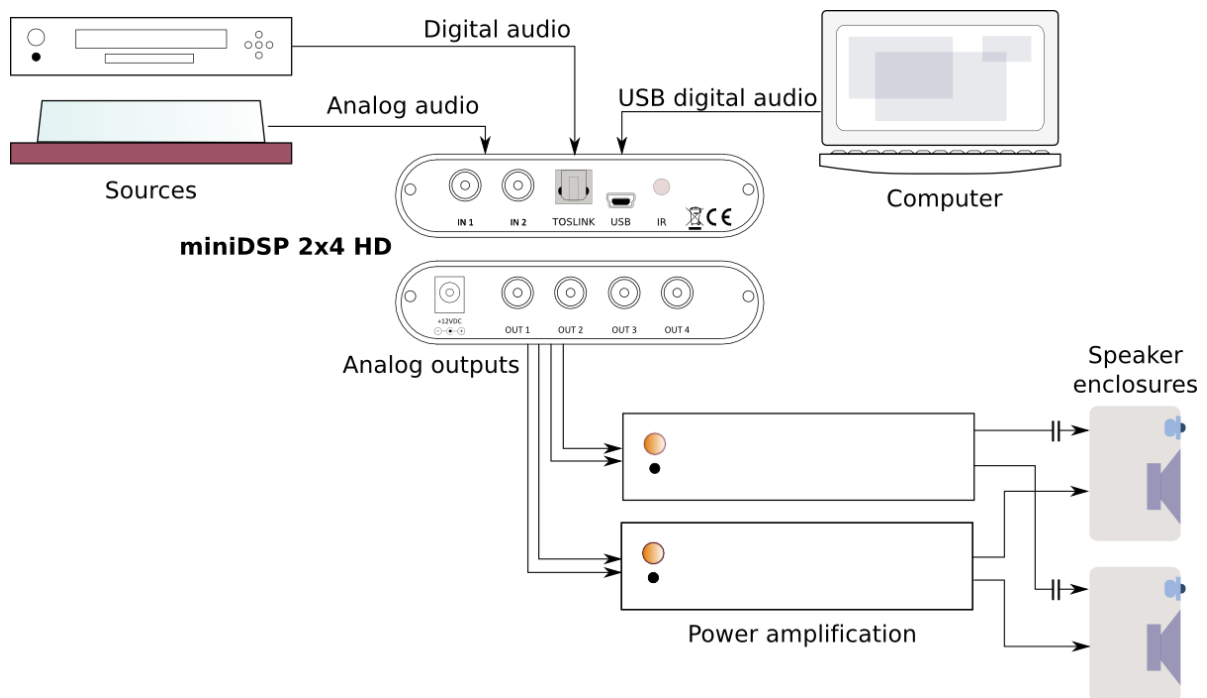


Figure 1 - 2x4 HD as preamp and 2-way crossover

The miniDSP 2x4 HD is a perfect solution for subwoofer integration, providing accurate and flexible integration between the subwoofers and the main speakers, as well as comprehensive EQ and tuning options. In this role, the 2x4 HD is added after the preamp, as shown here:

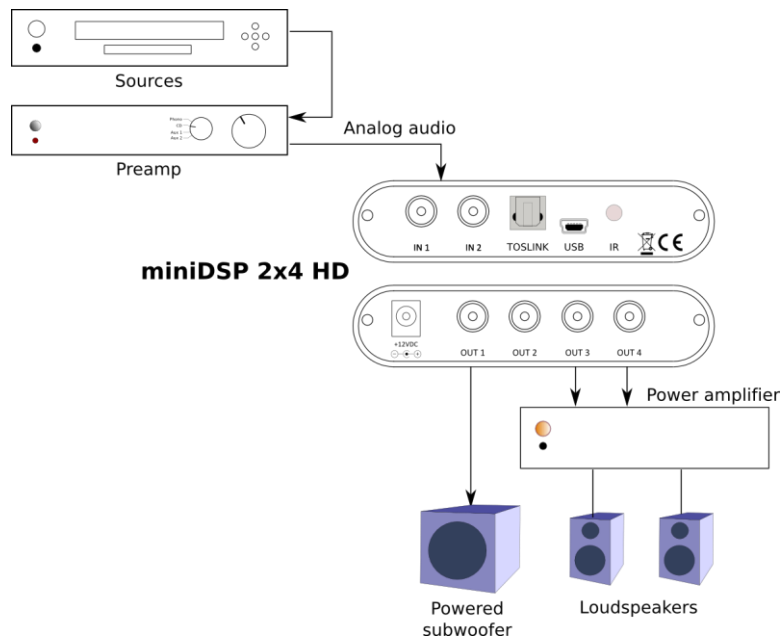


Figure 2 - 2x4 HD for subwoofer integration (2.1-way)

The miniDSP 2x4 HD is a perfect solution for multi-subwoofer management for home theater. Driven from the A/V receiver's subwoofer output, the 2x4 HD can be used to adjust gain and delays individually for each subwoofer, and provide parametric EQ both singly and for all subwoofers:

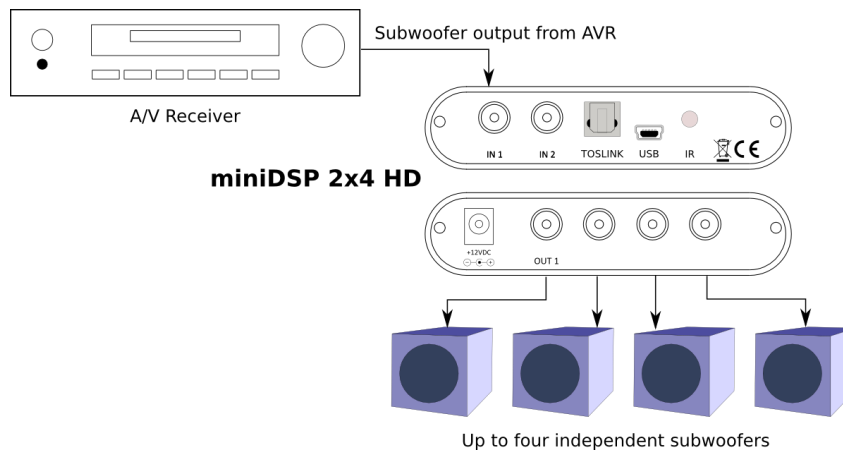


Figure 3 - 2x4 HD for multisub management

The 2x4 HD Kit allows construction of the 2x4 HD circuit board into more sophisticated builds that combine other components such as power amplifier boards.



## 2 THE MINIDSP WORKFLOW

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We strongly recommend taking a methodical approach to your new miniDSP audio processor. Remember that the audio processor is a powerful tool and incorrect settings can potentially cause damage to your system. Please follow the steps below carefully.

### 1. Download and install the software

When your order ships, your ordered plugin (or plugins) will be available from the [User Downloads](#) section of the miniDSP website. Download and install the plugin and USB driver (for Windows), as described in [Software installation](#) on page 11.

### 2. Familiarize yourself with the plugin

Before connecting your computer to the processor, start the plugin and familiarize yourself with its user interface. Click on and explore each of the tabs (**Input&Routing**, **Output**). For more information, see [Configuring the 2x4 HD](#) starting on page 16.

At this time, the plugin is still in *offline* mode, so any changes you make will not be downloaded into the miniDSP hardware. Since all of the processing parameters will be reset in the next step, you can feel free to experiment at this point.

### 3. Reset all parameters

From the **Restore** menu, select the **Factory Default** option. This will reset all processing parameters back to the defaults and ensure that you have a "clean slate" for the next step.

### 4. Perform initial configuration

Use the plugin to set up an initial audio processing configuration as intended for your application. For example, typical things that can be done at this point are to:

- Set the labels of input and output channels
- Mute unused input and output channels
- Set crossover frequencies and slopes
- Set up any essential equalization

### 5. Save configuration

Save your initial configuration to a file. A *configuration* is the set of all audio processing parameters. You should save your configuration to a file on a regular basis, to ensure that you do not lose your work if you inadvertently restore the processor to default settings. For more information on configurations, see [Working with configurations](#) on page 34.

### 6. Make audio connections

With the initial configuration done and now that you are familiar with the various controls, it's time to connect the miniDSP 2x4 HD into your system. *Ensure that all power is turned off when making audio connections.* See [Hardware connectivity](#) on page 13.

## 7. Go into online mode

Apply power to the processor. (See [DC power](#) on page 14 for more information. Leave other equipment turned off at this stage.) Connect the USB cable to your computer, click on the **Connect&Synchronize** button, and select the **Synchronize** option. If all goes well, you are now in *online mode*, and any changes that you now make in the plugin user interface will be immediately sent to the processor.



Before proceeding, click on each Configuration preset button (Config 1, 2, etc.) to ensure that all presets have been synchronized and/or set to defaults.

For more information, see [Synchronizing with the processor](#) on page 17.

## 8. Initial audio check

Power on your connected equipment, first on the input side (e.g. source or preamp), then on the output side (e.g. power amps). Set a low volume (turn the volume on your source down low, or program the infrared remote and reduce the master volume to -50 dB) and start playing music or a pink noise test signal. Gradually increase the volume until you hear audio quietly coming from the speakers. Verify that the plugin is performing the intended function. (For example, if implementing a two-way crossover, confirm that the tweeter is playing high frequencies, and that the woofer is playing low frequencies.)

## 9. Fine-tune your configuration

With your initial setup running, you can now proceed to fine-tune and optimize your system. You will need to perform acoustic measurements and use the configuration screens to adjust processing parameters.



Be sure to save your configuration on a regular basis while working on fine-tuning it. Configurations can be saved to different files, in order to archive different versions, or to enable auditioning of alternative configurations.

## 10. Continue to operate offline

With your miniDSP processor configured, you can continue to operate in *offline mode*—that is, without the plugin software running. The processor “remembers” the configuration last set, and will continue to operate without the computer.

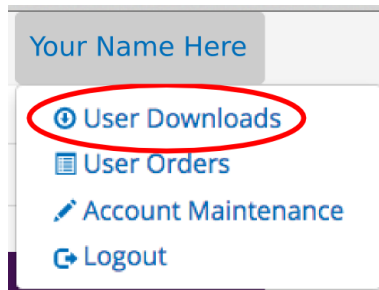
**Note:** while in offline mode, the configuration can still be modified in the plugin interface. These changes will not be downloaded to the processor until it is synchronized again. See [Working with configurations](#) on page 34.

## 3 INSTALLATION AND SETUP

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### 3.1 SOFTWARE INSTALLATION

When your order ships, your ordered plugin (or plugins) will be available from the [User Downloads](#) section of the miniDSP website. You will need to be logged into the 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 **miniDSP Plug-ins** section of User Downloads, then download the zip file under the heading **miniDSP-2x4-HD**. Double-click on it to unzip it. Then follow the installation procedure below according to your computer type.

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](#).

#### 3.1.1 Windows – plugin installation

Prior to installing the plugin, download and install the following programs. You will need to accept the license agreements in order to successfully complete the installation.

- [Microsoft .NET framework](#) (version 3.5 or later)
- Latest version of [Adobe Flash](#)
- Latest version of [Adobe Air](#)

If you haven't updated these recently, you should download and install the latest versions prior to running the miniDSP install program.

To install the miniDSP software, open the **Plugins** folder of the download and double-click on the **MiniDSP\_2x4\_HD.exe** program. We recommend accepting the default installation settings. Once installation is complete, the plugin will automatically start.

#### 3.1.2 Windows – USB driver installation

The driver for the miniDSP 2x4 HD enables the plugin to communicate with it. In addition, it allows USB streaming from the computer to the 2x4 HD and installs a control panel for this purpose.

To install the driver, the miniDSP 2x4 HD **must** be connected to the computer by USB and powered on. Go to the **WinDrivers** folder of the installation download and double-click on the appropriate installer:



- miniDSP\_UAC2\_v2.29.0\_ForWinXP\_Vista.exe for Windows XP and Vista
- miniDSP\_UAC2\_v3.34.0\_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.

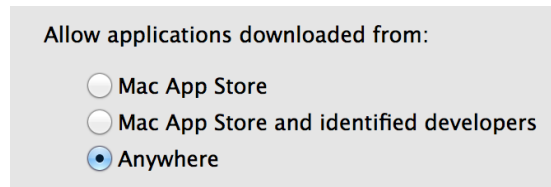


The Windows PC will not be able to communicate properly with the miniDSP 2x4 HD if you did not have the 2x4 HD connected by USB and powered on when you ran the installer. If that is the case, you will need to uninstall the driver, connect the 2x4 HD, power it on, and run the installer again.

### 3.1.3 Mac OS X

On versions of OS X from 10.7 (Lion) and later, you will need to inform the GateKeeper program that it is OK to install and run this software. Go to System Preferences, then click on Security & Privacy and select the General tab:

1. Click on the padlock icon in the lower left corner and enter your password, in order that you can make changes to the settings.
2. Under the text “Allow Applications downloaded from:”, click on “Anywhere.”



Then, download and install the following programs. You will need to accept the license agreements in order to successfully complete the installation:

- Latest version of [Adobe Flash](#)
- Latest version of [Adobe Air](#)

If you haven't updated these recently, you should download and install the latest versions prior to running the miniDSP install program.

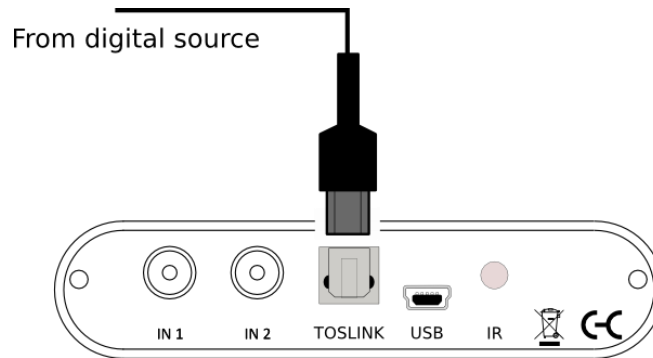
To install the miniDSP software, open the **Plugins** folder of the download, and double-click on the **MiniDSP\_2x4\_HD.dmg** file to open it in a new window. Then double-click on the installer program, named **Install miniDSP-2x4-HD.app**. Once installation is complete, the plugin will automatically start.

Note: No USB driver installation is required for Mac OS X.

## 3.2 HARDWARE CONNECTIVITY

### 3.2.1 Digital input

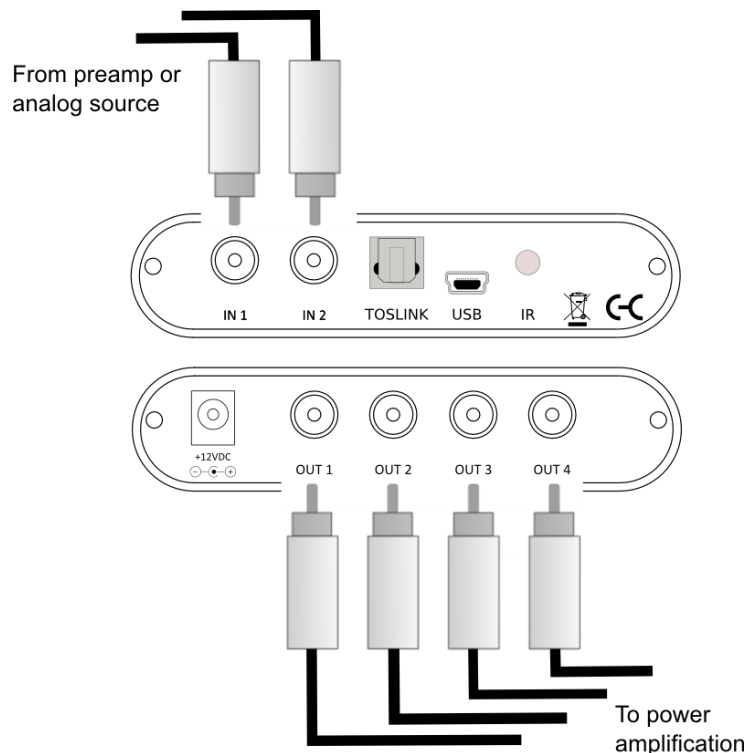
Connect a single digital source to the optical (TOSLINK) connector. Thanks to its asynchronous sample rate convertor (ASRC), all sample rates between 44.1 and 192 kHz are accepted.



Note: the digital input accepts only a stereo PCM digital signal. It does not accept encoded or multichannel digital audio (such as Dolby Digital or DTS).

### 3.2.2 Analog inputs and outputs

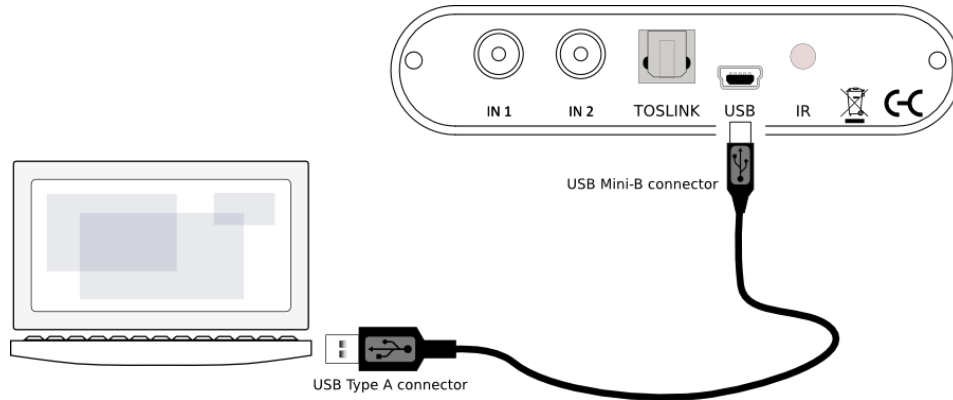
Connect an analog source or preamplifier to the RCA connectors on the front panel, and connect the four analog outputs to power amplification or subwoofer(s). Be sure to take careful note of the channel numbering shown in this diagram and on the rear panel.



### 3.2.3 USB

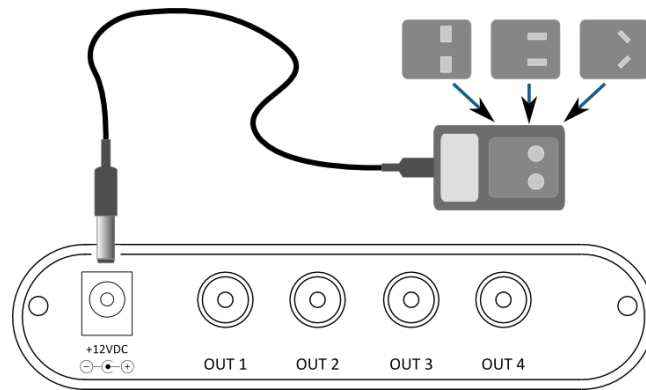
Connect the USB (Type B mini) port of the miniDSP 2x4 HD to a free USB 2.0 port on your computer using the supplied cable. The USB is used for:

- Configuration and real-time control of the miniDSP 2x4 HD,
- Streaming audio from a computer or other device such as a music streamer, at up to 192 kHz, and
- Streaming the processed audio (four output channels) from the miniDSP 2x4 HD to the computer.



### 3.2.4 DC power

The supplied 12 VDC power supply includes a set of interchangeable power pins (for USA, UK, Europe and Australia). Fit the correct pins for your country. Connect the DC plug to the 12 VDC power socket.

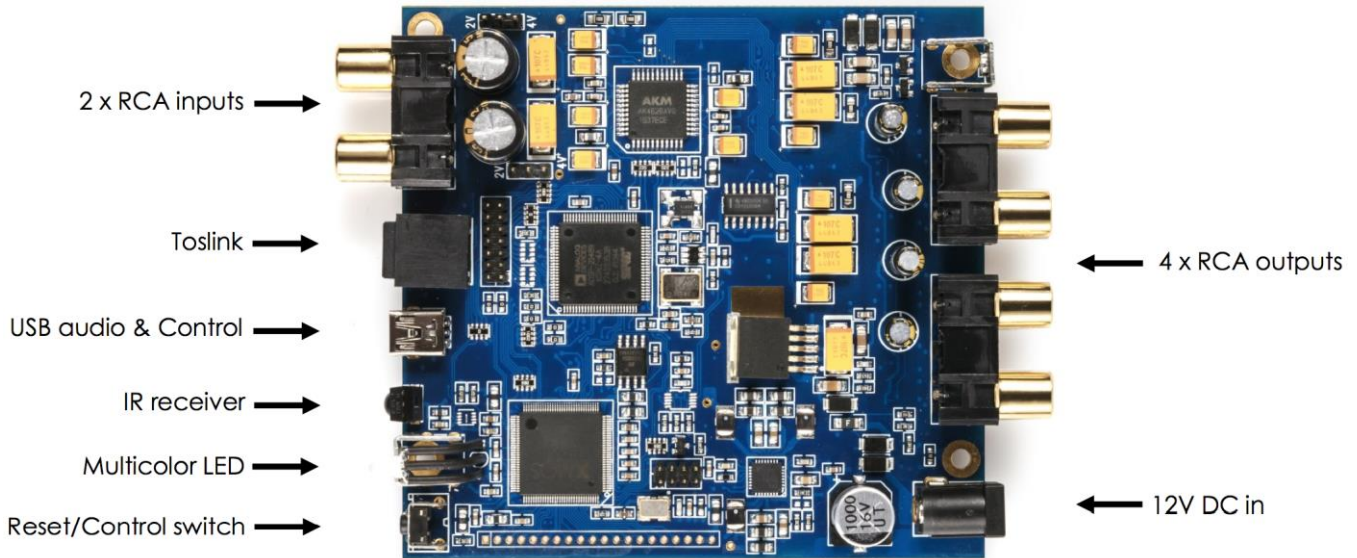


Apply power to the processor only after all input and output connections have been made. The processor can usually be left powered on. If powered on and off, the following sequence is recommended:

- On: Power on line-level equipment, including the 2x4 HD, then turn on power amplification.
- Off: Turn power amplification off, then power off line-level equipment, including the 2x4 HD.

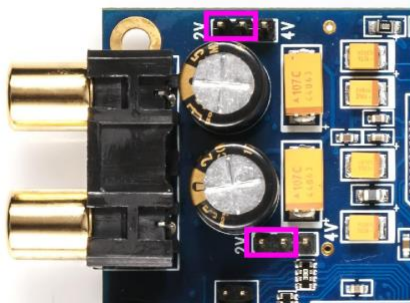
## 4 THE 2X4 HD KIT BOARD

The 2x4 HD Kit board is exactly the same as the 2x4 HD “in a box”, but without the box. All functionality and I/O is identical. The board can be built into your own box and/or combined with other boards such as power amplifiers. This annotated photograph shows the main features of the board:

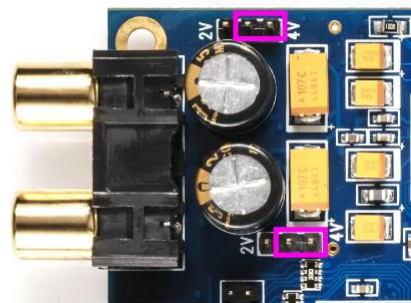


The only user-accessible jumpers are used to set the analog input sensitivity. The jumpers are located near the RCA input jacks, and can be used to set the analog input sensitivity to either 2 VRMS or 4 VRMS, as shown by the highlighted locations below. The default setting as shipped is 2 VRMS.

2 VRMS sensitivity



4 VRMS sensitivity

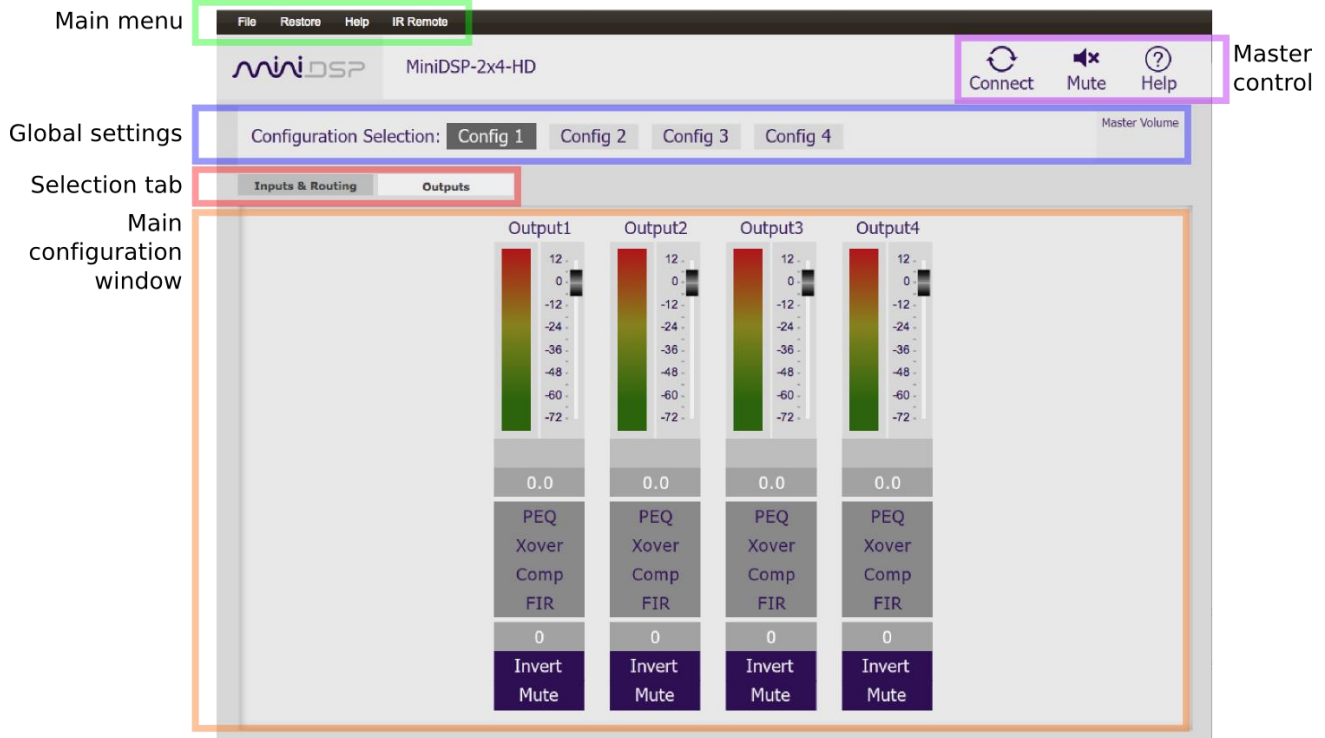


The board must be powered **OFF** when changing these jumpers.

## 5 CONFIGURING THE 2x4 HD

The 2x4 HD processor is configured with the **miniDSP-2x4-HD** plugin / user interface program. Once fully configured, the computer is no longer required, as source and preset selection can be done with a remote control—see [Infrared Remote Control](#) on page 37. If desired, however, the plugin can remain connected during use for real-time (“live”) control of all audio processing.

This screenshot shows the **miniDSP-2x4-HD** plugin with the key areas highlighted:

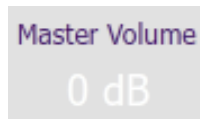


During *initial* configuration of the processor, it is strongly recommended that any connected amplification be powered off.

The **Mute** button disables all audio output:



The **Master Volume** display shows the current volume setting:



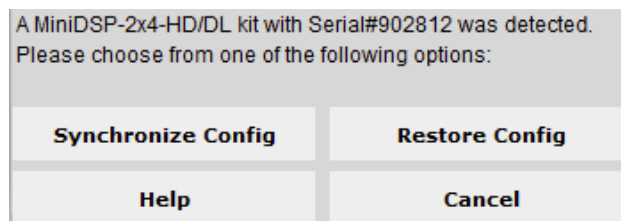


## 5.1 SYNCHRONIZING WITH THE PROCESSOR

Communication with the processor takes place over a USB connection. Ensure that the computer is connected to the miniDSP 2x4 HD by a USB 2.0 port. Then click on the Connect button:



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:



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

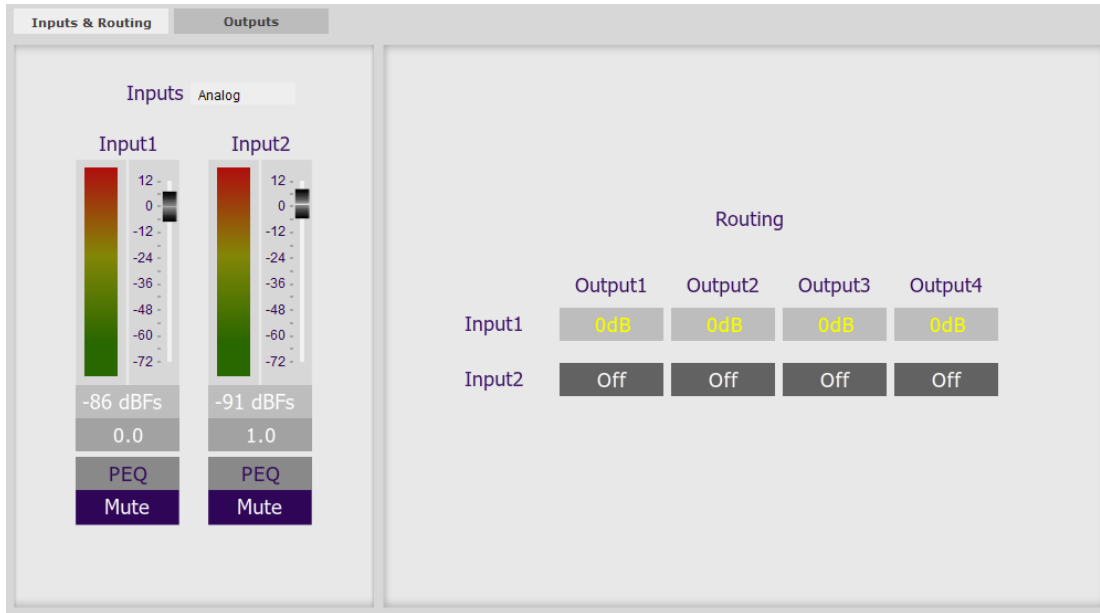
This option brings up a help screen explaining the options.

### Cancel

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

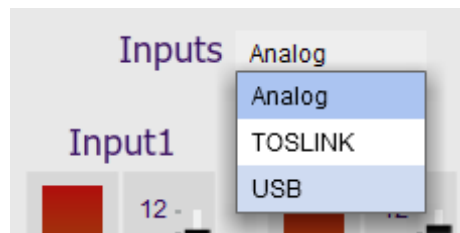
## 5.2 INPUTS & ROUTING TAB

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

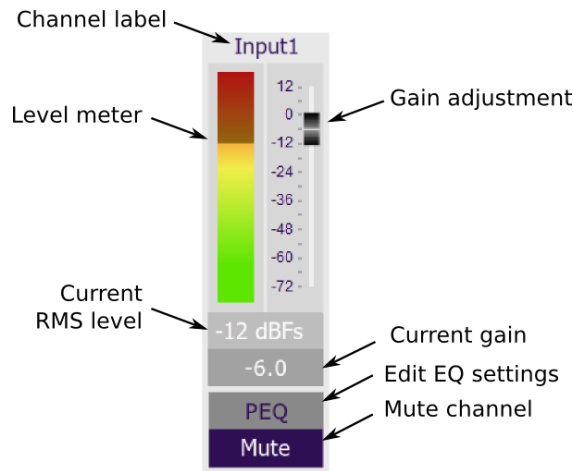


### 5.2.1 Input selection

When the plugin is connected to the 2x4 HD, 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 – see page 37.)



## 5.2.2 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 22.

### Mute

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



### 5.2.3 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 show the mix level (0 dB by default).

The matrix mixer makes the 2x4 HD adaptable to many audio processing situations. Here is an example for a two-way active loudspeaker, where the left input channel is routed to outputs 1 and 2 and the right input channel is routed to outputs 3 and 4 (to change the input and output channel labels, see pages 19 and 21):

	Woof L	Tweet L	Woof R	Tweet R
Left	0dB	0dB	Off	Off
Right	Off	Off	0dB	0dB

For subwoofer integration, the left and right inputs are routed to outputs 1 and 2. They are also summed and routed to outputs 3 and 4 for use as mono subwoofer feeds:

	Out L	Out R	Sub 1	Sub 2
In L	0dB	Off	0dB	0dB
In R	Off	0dB	0dB	0dB

For multisub control and integration, input 1 is connected to the AVR subwoofer output and is routed to all four outputs:

	Sub 1	Sub 2	Sub 3	Sub 4
From AVR	0dB	0dB	0dB	0dB
Unused	Off	Off	Off	Off

At each cross-point, the gain of the signal being mixed can be adjusted to a value between -72 and +12 dB. To adjust the gain, right-click on the crosspoint and a gain control will appear. Adjust the gain with the slider, or by typing in the value directly, then click **close**:

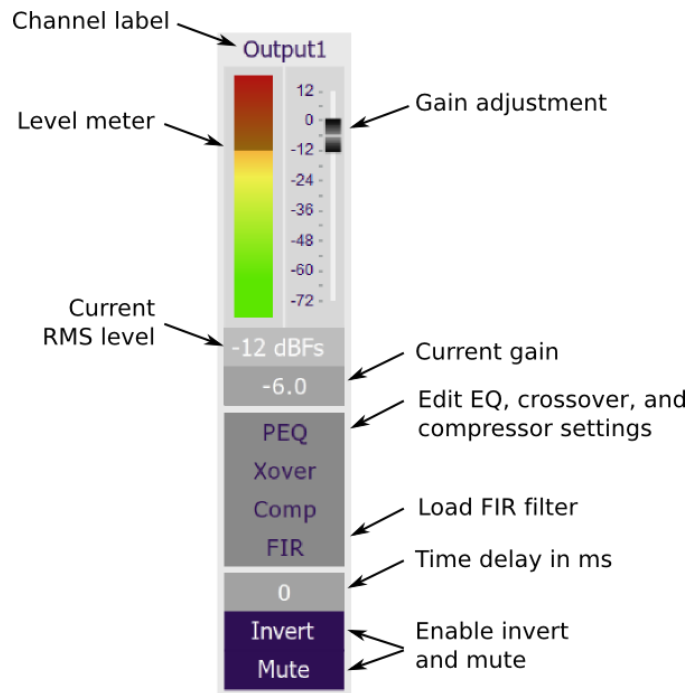
	Output1	Output2	Output3	Output4
Input1	0dB	-3.1	Off	Off
Input2	Off	Off	0dB	0dB

### 5.3 OUTPUT TAB

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

#### 5.3.1 Channel strip layout

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



#### 5.3.2 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.

#### 5.3.3 Level meter and gain control

##### Level meter, current RMS level

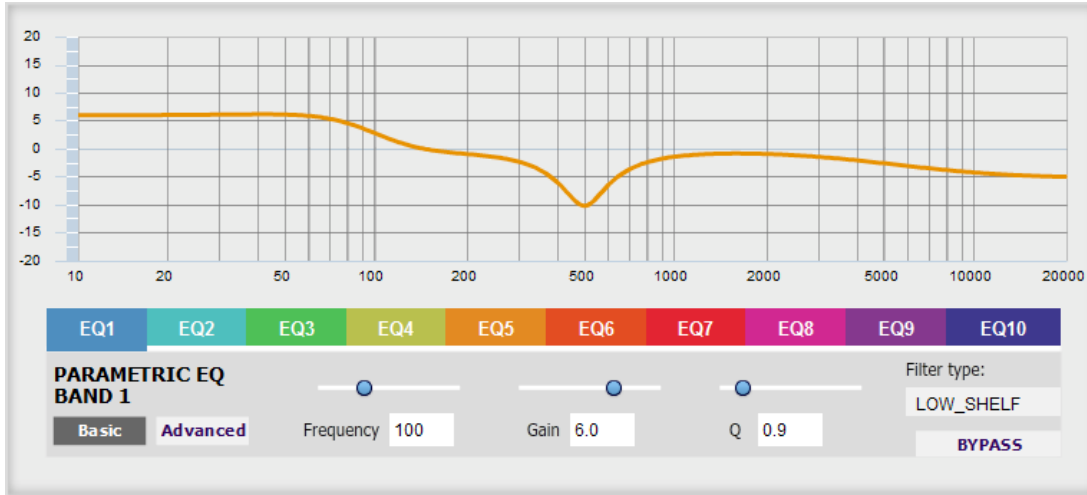
Displays the current signal level in real time. (The plugin must be in online mode to display signal levels.)

##### 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.)

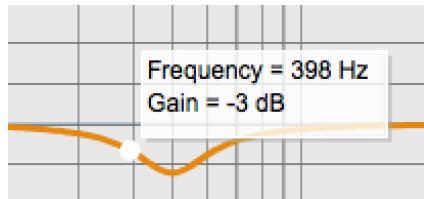
### 5.3.4 Parametric EQ

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 28.

## 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.
- SUB\_EQ** Create a dip or a peak in the frequency response at low frequencies (10 to 50 Hz). This filter type is similar to PEAK but gives more accurate results for low frequencies. Note that activating any SUB\_EQ filter reduces the number of available filters on that channel from ten to nine.

## Frequency

For the PEAK and SUB\_EQ filter types, 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.

## Gain

For the PEAK and SUB\_EQ filter types, 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.

## Q

Q controls the “sharpness” of the filter. For the PEAK and SUB\_EQ filter types, 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.

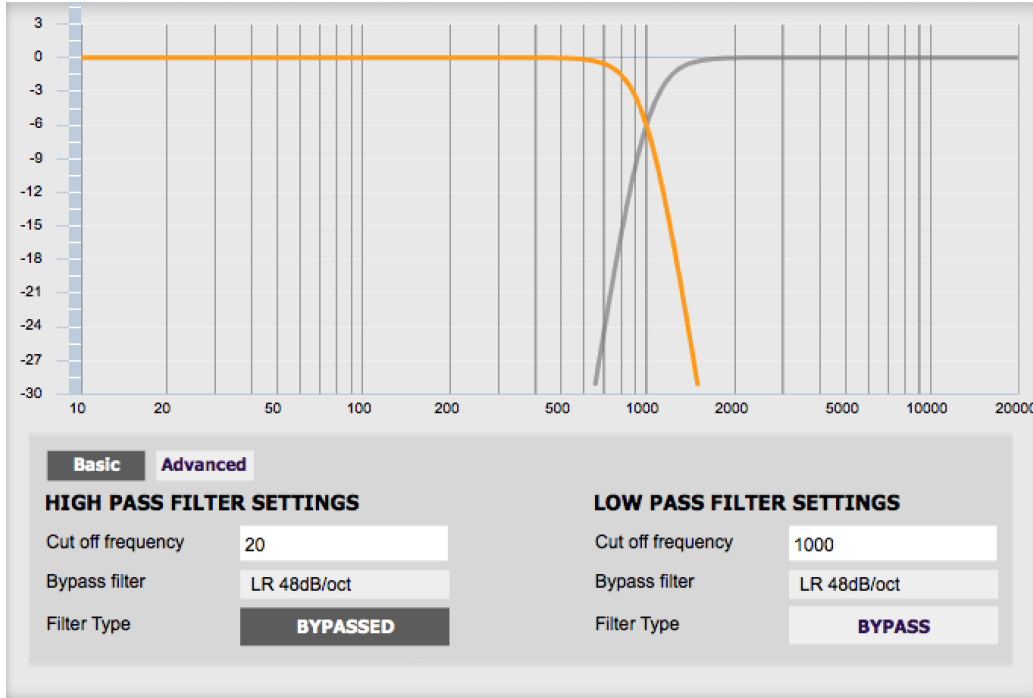
## Bypass

The **Bypass** button enables or disables a filter. The filter is bypassed if the button is "lit". (Note that all other filters are still operational unless individually bypassed.) A filter will also have no effect if its gain is set to 0.0.



### 5.3.5 Crossover

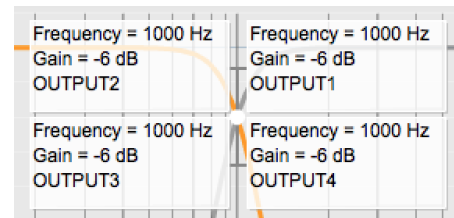
Each output channel has independent high pass and low pass filters. Click on the **Xover** button to open the crossover settings window:



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 28.

## 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. The filter is bypassed when the button is "lit".



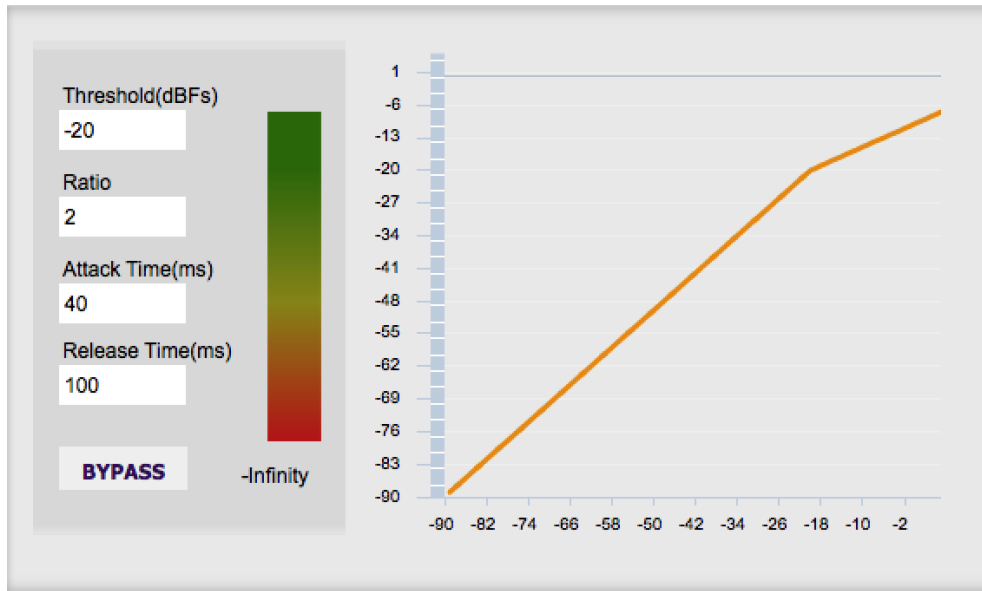
Each channel can be linked to one other channel. When a channel is linked to another, the crossover settings of that channel are mirrored to the other. Typically, the corresponding drivers on the left and right channels are linked: left and right tweeter, left and right woofer, and so on. To link a channel,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



### 5.3.6 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:



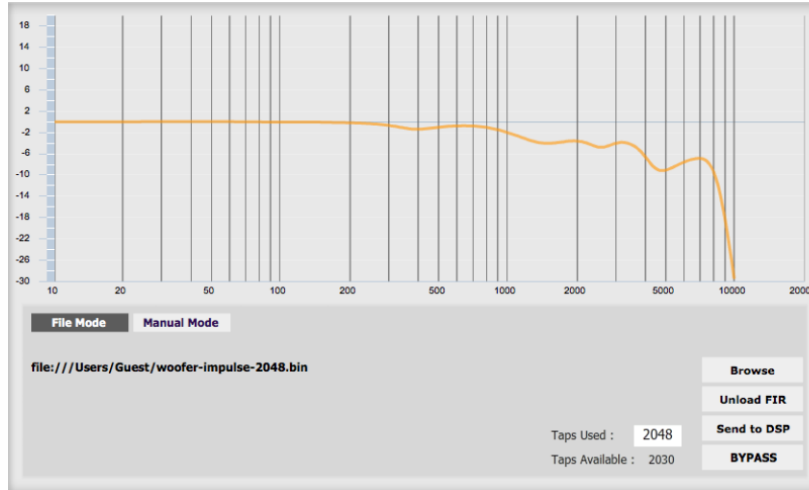
(Note that the compressor algorithm is bypassed by default, so click on the **Bypass** button to see the curve as shown here.)

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](#).

### 5.3.7 FIR

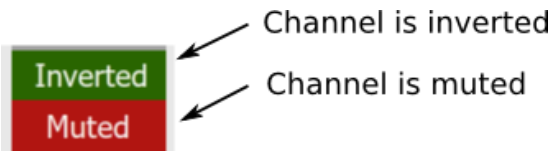
Each output channel has an FIR filter bank with a variable number of taps. Click on the **FIR** button to open the FIR filter settings window:



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 of both. FIR filtering is described more in [FIR filtering and design](#), starting on page 31.

### 5.3.8 Invert and mute

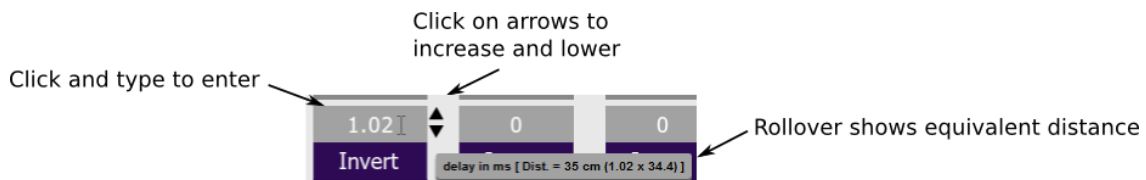
Each channel can be inverted in polarity, and individually muted. When either of these options is selected, the visual indicator on the button is "lit":



### 5.3.9 Time delay

A delay of up to 80 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 (0.02 ms) increments. The maximum time delay of 80 ms corresponds to a distance of approximately 27.5 meters (about 90 feet).

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



## 5.4 CUSTOM BIQUAD PROGRAMMING

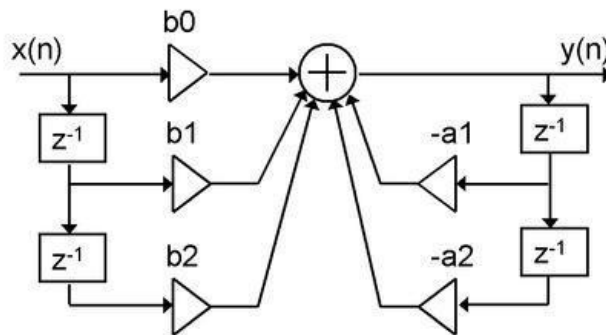
Custom biquad programming is available in the PEQ and Crossover blocks. Its purpose is to allow you to directly provide the low-level parameters aka *biquad coefficients* that control the digital filters of the processor, thus providing an almost infinite degree of flexibility.

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.

### 5.4.1 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:

$$H(z) = \frac{b_0 + b_1z^{-1} + b_2z^{-2}}{1 + a_1z^{-1} + a_2z^{-2}}$$



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:  $a_1$ ,  $a_2$ ,  $a_0$ ,  $b_1$ , and  $b_2$ . These numbers are called the *coefficients*.

### 5.4.2 Using custom biquad programming

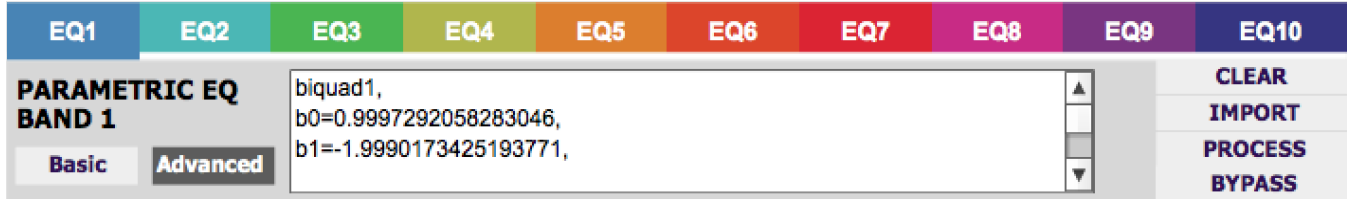
Each crossover block and PEQ filter has a selector that switches it to advanced mode:



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 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:

```

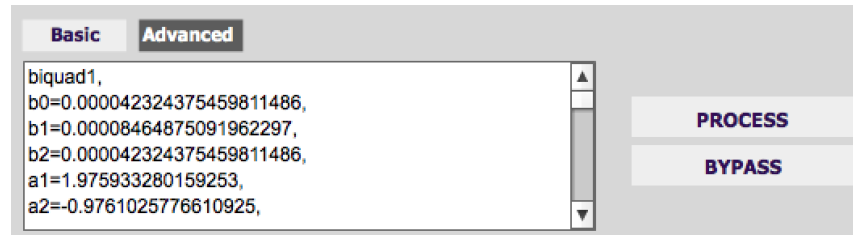
biquad1,
b0=0.998191200483864,
b1=-1.9950521500467384,
b2=0.996920046761057,
a1=1.9950521500467384,
a2=-0.9951112472449212,
biquad2,
b0=0.999640139948623,
b1=-1.9981670485581222,
b2=0.9985489719847982,
a1=1.9981670485581222,
a2=-0.9981891119334211,
biquad3,
...
biquad4,
...
biquad10,
b0=1.0010192374642126,
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.



### 5.4.3 Biquad design software

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

#### 5.4.3.1 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):

- [http://www.minidsp.com/images/fbfiles/files/All\\_digital\\_coefs\\_v1-20101026.zip](http://www.minidsp.com/images/fbfiles/files/All_digital_coefs_v1-20101026.zip)

#### 5.4.3.2 Room EQ Wizard (REW)

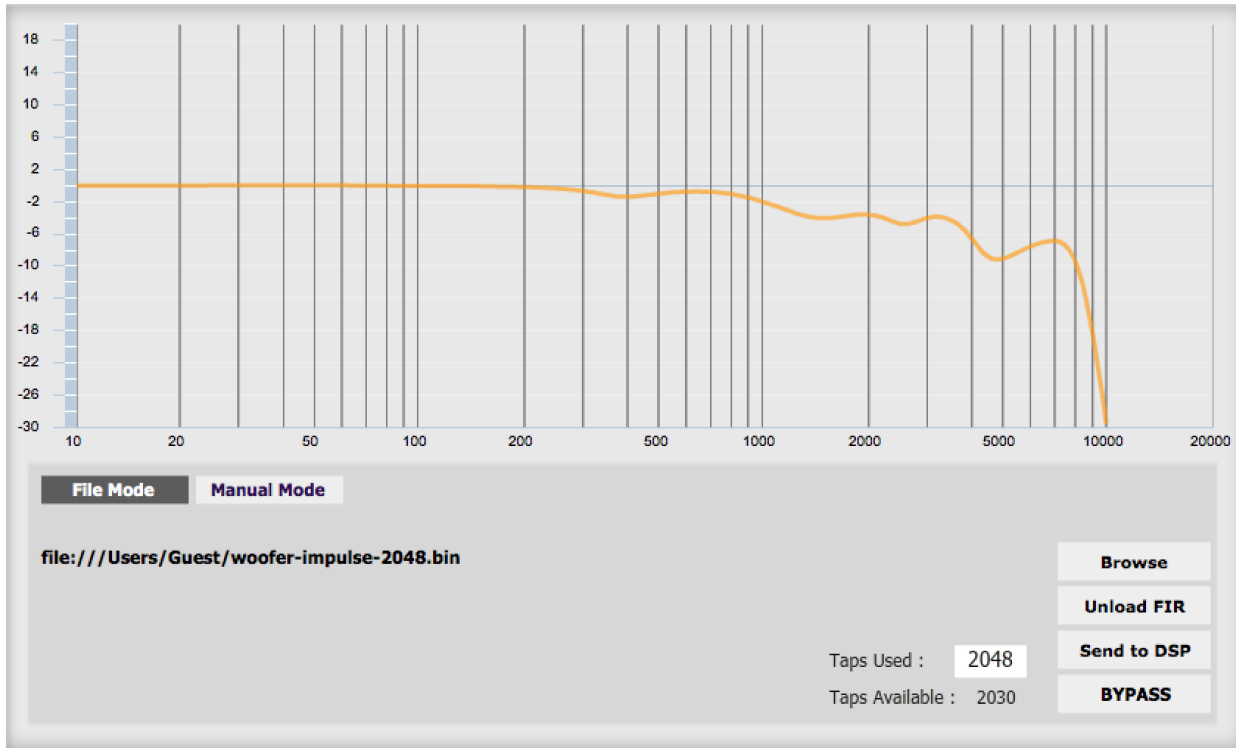
Room EQ Wizard is a free acoustic measurement and analysis tool, available for Windows, Mac and Linux platforms. It includes the ability to automatically generate a bank of parametric EQ biquads based on a measurement. These coefficients can be saved to a file from REW and loaded directly into a PEQ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

- <http://www.roomeqwizard.com/#downloads>

For guidance on using this feature, please refer to the app note [Auto EQ with REW](#).

## 5.5 FIR FILTERING AND DESIGN

FIR filtering is a powerful and advanced feature of the 2x4 HD. 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.



### 5.5.1 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 2x4 HD can compute a total of 4096 taps. 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 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**):

Taps Used :	1024
Taps Available :	2036

### 5.5.2 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 software packages available for this purpose (both freeware and commercial). Please see the [FIR filter tools](#) page on our website.

### 5.5.3 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:

```
b0 = 1,  
b1 = -1,  
b2 = 0.5,  
b3 = -0.5,  
b4 = 0.2,  
b5 = 1,
```



### 5.5.4 Loading filter coefficients

#### In **File Mode**:

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

#### In **Manual Mode**:

1. Cut and paste the coefficients from the text output of the design program.
2. Press the **Process** button.
3. Confirm that the frequency response graph is as you expect.
4. Press **Send to DSP**. This will write the coefficients into the DSP's memory.
5. 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.

## 5.6 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.

### 5.6.1 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.

### 5.6.2 Selecting a configuration

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



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.

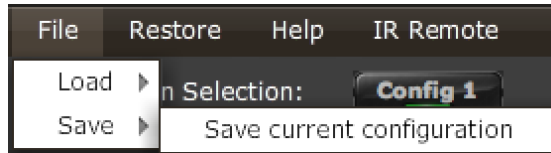
#### The plugin is in offline mode

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.

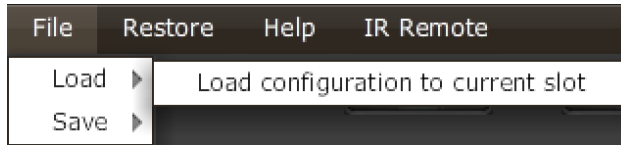
### 5.6.3 Saving and loading configurations

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.

### 5.6.4 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.



## 5.7 KEYBOARD SHORTCUTS

The **miniDSP-2x4-HD** 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

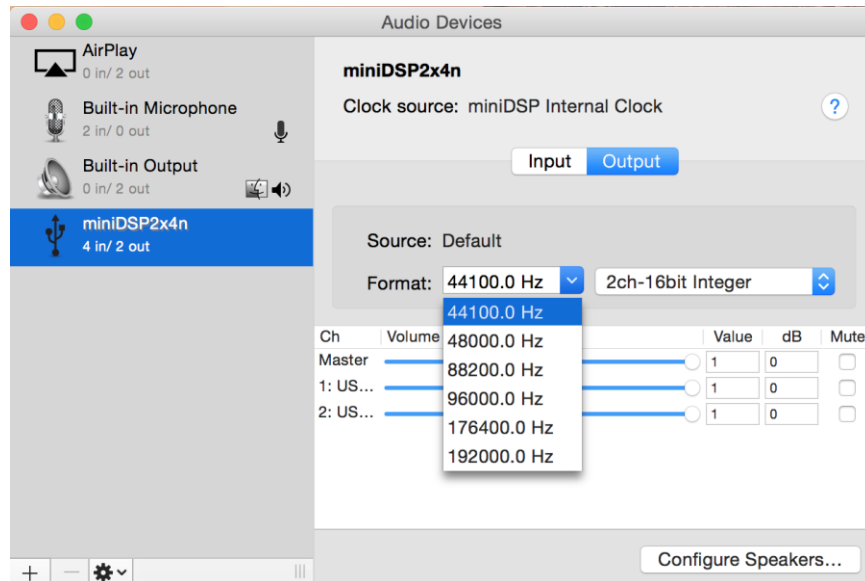
The Space bar toggles buttons that have two states, such as **Bypass**, **Invert**, and **Mute**, if they have the focus.

## 6 USB AUDIO

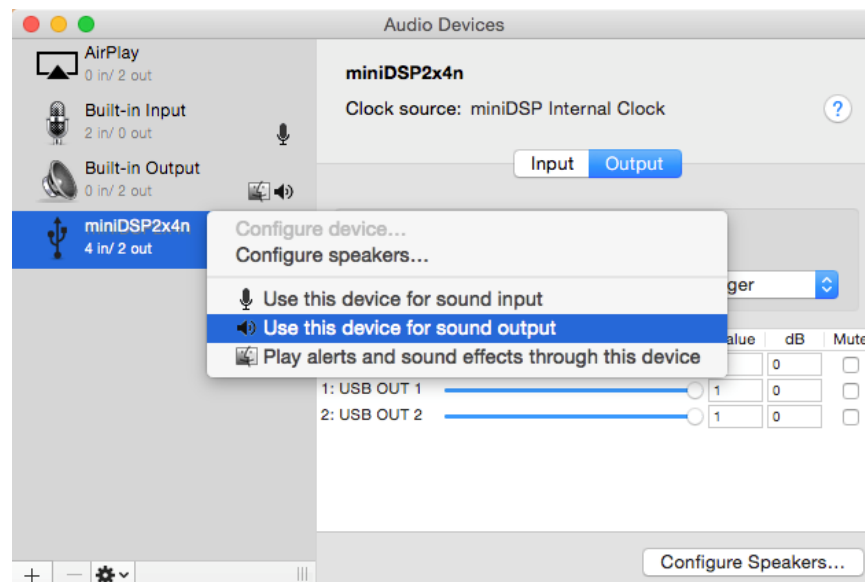
The miniDSP 2x4 HD accepts stereo PCM audio at sample rates of 44.1, 48, 88.2, 96, 176.4, and 192 kHz on its USB audio input. The same USB connector is used both for streaming audio and configuration.

### 6.1 MAC OS X

Open the program **Audio MIDI Setup** (in **Applications->Utilities**). The 2x4 HD will appear as “miniDSP2x4n” in the list on the left hand side. Clicking on it will show the input and output channels. In this example, the available sample rates are shown in the dropdown menu:



To set the 2x4 HD as the default audio output device, right-click and select “Use this device for sound output”.

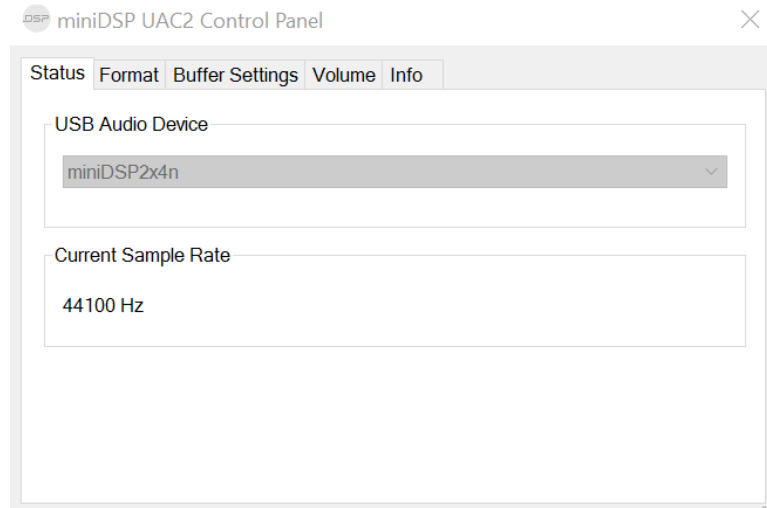


## 6.2 WINDOWS

To configure the 2X4 HD, open the **miniDSP UAC2 Control Panel** (from Start Menu -> miniDSP Ltd). It has several panes, described below.

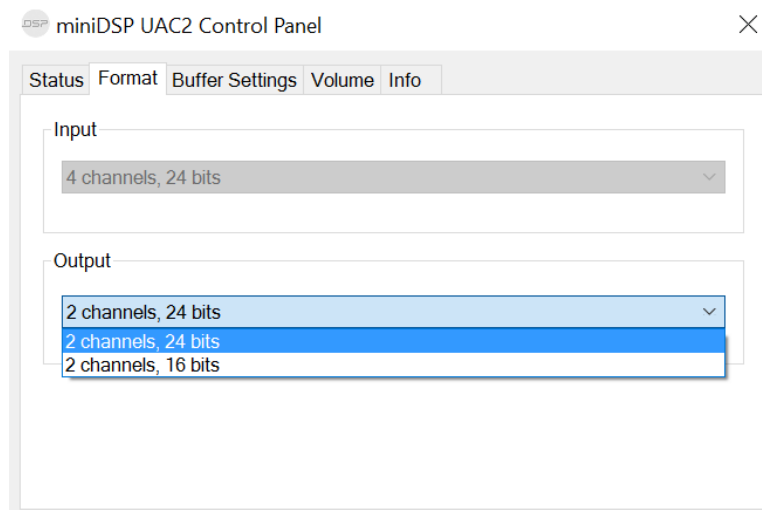
### 6.2.1 Status

This panel shows the current sample rate of the 2x4 HD. This setting cannot be changed in the Control panel, but simply reflects the current sample rate of the 2x4 HD.



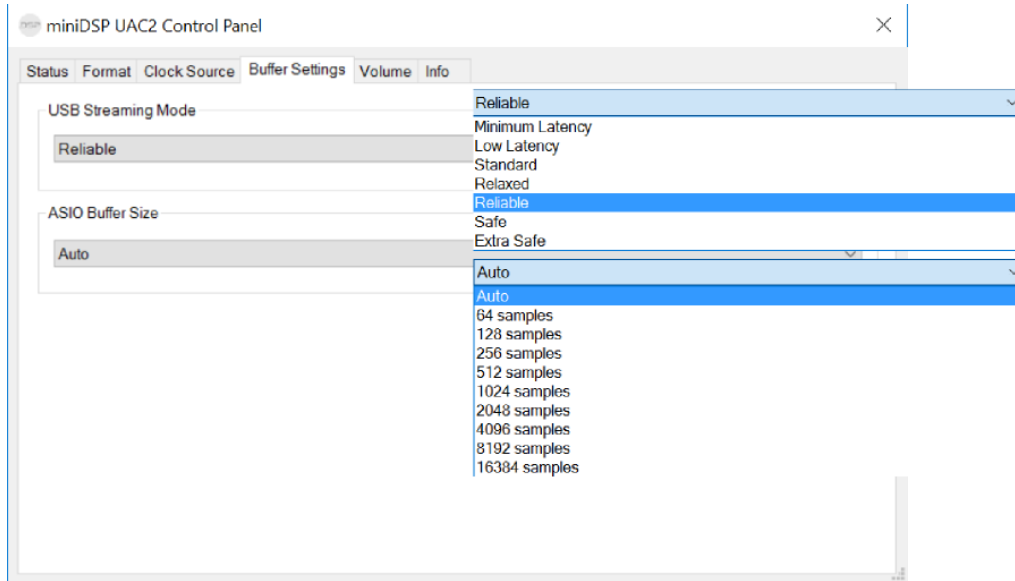
### 6.2.2 Format

This panel selects the input and output data format.



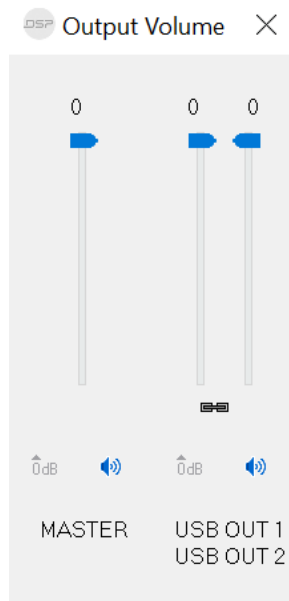
### 6.2.3 Buffer settings

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.



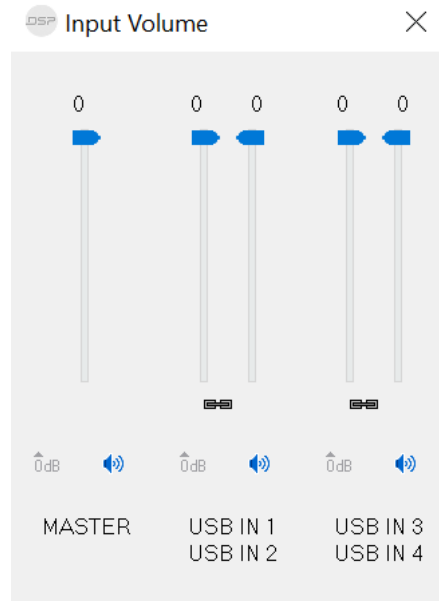
### 6.2.4 Volume

This panel contains the volume controls for input and output channels. This is the output volume display:



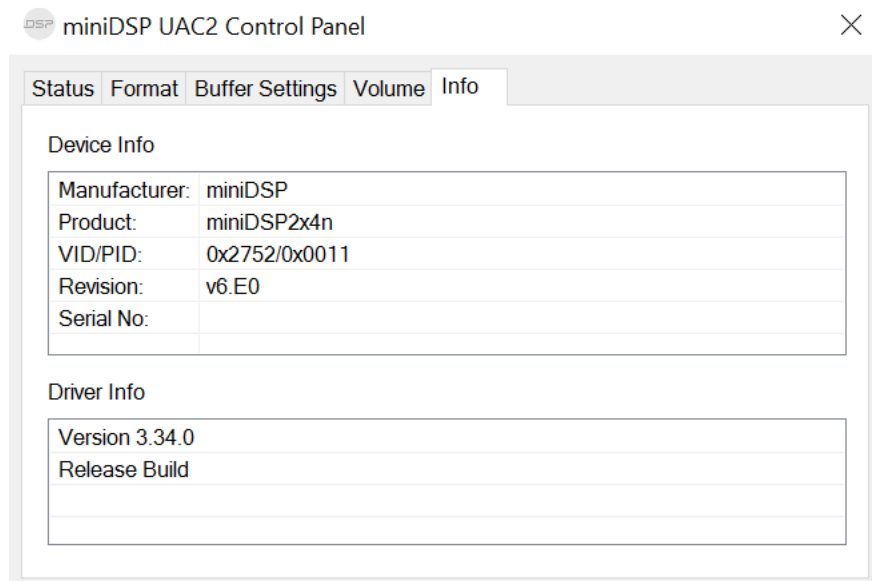
- To reset the master volume control or a pair of channels to 0 dB (no attenuation), click the **0dB** button.
- To mute all channels, click either of the speaker icons.
- To control volume separately for each channel, click on the “Link” icon to turn it off.

There is a similar set of controls for the input channels:



### 6.2.5 Info

This panel shows information about the 2x4 HD.





## 7 INFRARED REMOTE CONTROL

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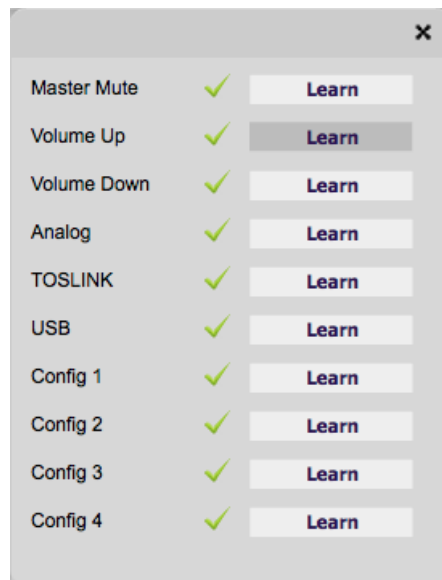
Once configuration is complete, the computer is not required and can be disconnected. An infrared remote can be used to control volume, mute, preset selection, and input selection.

The 2x4 HD can “learn” the control codes of your current remote if it supports one of the following remote control codes:

- 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.

This screenshot shows the IR learning screen:



To "unlearn" a command, press the **Learn** button and wait for the plugin to time out.

## 8 ADDITIONAL INFORMATION

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### 8.1 SPECIFICATIONS

<b>Computer connectivity</b>	Driverless USB 2.0 control interface for Windows and Mac OS X
<b>USB audio input and output</b>	<p>XMOS asynchronous USB audio, 44.1 to 192 kHz, USB Audio Class 2 compliant. ASIO driver for Windows, driverless for Mac OS X.</p> <ul style="list-style-type: none"> <li>- Input (computer to 2x4 HD): two channels selected from plugin or by IR remote</li> <li>- Output (2x4 HD to computer): four channels (processed output channels from DSP)</li> </ul>
<b>Digital audio Input</b>	TOSLINK optical. A high quality onboard Asynchronous Sample Rate Converter ensures compatibility with most sample rates, from 20–216kHz.
<b>Analog audio inputs</b>	<p>Unbalanced stereo (2 channels) analog audio on RCA connectors</p> <ul style="list-style-type: none"> <li>- Max input of 4V or 2V RMS, jumper–selectable</li> <li>- Input impedance: 10k<math>\Omega</math></li> <li>- THD+N: 0.003% (RCA to USB)</li> <li>- Dynamic range: 102dB</li> </ul>
<b>Analog audio outputs</b>	<p>Unbalanced analog audio (4 channels) on RCA connectors</p> <ul style="list-style-type: none"> <li>- Max output: 2V RMS</li> <li>- Output impedance: 560<math>\Omega</math></li> <li>- THD+N: 0.001% (USB to RCA)</li> <li>- Dynamic range: 103dB</li> </ul>
<b>Audio resolution</b>	24-bit input and output, 96 kHz internal sample rate (depending on plugin)
<b>Audio processing</b>	32-bit floating-point processor
<b>FIR capabilities</b>	FIR filtering with number of taps assignable to each output channel. FIR filters are designed by third-party programs. FIR file format: IEEE 754 single-precision binary floating-point.
<b>Storage/presets</b>	<p>All settings controllable in real time from software user interface.</p> <p>4 onboard presets stored in local flash memory.</p>
<b>Infrared remote</b>	<p>“Learning remote” capabilities (NEC, Philips, Sony)</p> <p>Controls master volume, mute, input selection, preset selection.</p>
<b>Power supply</b>	12 VDC single supply @ 300mA, 2.1 mm center-positive
<b>Dimensions (H x W x D)</b>	27 x 119 x 107 mm

## 8.2 FIRMWARE UPGRADE

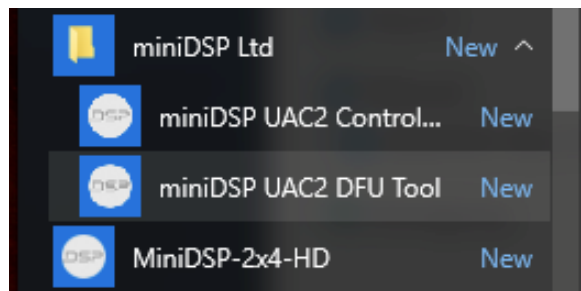
miniDSP may occasionally provide an upgrade to the 2x4 HD MCU firmware to enable new features. To upgrade the MCU firmware, first download and install the latest version of the 2x4 HD plugin from the **User Downloads** section of the miniDSP website.



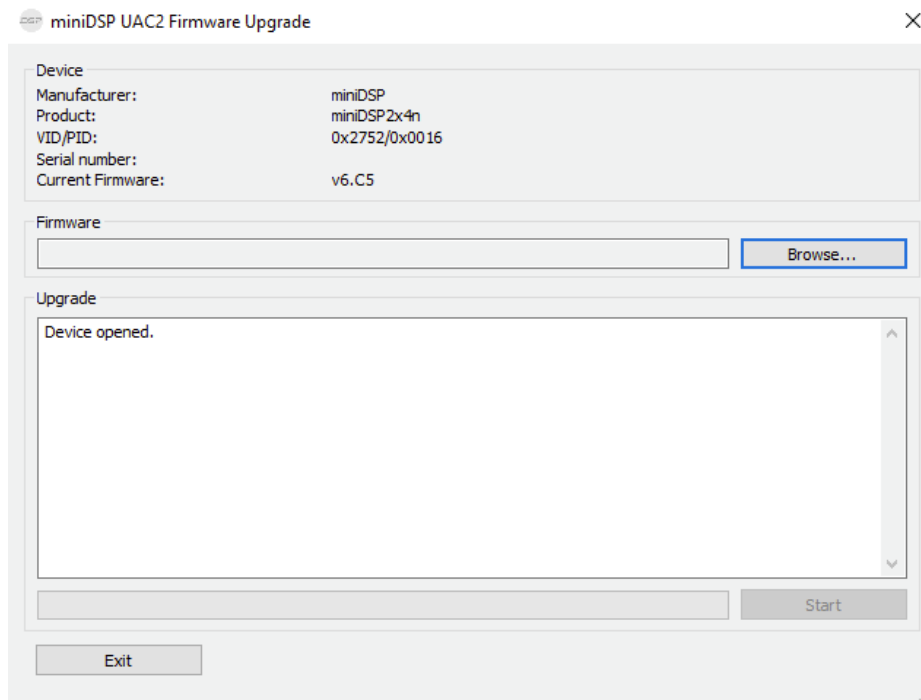
DO NOT DISCONNECT THE USB CABLE OR POWER FROM THE 2x4 HD WHILE FIRMWARE UPGRADE IS IN PROGRESS. DOING SO MAY “BRICK” YOUR 2x4 HD.

### 8.2.1 Windows

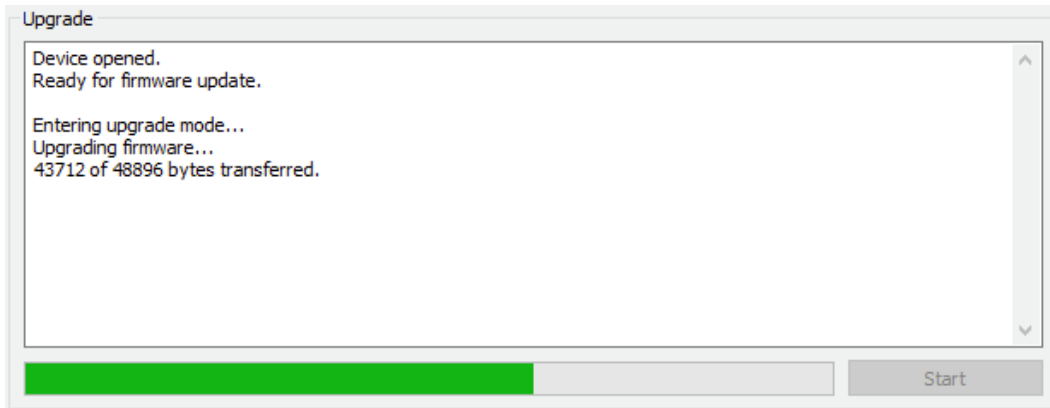
1. Connect the 2x4 HD (or Kit) to your computer via USB (if not already connected) and power it on.
2. Start the **miniDSP UAC2 DFU Tool**.



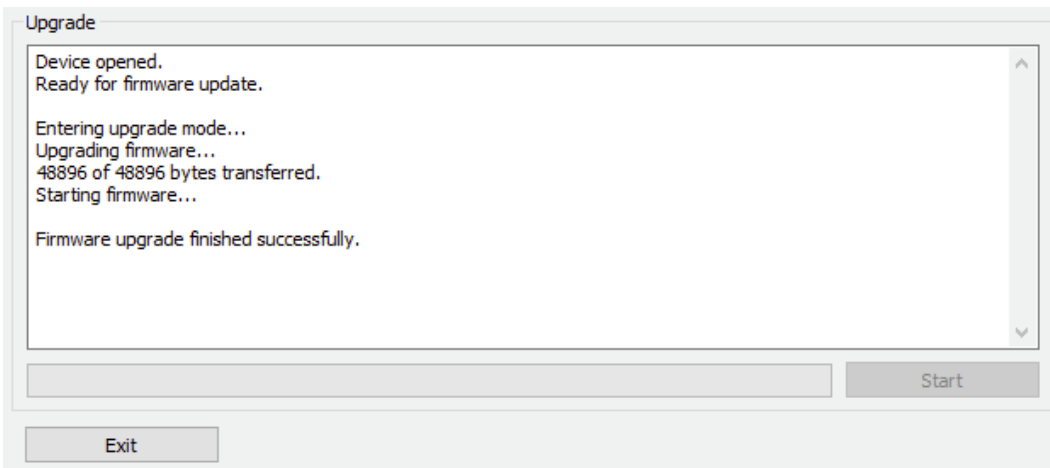
3. The upgrade program will start:



- Click on the **Browse** button, navigate to the folder **XMOS\_Firmware** in the plugin download folder, and select the firmware file. It will have a name like “miniDSP2x4n\_XMOS\_v1.3.bin.” (The version number may be different.)
- Click on the **Start** button.
- You will get a progress bar as upgrade proceeds:



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



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

## 8.2.2 Mac OS X

To load firmware using Mac OS X requires that you use the Terminal program (located in the Applications/Utilities folder). In the examples that follow, black text is the “prompt” printed by Terminal, blue text is text typed in by you, and red text is the program output.



It is important that you type exactly as shown including characters like “.” and “/” where noted. Press the Tab key after typing the first two characters of any filename, to activate auto-completion.

Download the latest software for the 2x4 HD from the User Downloads area of the [minidsp.com](http://minidsp.com) website. Double-click on it to unzip it. Assuming that you have placed it into the Downloads folder on your Mac, you will then type:

```
mymac:~ myname$ cd Downloads/MiniDSP_2x4_HD_101/XMOS_Firmware/
mymac:XMOS_Firmware myname $ ls
miniDSP2x4n_XMOS_v0.13.bin  miniDSP_UAC2_DFU_OSX      readme.txt
mycomputer:Firmware myname $
```

You see in the above list the name of the firmware file, **miniDSP2x4n\_XMOS\_v0.13.bin** (the version number may change). Now change to the folder holding the actual firmware upgrade tool, and then run it:

```
mymac:XMOS_Firmware myname$ cd miniDSP_UAC2_DFU_OSX/
mymac:miniDSP_UAC2_DFU_OSX myname $ source setup.sh
mymac:miniDSP_UAC2_DFU_OSX myname $ ./xmosdfu --download ../miniDSP2x4n_XMOS_v0.13.bin
VID = 0x5ac, PID = 0x8007, BCDDevice: 0x300
VID = 0x5ac, PID = 0x8007, BCDDevice: 0x300
...
VID = 0x2752, PID = 0x11, BCDDevice: 0x6e0
XMOS DFU application started - Interface 3 claimed
Detaching device from application mode.
Waiting for device to restart and enter DFU mode...
VID = 0x5ac, PID = 0x8007, BCDDevice: 0x300
VID = 0x5ac, PID = 0x8007, BCDDevice: 0x300
...
VID = 0x2752, PID = 0x11, BCDDevice: 0x6e0
... DFU firmware upgrade device opened
... Downloading image (../miniDSP2x4n_XMOS_v0.13.bin) to device
... Download complete
... Returning device to application mode
mymac:miniDSP_UAC2_DFU_OSX myname $
```

You can now proceed to use the 2x4 HD as normal.

### 8.3 TROUBLESHOOTING

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

Item#	Symptoms	Troubleshooting recommendation
1	Cannot install software	a. Confirm that you downloaded and installed the required frameworks first (see <a href="#">Software Installation</a> ).
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. b. The Adobe Air environment may require a version update. Download the latest version from <a href="http://get.adobe.com/air/">http://get.adobe.com/air/</a> .
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	No audio or low audio on outputs	a. Check the cabling from the processor to your amplifiers. b. Check that your amplifiers are turned on and that any volume controls are turned up. c. Check that the input meters and output meters are showing adequate signal (if not, see item 5). d. Check that master mute is not enabled. e. Check that the mute buttons in the output channel strips are not enabled. f. Check that the FIR filters have valid coefficients (and are bypassed if they do not). g. Check that your crossover frequencies are correct e.g. that you don't have high pass and low pass frequencies incorrectly set. h. Check that the matrix mixer is sending the correct inputs to the correct outputs.
5	No signal showing on input meters	a. Check the cabling from your source. b. Check that your source is playing audio and that it is not muted or have volume control turned down. c. Check that the plugin is synchronized with the hardware unit.
6	Audio sounds distorted	a. Check the output meter and ensure that you are not overloading the outputs. If necessary, reduce the output gain and/or the amount of boost in the EQ blocks.

6	Audio is coming through the wrong outputs	<ul style="list-style-type: none"> <li>a. Check the cabling from the processor to your amplifiers.</li> <li>b. Check that you have correctly set up the matrix mixer to send the correct inputs to the correct outputs.</li> </ul>
7	Cannot reload a configuration	<ul style="list-style-type: none"> <li>a. Confirm the file format of your file (.xml).</li> <li>b. Confirm the version of the file.</li> </ul>

## 8.4 OBTAINING SUPPORT

1. Check the forums on [minidsp.com](http://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](http://minidsp.desk.com) with:
  - a. The specific product you are having an issue with (in this case, 2x4 HD).
  - b. A clear explanation of the symptoms you are seeing.
  - c. A description of troubleshooting steps (see [Troubleshooting](#) above) performed and your results.