

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
V1.4	Updated installation procedure	31 January 2017
V1.5	Updated software installation, firmware update procedure. Added IP address field, miniDSP remote.	17 August 2018
V1.6	Various corrections, updated USB Audio	22 August 2018
V1.7	Adding I2S support for outputs	27 November 2018
V1.8	Added configuration examples	31 December 2018

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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 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)
- Routing matrix 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:

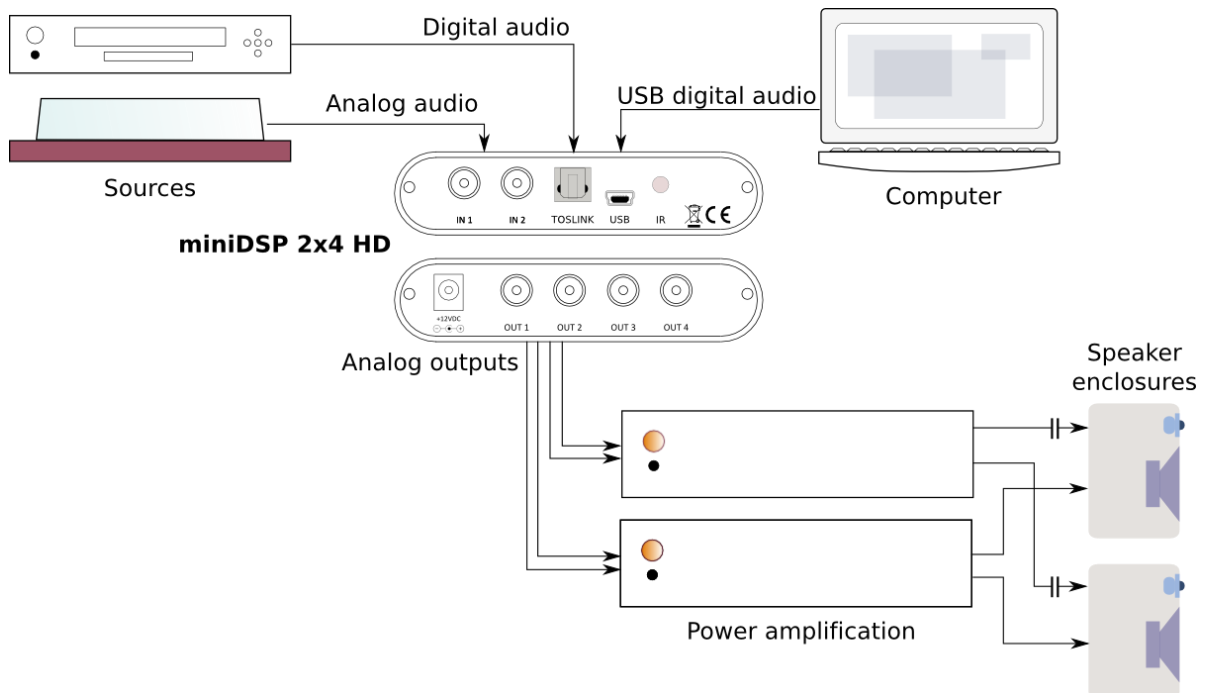


Figure 1 - 2x4 HD as preamp and 2-way crossover

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:

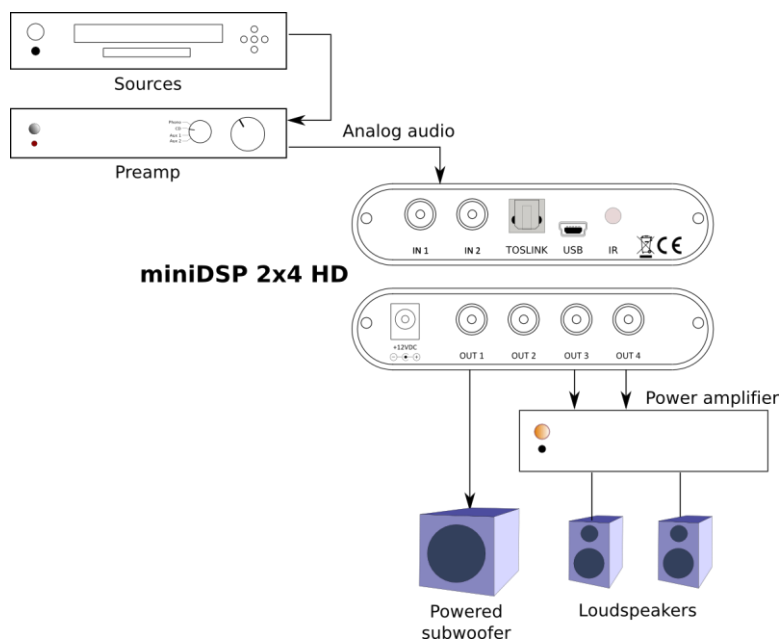


Figure 2 - 2x4 HD for subwoofer integration (2.1-way)

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:

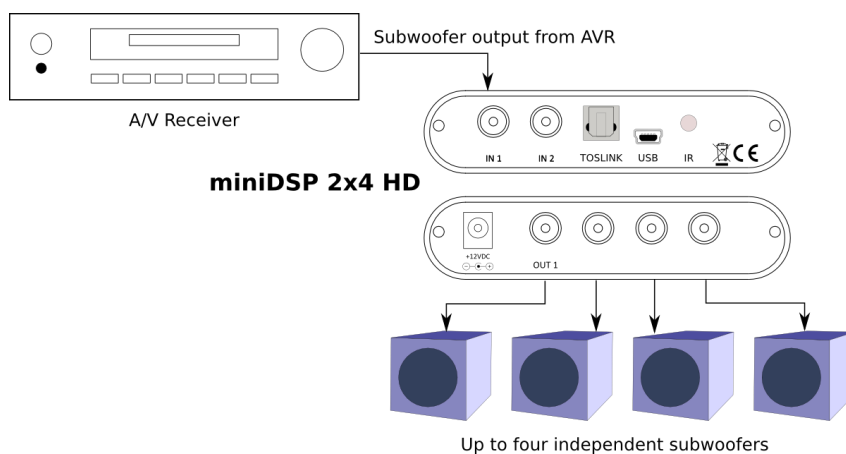


Figure 3 - 2x4 HD for multisub management

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

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 19.

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 43.

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 14.

7. Go into online mode

Apply power to the processor. (See [DC power](#) on page 15 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 20.

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 43.

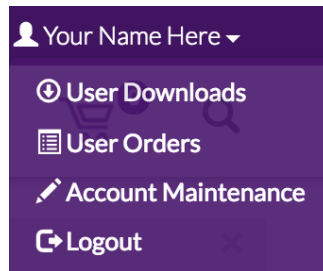
3 SOFTWARE INSTALLATION

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

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 “MiniDSP 2x4 HD1” 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 **miniDSP Plug-ins** section and download the zip file under the heading **miniDSP 2x4 HD1**. 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 2x4 HD plugin installation

1. Navigate to the **Plugins** folder of the software download and then to the **Windows** folder.
2. Double-click on the **MiniDSP_2x4_HD.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

1. Connect the 2x4 HD to the computer using the supplied USB cable, and power it on.
2. Navigate to the **WinDrivers** folder of the software download and double-click on the appropriate installer:
 - **miniDSP_UAC2_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

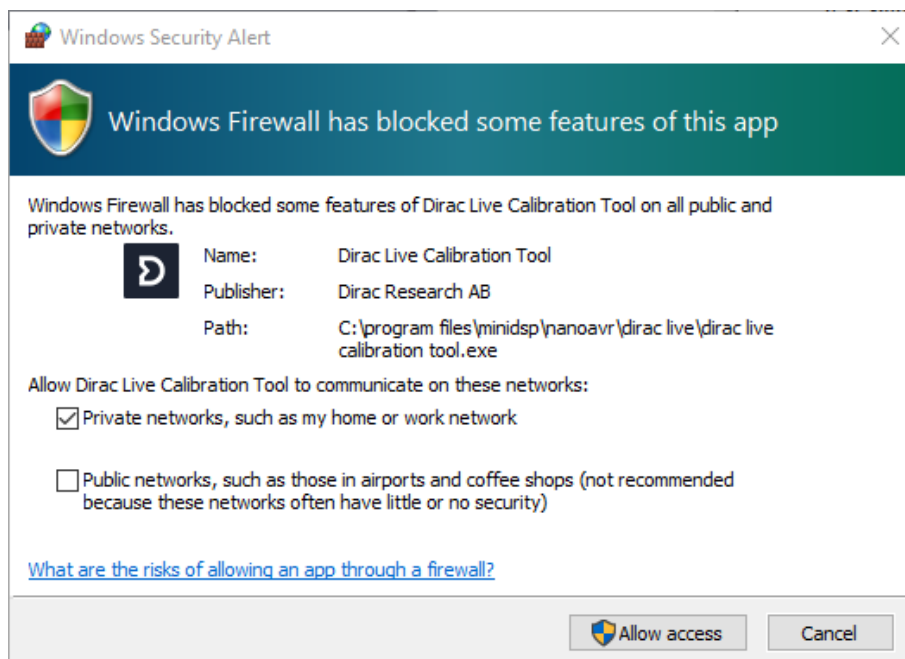
(The version number embedded in the filename may be different.)

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 2x4 HD if you did not have the 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 2x4 HD, 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.”

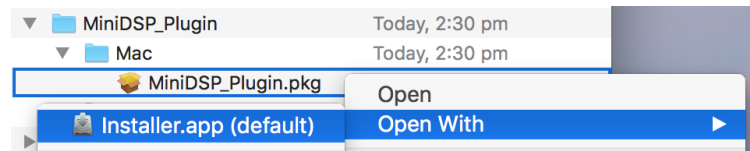


3.2 MAC

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 installer will be **MiniDSP-2x4-HD.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.”



3.2.2 2x4 HD plugin installation

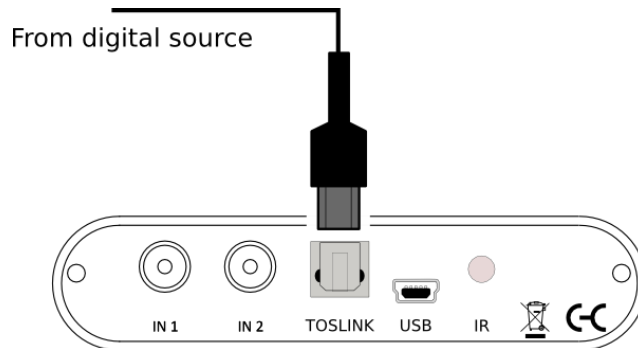
1. Navigate to the **Plugins** folder of the software download and then to the **Mac** folder.
2. The installer program is named **MiniDSP-2x4-HD.pkg**. To run it, double-click on it, or right-click and open as described above. We recommend that you accept the default installation settings.
3. To run the MiniDSP-2x4-HD 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 HARDWARE CONNECTIVITY

4.1 DIGITAL INPUT

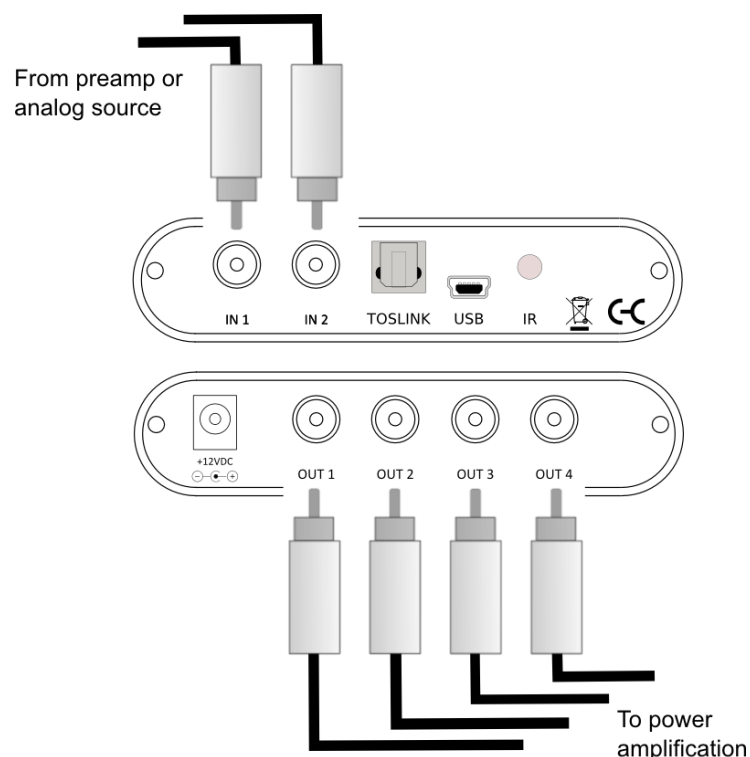
Connect a single digital source to the optical (TOSLINK) connector. Thanks to its asynchronous sample rate convertor (ASRC), all sample rates between 44.1 and 192 kHz are accepted.



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).

4.2 ANALOG INPUTS AND OUTPUTS

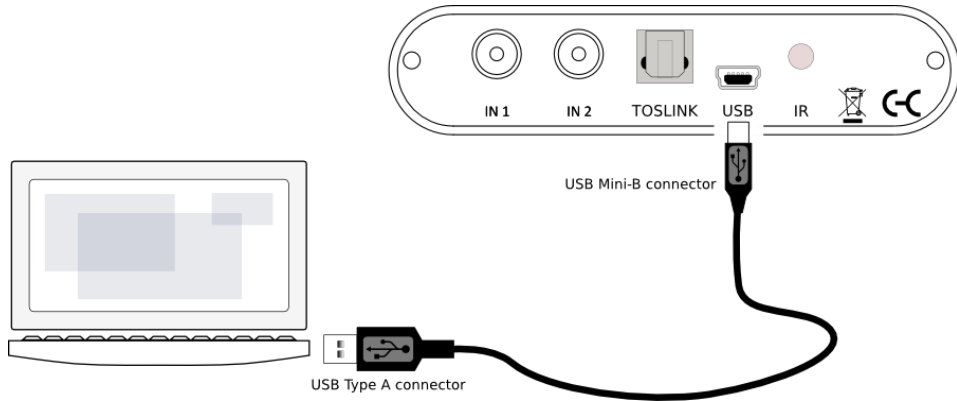
Connect an analog source or preamplifier to the RCA connectors on the front panel, and connect the four analog outputs to power amplification or subwoofer(s). Be sure to take careful note of the channel numbering shown in this diagram and on the rear panel.



4.3 USB

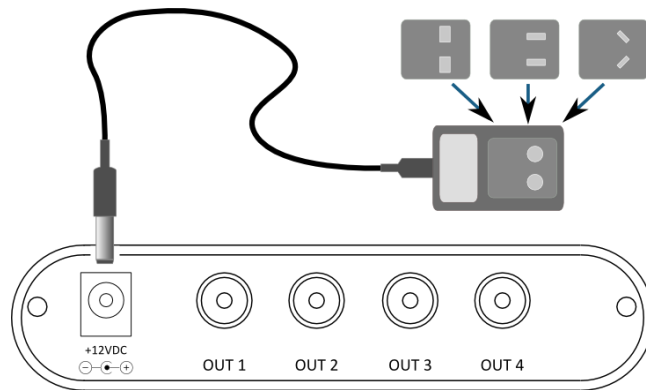
Connect the USB (Type B mini) port of the miniDSP 2x4 HD to a free USB 2.0 port on your computer using the supplied cable. The USB is used for:

- 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.



4.4 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.



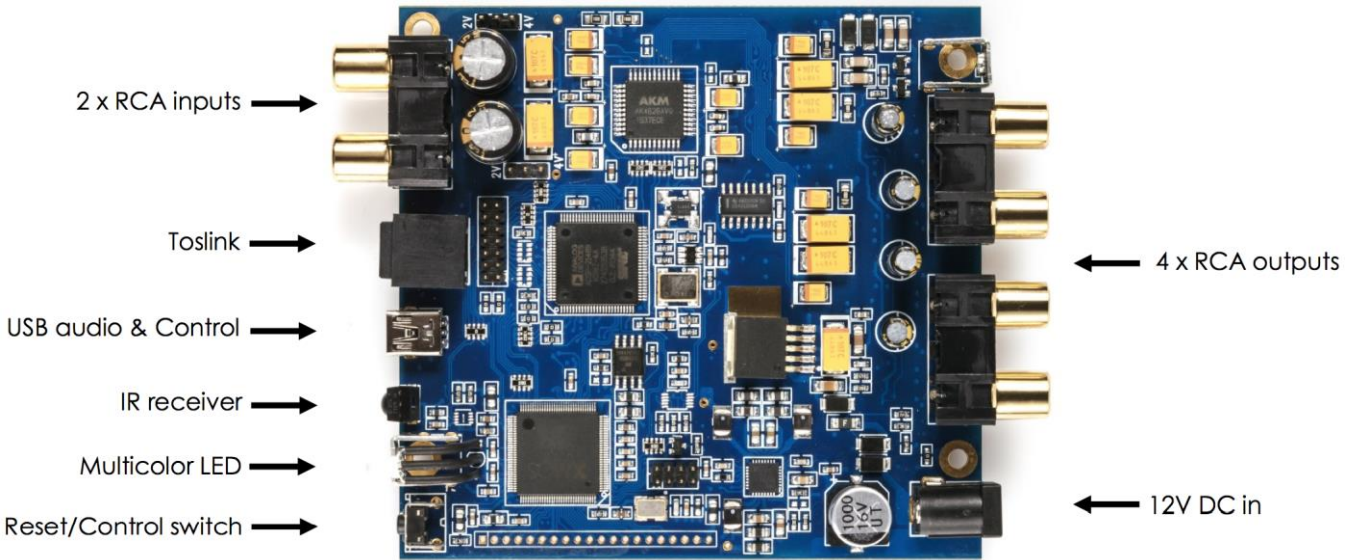
Apply power to the processor only after all input and output connections have been made. The processor can usually be left powered on. If powered on and off, the following sequence is recommended:

- 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.

5 THE 2x4 HD KIT BOARD

5.1 INPUT SENSITIVITY JUMPERS

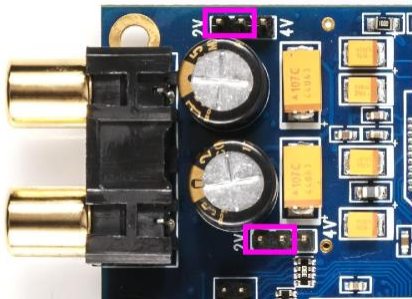
The 2x4 HD Kit board is exactly the same as the 2x4 HD “in a box”, but without the box. All functionality and I/O is identical. The board can be built into your own box and/or combined with other boards such as power



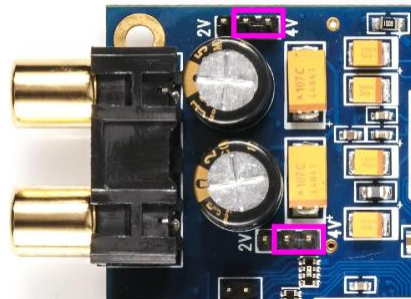
amplifiers. This annotated photograph shows the main features of the board:

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



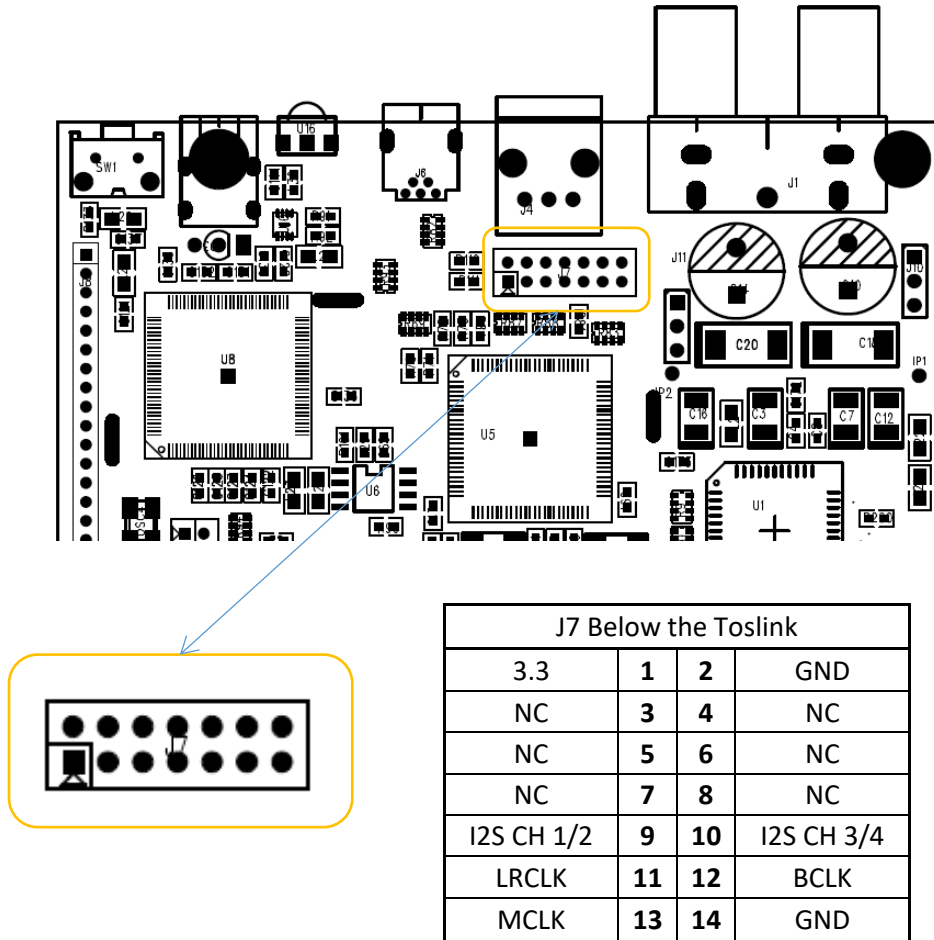
4 VRMS sensitivity



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

5.2 I2S HEADERS

Headers J7 and J8 are provided for connection of output circuitry via I2S. The pinouts are shown below.



5.2.1 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 2x4HD uses internally. This clock is always provided as an output by the 2x4HD, and connected circuitry can choose whether or not to use it.
- LRCLK** The frame synchronization clock, also known as the word clock. This clock is equal to the sampling frequency (F_s) of the audio signal.
- BCLK** The bit clock (also known as shift clock or system clock). This is always equal to $64 \times F_s$.

Table 1 summarizes the relation between the clocks for the 96 kHz plugin.

Table 1. I2S clock ratios

Plugin sample rate (LRCLK)	Master clock (MCLK)	Bit clock (BCLK)	MCLK/LRCLK	BCLK/LRCLK
96 kHz	24.576 MHz	6.144 MHz	256	64

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

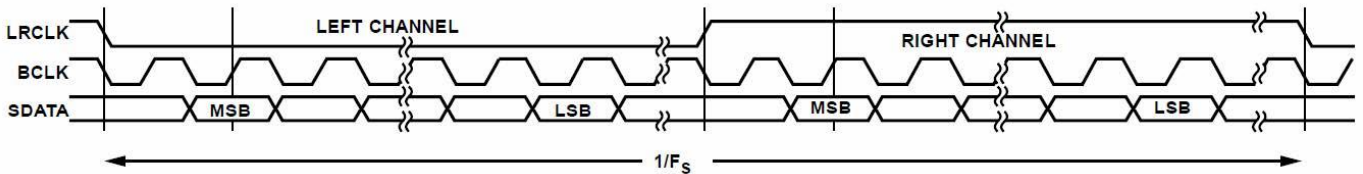


Figure 31. I²S Mode—16 Bits to 24 Bits per Channel

The 2x4HD board has two I2S output data lines, each carrying two channels of audio. Only output I2S is enabled in the plugin.

5.2.2 I2S details

The 2x4HD acts in **master** mode with respect to I2S clocking.

That is, the 2x4HD provides the I2S clocks and the connected devices are expected to receive data using those clocks. The clocks will match the sample rate of the plugin loaded onto the 2x4HD. See the [miniSHARC User Manual](#) for more information about I2S master and slave modes. 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 2x4HD's clock domain.

5.2.3 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. miniDSP cannot be held responsible for any damage you do to the board (outside normal warranty) or to your external equipment.

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

General I2S usage notes

- Unbuffered I2S lines must be kept short to ensure clock and data integrity.
- If driving longer lines, buffers may be required for the clock signals (MCLK, LRCLK, and BCLK).
- Observe correct grounding and shielding, and keep analog and digital grounds separated.
- Ensure that the clock ratios (as listed in Table 1) are compatible with connected circuits.
- All lines use a 3.3V logic level. Ensure that connected circuits use a compatible level (1.8V, for example, will not work).

6 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. 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:

The screenshot shows the miniDSP-2x4-HD software interface. The following components are highlighted with colored boxes and labeled:

- Main menu:** A green box highlights the top menu bar containing 'File', 'Restore', 'Help', and 'IR Remote'.
- Configuration selection:** A blue box highlights the 'Configuration Selection' area with buttons for 'Config 1', 'Config 2', 'Config 3', and 'Config 4'.
- Selection tab:** A red box highlights the 'Inputs & Routing' and 'Outputs' tabs.
- Main configuration window:** An orange box highlights the main control area, which is divided into 'Inputs' and 'Routing' sections.
- Master control:** A purple box highlights the top right corner, containing 'Connect', 'Mute', 'Master Volume', and an 'IP address' field with an 'Auto' checkbox.

The 'Inputs' section shows two input channels (Input1 and Input2) with volume sliders (ranging from -72 to 12 dB), gain controls (0.0), PEQ buttons, and Mute buttons. The 'Routing' section shows a matrix of buttons for routing Input1 and Input2 to Output1, Output2, Output3, and Output4. For example, Input1 is routed to Output1 and Output2 (both at 0dB), while Input2 is routed to Output3 and Output4 (both at 0dB).



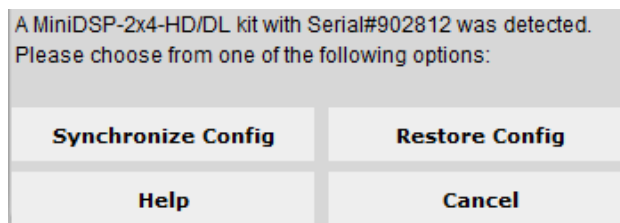
During *initial* configuration of the processor, it is strongly recommended that any connected amplification be powered off.

6.1 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

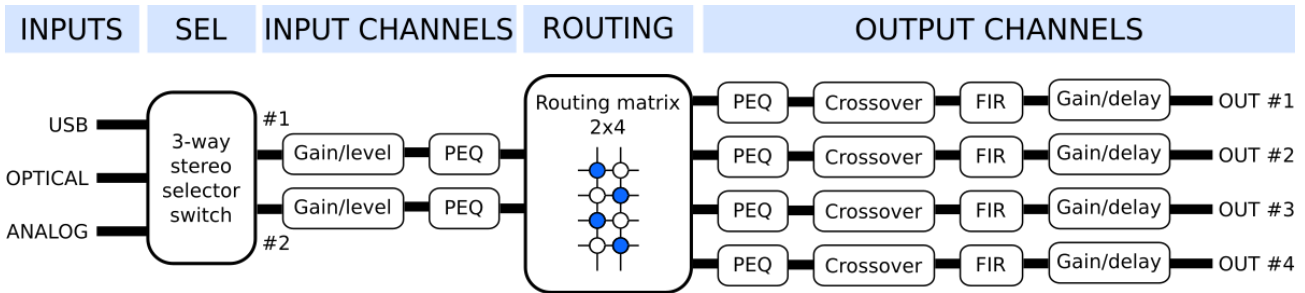
This option brings up a help screen explaining the options.

Cancel

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

6.2 SIGNAL FLOW AND PROCESSING OVERVIEW

The signal flow diagram of the 2x4 HD is shown in the diagram below.



One of the three stereo input sources is selected by the user and passed to the input channel processing blocks. The stereo analog input is converted to digital form. These blocks include input gain, level metering, and parametric EQ.

The processed input channels are fed into a 2-in 4-out routing matrix. The routing matrix enables the 2x4 HD to be used in a number of different applications. Section 6.4 provides a configuration guide for several common applications.

The outputs from the routing matrix are processed through a comprehensive set of DSP functions – crossover filters (high pass and low pass), parametric EQ, FIR filtering, and individual gain and delay adjustments. You can configure these as needed according to your particular application. Finally, the four output channels are converted to analog so they can be fed to your power amps and/or subwoofers.

6.3 KEY FEATURES

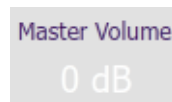
This section summarizes the key features of the plugin.

6.3.1 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:

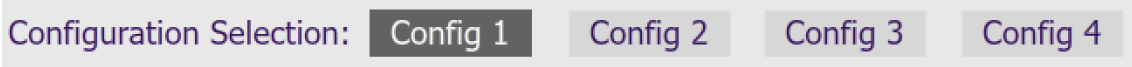


The **IP Address** and **Auto** fields are for use with networked control of the plugin using the miniDSP [WI-DG Wifi/Ethernet to USB bridge](#). See the [Wi-DG User Manual](#) for details.

6.3.2 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:

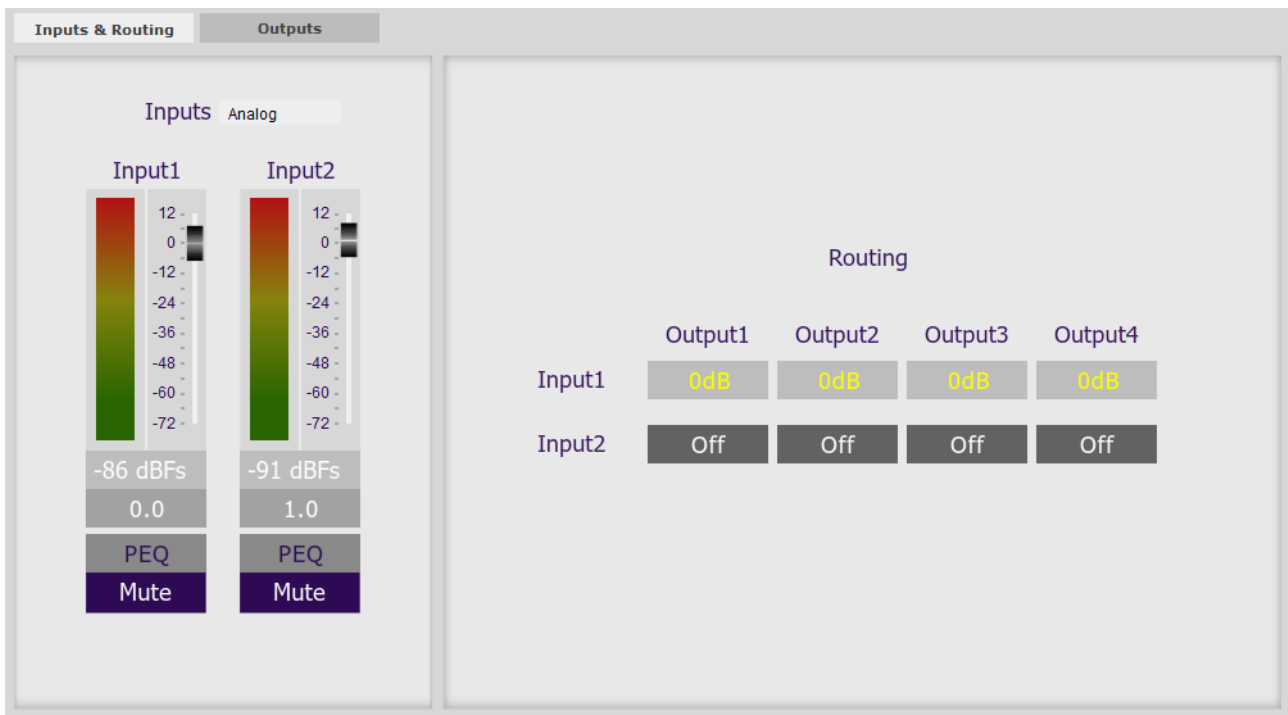


To switch to a different preset, just click on the desired button:



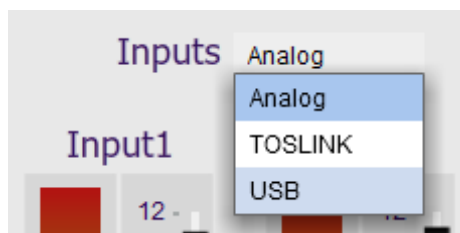
6.3.3 Inputs & Routing

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



6.3.4 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 46.)



6.3.5 Routing matrix

The routing matrix on the **Inputs & Routing** tab directs input channels (along the left) to output channels (along the top). To turn on routing for a cross-point, click on that cross-point. Both inputs can be mixed to each output if desired, and the mix level can be set individually for each input.

	Output1	Output2	Output3	Output4
Dirac 1	0dB	Off	Off	Off
Dirac 2	Off	0dB	Off	Off

6.3.6 Outputs

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

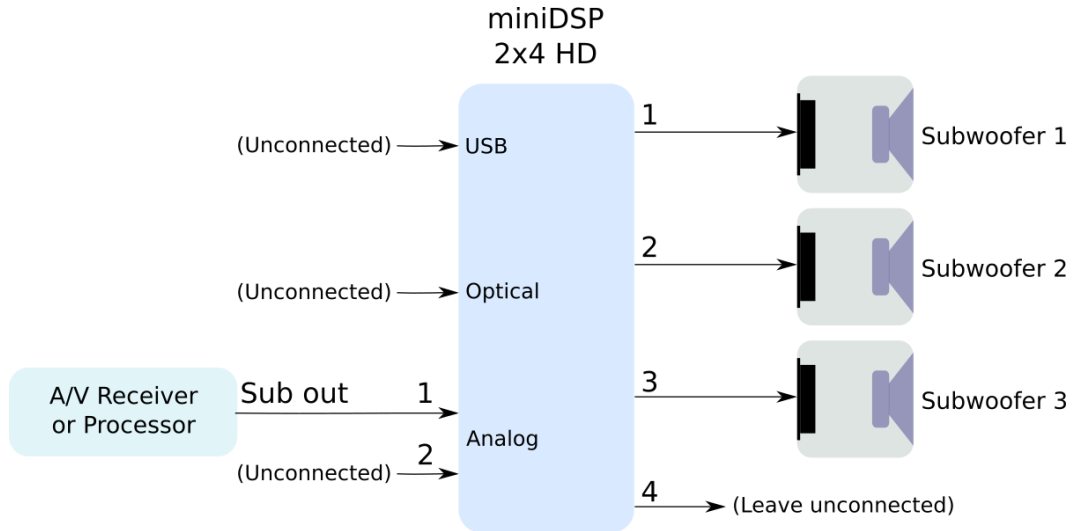
Each channel has an individual gain adjustment slider, and a graphical and numerical display of the current signal level on that channel. A comprehensive set of signal processing functions is accessed with the buttons PEQ (parametric EQ), Xover (crossover), Comp (compressor/limiter) and FIR (FIR filter).

6.4 PLUGIN CONFIGURATION GUIDE

This section provides a quick overview of a range of common configurations using the 2x4 HD.

6.4.1 Subwoofer control / multisub

The miniDSP 2x4 HD can be used to provide independent control of up to four subwoofers. Each subwoofer can have individual time delay, gain, and EQ applied as needed. Here is a typical connection diagram for a home theater system with three subwoofers:



Following is a brief summary of the key configuration settings needed. On the **Outputs** tab:

- Rename the output channels to match the number of subs you have i.e. “Sub 1,” “Sub 2,” etc.
- Mute any unused output channels.
- Check that the crossover filters of all output channels are **bypassed**:



HIGH PASS FILTER SETTINGS		LOW PASS FILTER SETTINGS	
Cut off frequency	1000	Cut off frequency	1000
Bypass filter	LR 48dB/oct	Bypass filter	LR 48dB/oct
Filter Type	BYPASSED	Filter Type	BYPASSED

(Normally, upstream equipment like the A/V receiver or processor implements lowpass crossover filters as part of its bass management processing. In some cases, such as when using a 2x4 HD to add multisub capability to a stereo system, you may need to use lowpass filters in the 2x4 HD.)

On the **Inputs & Routing** tab, set the routing matrix like this (for the three-sub HT example shown above):

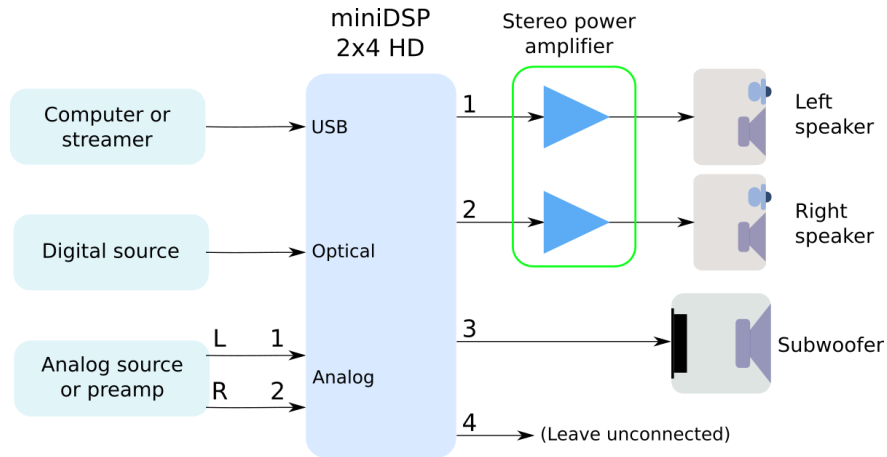
	Sub 1	Sub 2	Sub 3	Unused
Input1	0dB	0dB	0dB	Off
Input2	Off	Off	Off	Off

After you have set up the plugin, be sure to save your configuration to a file. For detailed information on tuning multiple subwoofers, see this application note on our website:

- [Tuning multiple subwoofers with miniDSP](#)

6.4.2 Subwoofer integration (2.1 system)

The miniDSP 2x4 HD is ideal for integrating a subwoofer into a stereo system. This diagram illustrates the typical connections:

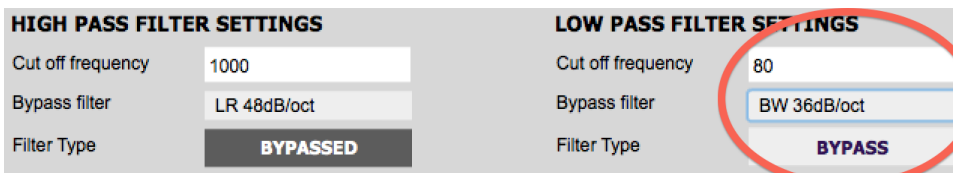


Following is a brief summary of the key configuration settings needed. On the **Outputs** tab:

- Rename the output channels (from left to right) to “Left Sp”, “Right Sp”, “Sub” and “Unused”.
- Mute channel 4 (“Unused”).
- Set a high pass crossover filter on channels 1 and 2 (“Left Sp” and “Right Sp”). For example:



- Set a low pass crossover filter on channel 3 (“Sub”). For example:



On the **Inputs and Routing** tab, set the routing matrix like this:

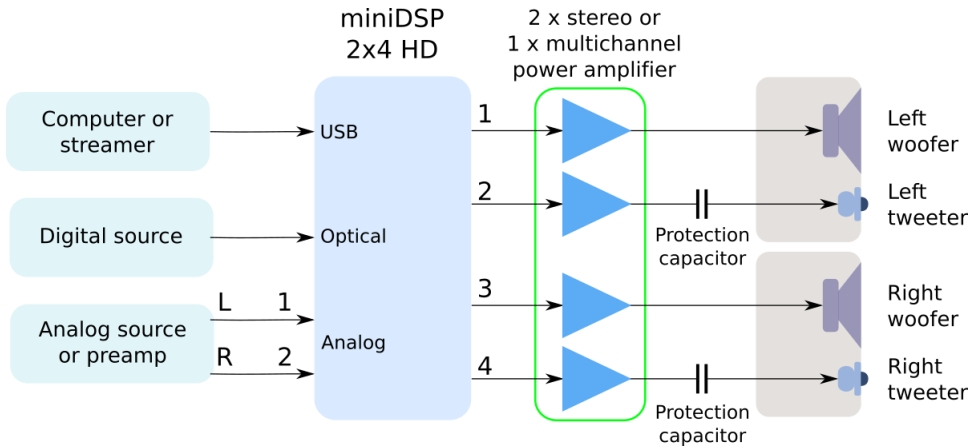
	Left Sp	Right Sp	Sub	Unused
Input1	0dB	Off	0dB	Off
Input2	Off	0dB	0dB	Off

We recommend that you do acoustic measurements to check and fine-tune your subwoofer crossover settings. For more detail, refer to the app note linked above. After you have set up the plugin, be sure to save your configuration to a file. For more information, see this application note on our website:

- [Subwoofer integration with the miniDSP 2x4 HD](#)

6.4.3 Two-way active speaker

The 2x4 HD is a perfect solution for implementing a two-way active speaker. This diagram illustrates the connections:



Following is a brief summary of the key configuration settings needed. On the **Outputs** tab, rename the output channels to “Left W”, “Left Tw”, “Right W” and “Right Tw”.

On the **Inputs & Routing** tab, set the routing matrix like this:

	Left W	Left Tw	Right W	Right Tw
Input1	0dB	0dB	Off	Off
Input2	Off	Off	0dB	0dB

On the **Outputs** tab again:

- Use Room EQ Wizard or a similar program to measure each individual woofer and tweeter and to equalize their response flat with the PEQ blocks. (See the app note linked above for details.)
- Set up your high pass and low pass crossover filters. This is a typical low pass setting for the woofers:

HIGH PASS FILTER SETTINGS		LOW PASS FILTER SETTINGS	
Cut off frequency	80	Cut off frequency	3000
Bypass filter	BW 24dB/oct	Bypass filter	LR 24dB/oct
Filter Type	BYPASSED	Filter Type	BYPASS

This is a typical high pass setting for the tweeters:

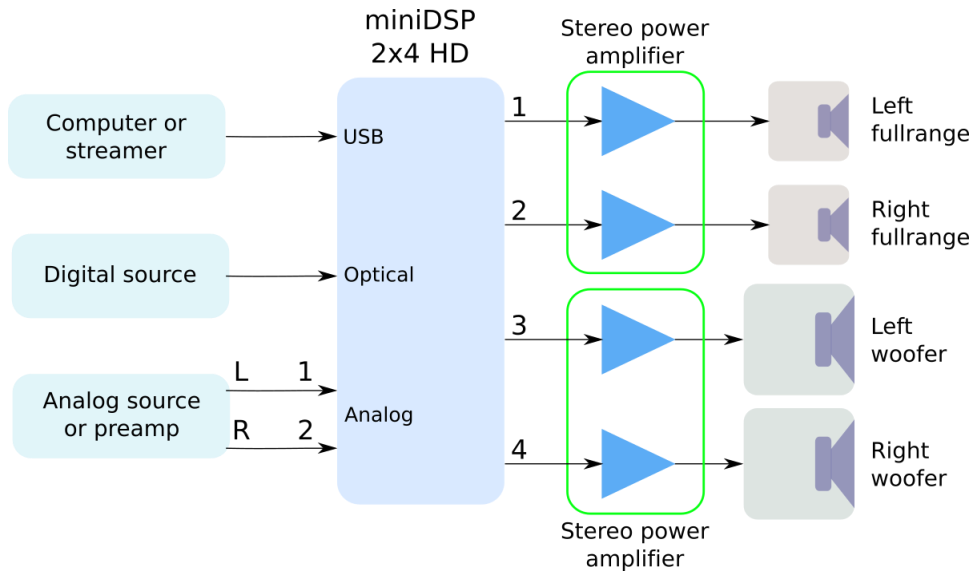
HIGH PASS FILTER SETTINGS		LOW PASS FILTER SETTINGS	
Cut off frequency	3000	Cut off frequency	5000
Bypass filter	LR 24dB/oct	Bypass filter	LR 48dB/oct
Filter Type	BYPASS	Filter Type	BYPASSED

After you have set up the plugin, be sure to save your configuration to a file. For more detailed information on implementing a two-way speaker, see this application note on our website:

- [Designing a 2-way active speaker with the miniDSP 2x4 HD](#)

6.4.4 Stereo supporting woofers/FAST

In a configuration with two full range speakers supplemented by supporting woofers (sometimes known as “FAST”), the plugin is set up to split the frequency range to the woofer and full range speakers. This diagram illustrates the connections:

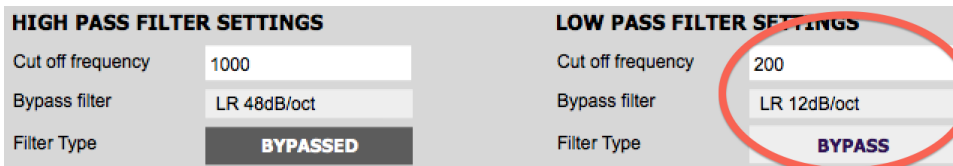


Following is a brief summary of the key configuration settings needed. On the **Outputs** tab:

- Rename the output channels (from left to right) to “Left FR”, “Right FR”, “Left W” and “Right W”.
- Set a high pass crossover filter on channels 1 and 2 (“Left FR” and “Right FR”). For example:



- Set a low pass crossover filter on channels 3 and 4 (“Left W” and “Right W”). For example:



On the **Inputs & Routing** tab, set the routing matrix like this:

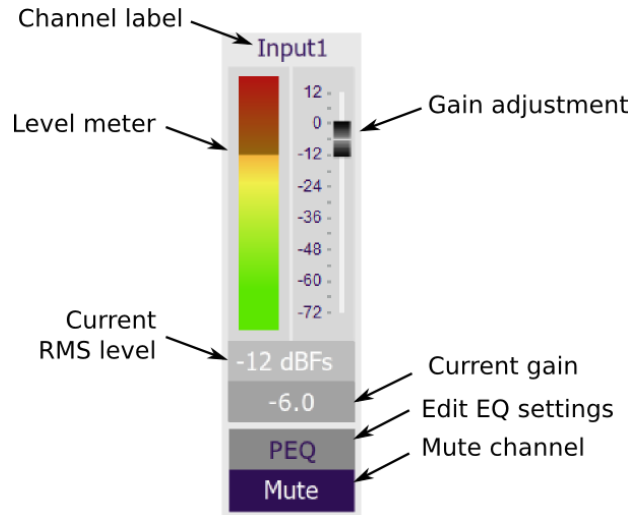
	Left FR	Right FR	Left W	Right W
Input1	0dB	Off	0dB	Off
Input2	Off	0dB	Off	0dB

We recommend that you take acoustic measurements to check and fine-tune your crossover settings. After you have set up the plugin, be sure to save your configuration to a file.

7 PLUGIN REFERENCE

This section provides full details on each of the plugin processing blocks.

7.1 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 31.

Mute

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



7.2 ROUTING

The routing matrix is used to direct input channels (along the left) to output channels (along the top). To turn on routing for a cross-point, click on that cross-point. The display will change from “Off” to show the mix level (0 dB by default).

Below is the default routing matrix. The input and output channel names displayed on the matrix match those set on the input and output control strips:

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

Changing the channel labels on the control strips changes them on the routing matrix, as an aid to ensuring correct configuration. For example:

	Left Sp	Right Sp	Sub 1	Sub 2
In Left	0dB	Off	0dB	0dB
In Right	Off	0dB	0dB	0dB

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 cross-point and a gain control will appear. Adjust the gain with the slider, or by typing in the value directly, then click **close**:

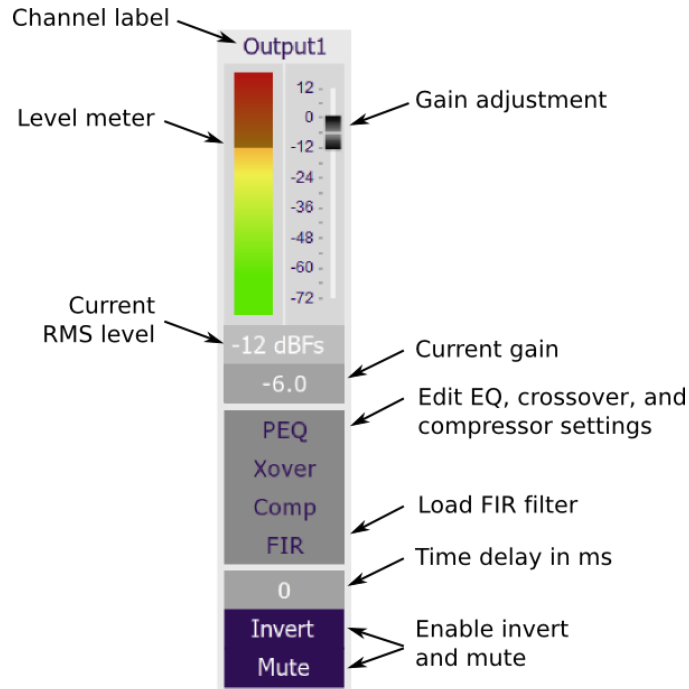
	Output1	Output2	Output3	Output4
Input1	0dB	-3.1	Off	Off
Input2	Off	Off	0dB	0dB

7.3 OUTPUT TAB

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

7.3.1 Channel strip layout

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



7.3.2 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.

7.3.3 Level meter and gain control

Level meter, current RMS level

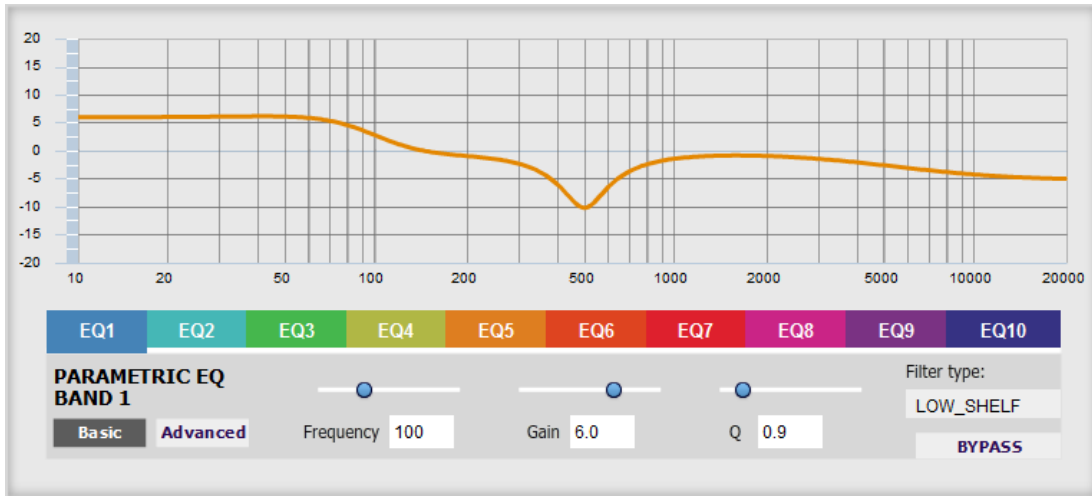
Displays the current signal level in real time. (The plugin must be in online mode to display signal levels.)

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.)

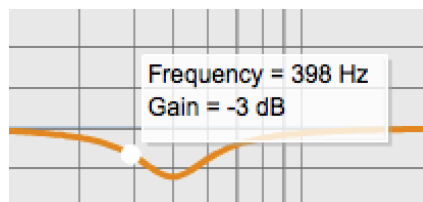
7.3.4 Parametric EQ

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 37.

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.

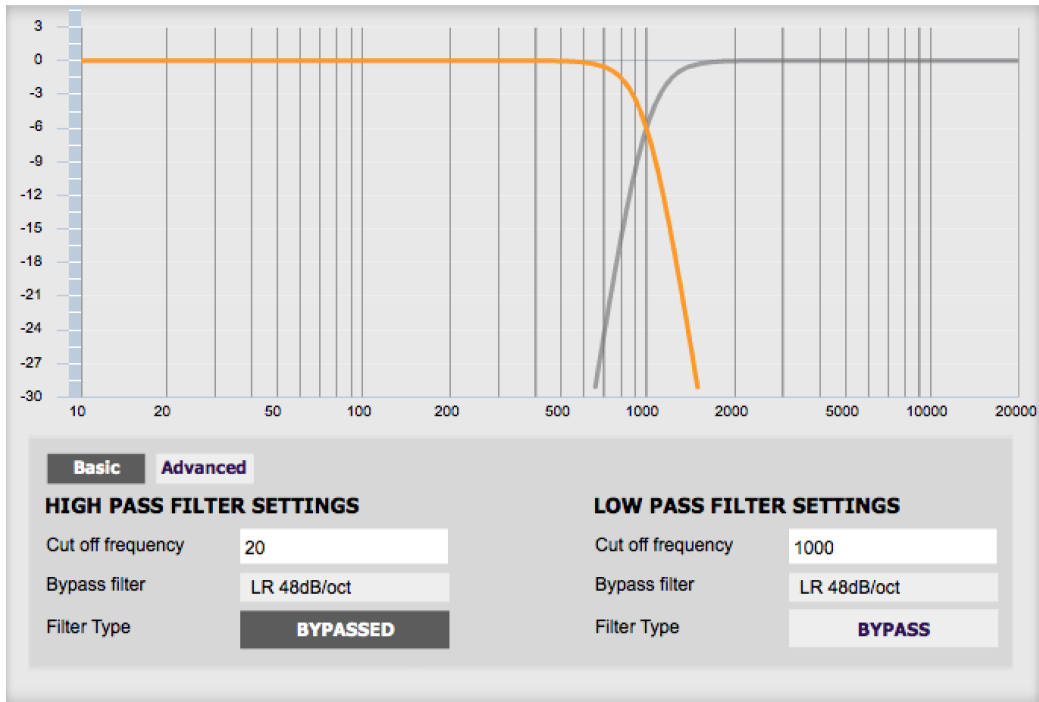
Bypass

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.



7.3.5 Crossover

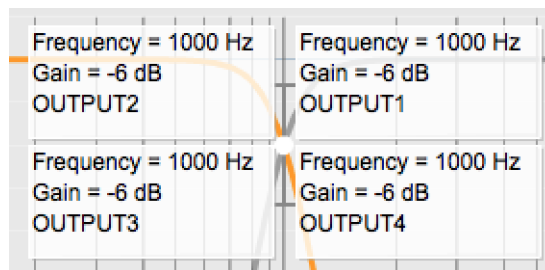
Each output channel has independent high pass and low pass filters. Click on the **Xover** button to open the crossover settings window:



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-exursion.

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 37.

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".



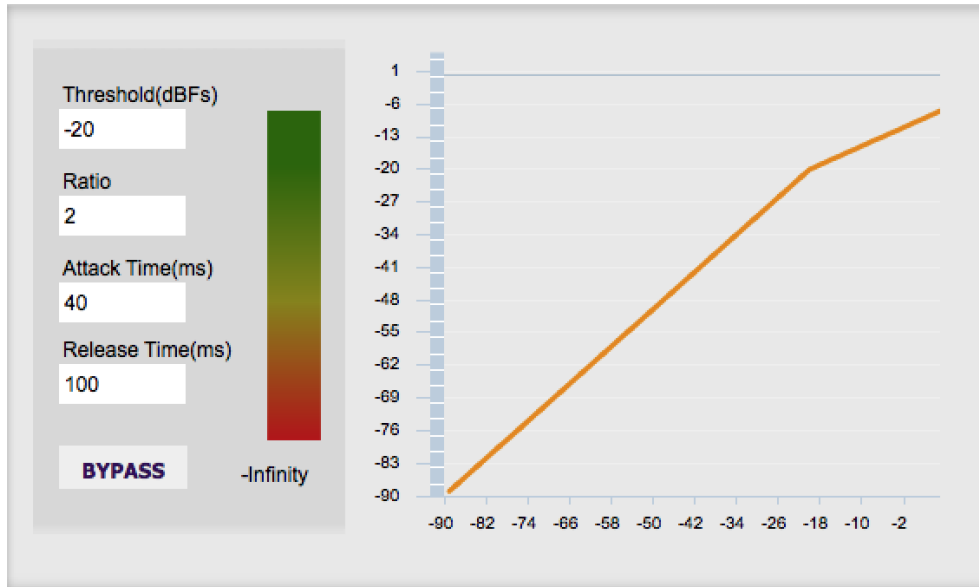
Each channel can be linked to one other channel. When a channel is linked to another, the crossover settings of that channel are mirrored to the other. Typically, the corresponding drivers on the left and right channels are linked: left and right tweeter, left and right woofer, and so on. To link a channel,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



7.3.6 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:



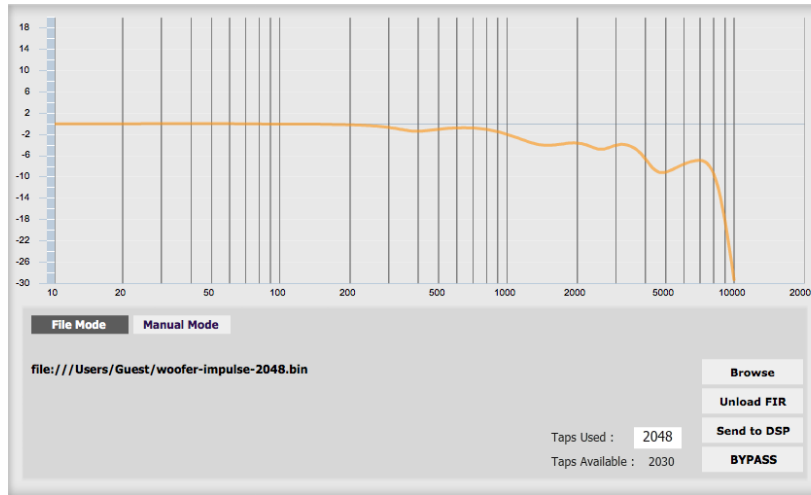
(Note that the compressor algorithm is bypassed by default, so click on the **Bypass** button to see the curve as shown here.)

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](#).

7.3.7 FIR

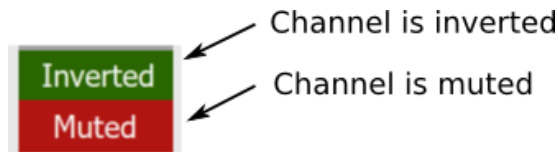
Each output channel has an FIR filter bank with a variable number of taps. Click on the **FIR** button to open the FIR filter settings window:



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 40.

7.3.8 Invert and mute

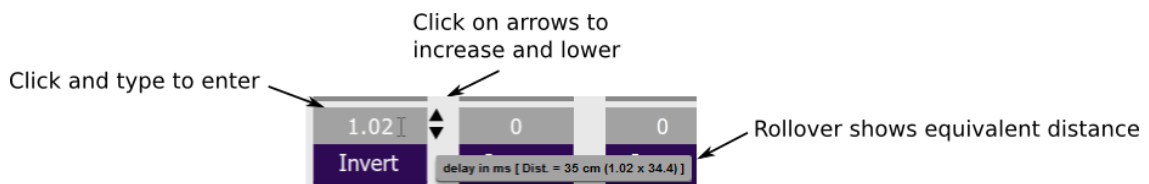
Each channel can be inverted in polarity, and individually muted. When either of these options is selected, the visual indicator on the button is "lit":



7.3.9 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 (0.01 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.



7.4 CUSTOM BIQUAD PROGRAMMING

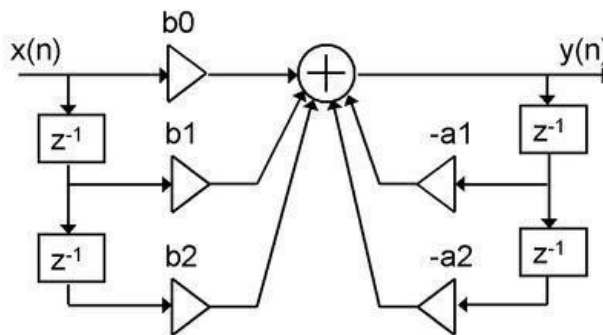
Custom biquad programming is available in the PEQ and Crossover blocks. Its purpose is to allow you to directly provide the low-level parameters aka *biquad coefficients* that control the digital filters of the processor, thus providing an almost infinite degree of flexibility.

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.

7.4.1 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:

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



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*.

7.4.2 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.

EQ1	EQ2	EQ3	EQ4	EQ5	EQ6	EQ7	EQ8	EQ9	EQ10
PARAMETRIC EQ BAND 1 <input type="button" value="Basic"/> <input checked="" type="button" value="Advanced"/>		<pre>biquad1, b0=0.9997292058283046, b1=-1.9990173425193771,</pre>							<input type="button" value="CLEAR"/> <input type="button" value="IMPORT"/> <input type="button" value="PROCESS"/> <input type="button" value="BYPASS"/>

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:

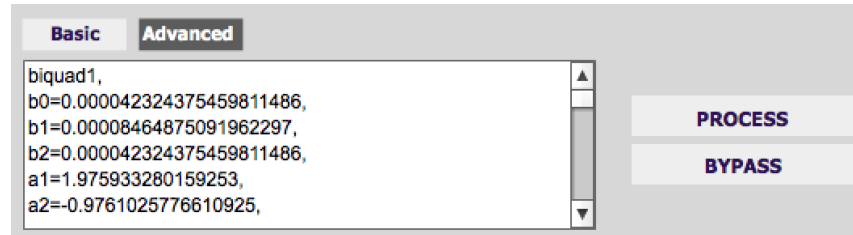
```
biquad1,
b0=0.998191200483864,
b1=-1.9950521500467384,
b2=0.996920046761057,
a1=1.9950521500467384,
a2=-0.9951112472449212,
biquad2,
b0=0.999640139948623,
b1=-1.9981670485581222,
b2=0.9985489719847982,
a1=1.9981670485581222,
a2=-0.9981891119334211,
biquad3,
...
biquad4,
...
biquad10,
b0=1.0010192374642126,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



7.4.3 Biquad design software

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

7.4.3.1 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

7.4.3.2 Room EQ Wizard (REW)

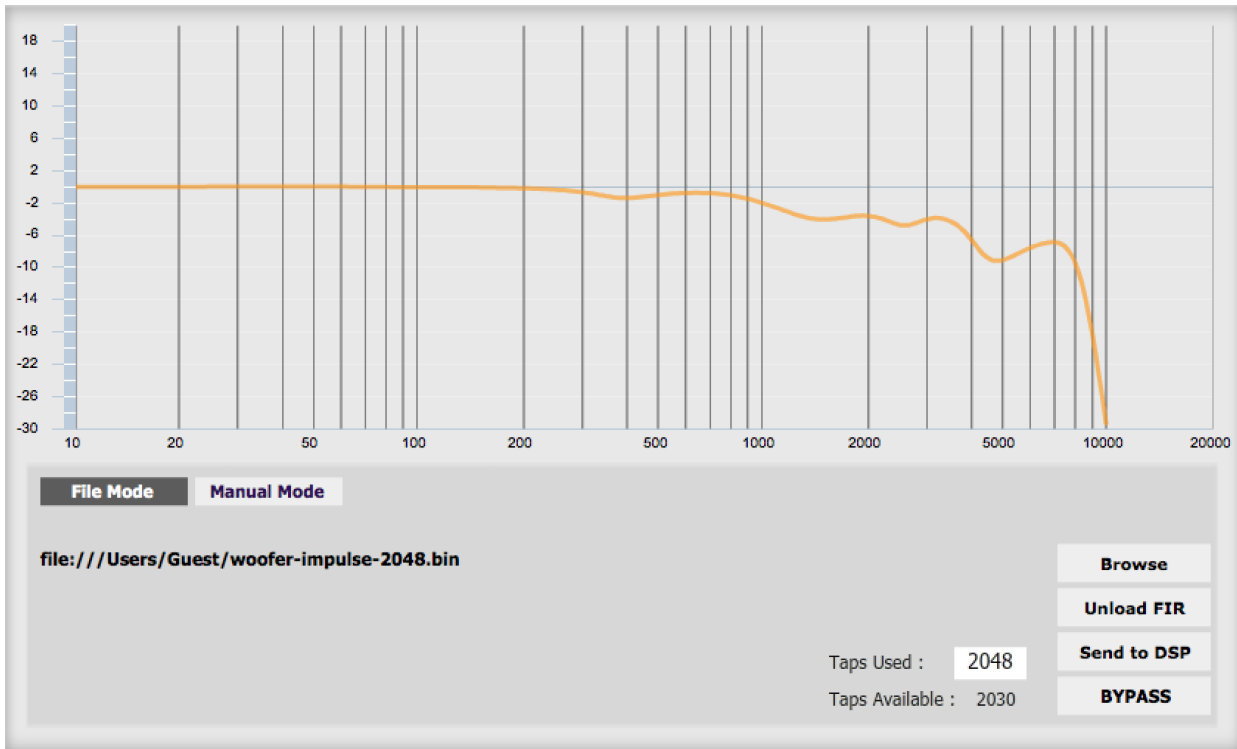
Room EQ Wizard is a free acoustic measurement and analysis tool, available for Windows, Mac and Linux platforms. It includes the ability to automatically generate a bank of parametric EQ biquads based on a measurement. These coefficients can be saved to a file from REW and loaded directly into a PEQ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

7.5 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

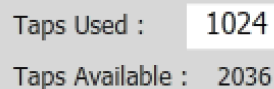
In File Mode, the window displays the **Browse** and **Unload** FIR buttons as shown above. In Manual Mode, the display changes to allow direct text entry of the FIR filter coefficients, as shown below. The coefficients can be pasted into the window from a text editor.



7.5.1 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

7.5.2 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.

7.5.3 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,
```

7.5.4 Loading filter coefficients

In **File Mode**:

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

In **Manual Mode**:

8. Cut and paste the coefficients from the text output of the design program.
9. Press the **Process** button.
10. Confirm that the frequency response graph 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 **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.

7.6 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.

7.6.1 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.

7.6.2 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.

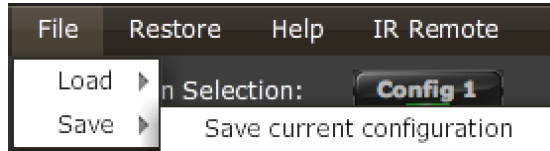
The plugin is in offline mode

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.

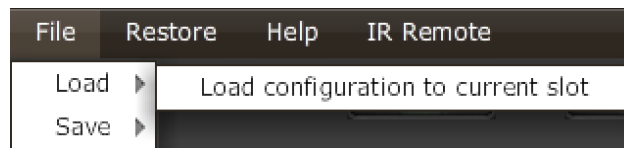
7.6.3 Saving and loading configurations

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.

7.6.4 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.

7.7 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

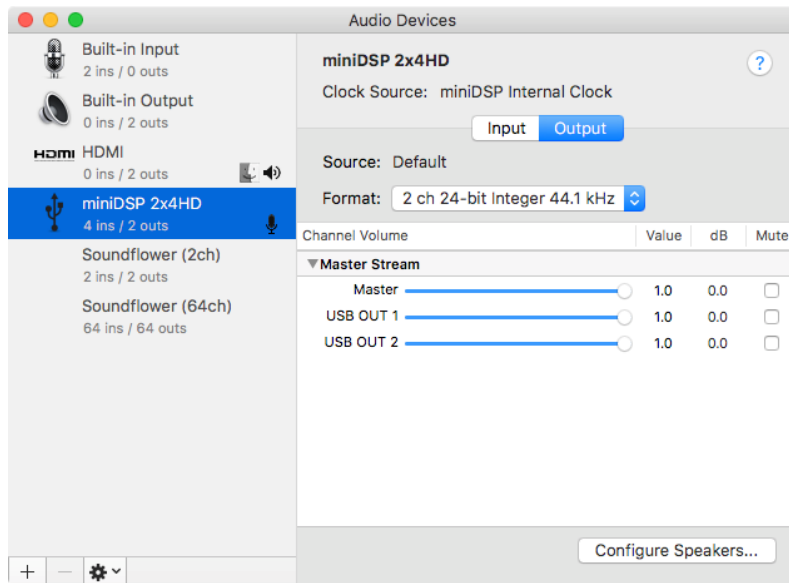
The Space bar toggles buttons that have two states, such as **Bypass**, **Invert**, and **Mute**, if they have the focus.

8 USB AUDIO

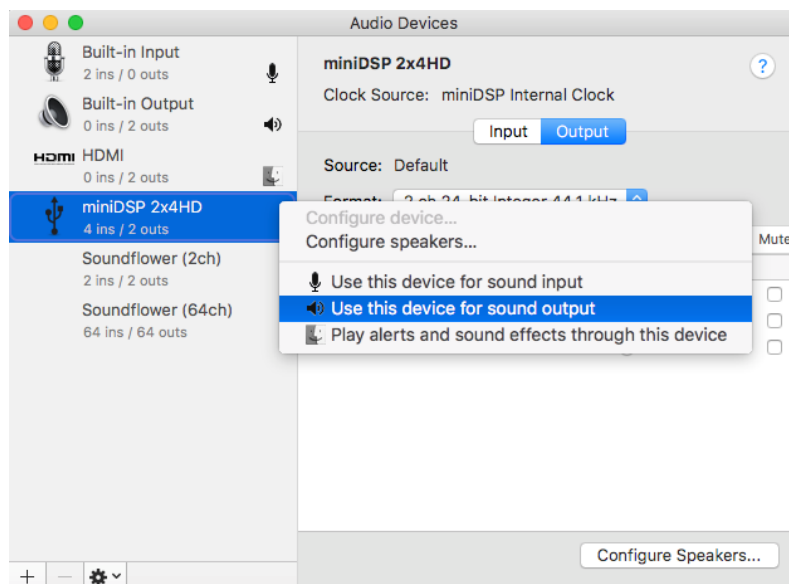
The miniDSP 2x4 HD accepts stereo PCM audio at sample rates of 44.1, 48, 88.2, 96, 176.4, and 192 kHz on its USB audio input. The same USB connector is used both for streaming audio and configuration.

8.1 MAC OS X

Open the program **Audio MIDI Setup** (in **Applications->Utilities**). The 2x4 HD will appear as “miniDSP 2x4HD” 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:

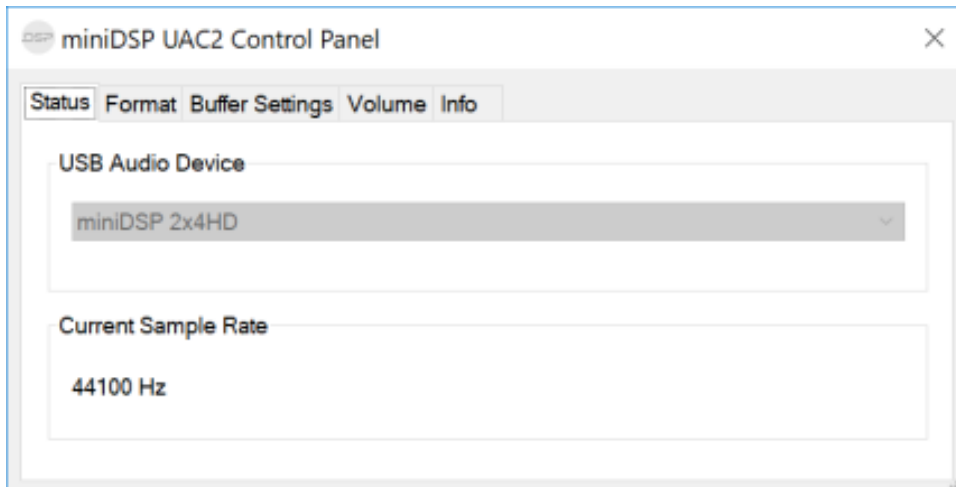


To set the 2x4 HD as the default audio output device, right-click and select “Use this device for sound output”.



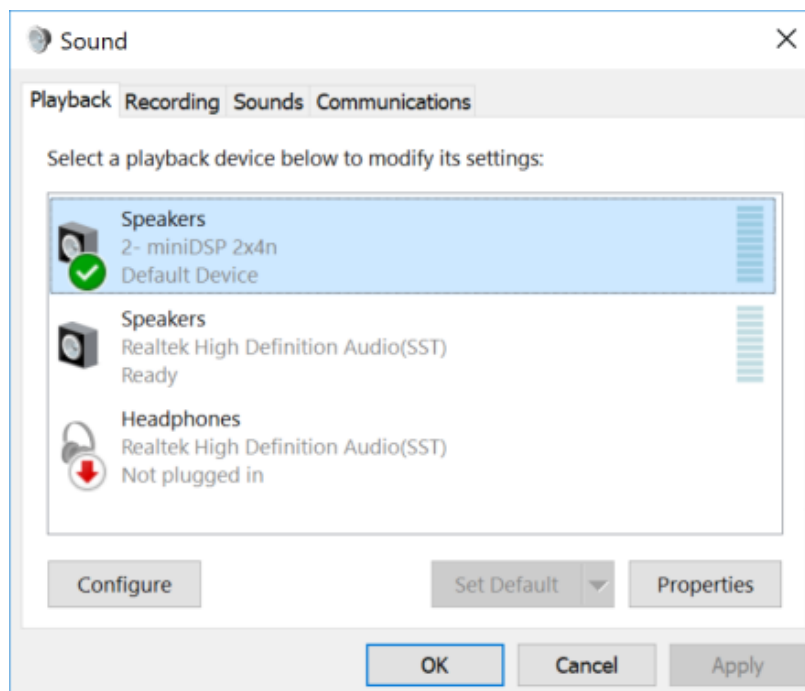
8.2 WINDOWS

Open the UAC Control Panel from the Windows Start menu. This control panel allows you to set a number of options, such as word length (Format tab) and buffer size (Buffer tab). We recommend that you leave these settings at their defaults.



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

To set the 2x4 HD as the default output device, open the Windows Control Panel and navigate to the Audio Devices section. On the Playback tab, select “miniDSP 2x4n” and click on the “Set Default” button. Individual audio playback programs may allow the 2x4 HD to be selected independently of the system default.



9 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 2x4 HD:

Source

Cycles through the input sources.

1, 2, 3, 4

Switches to the selected preset.

[Bell]

Has no function with the 2x4 HD.

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

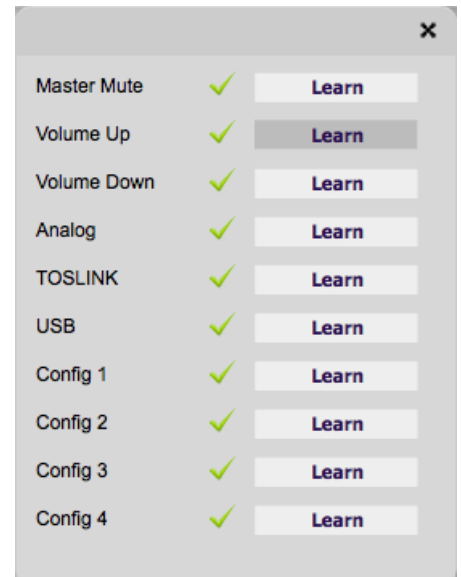
Mutes and unmutes audio output.



The 2x4 HD 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.

10 ADDITIONAL INFORMATION

10.1 SPECIFICATIONS

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

10.2 FIRMWARE UPGRADE

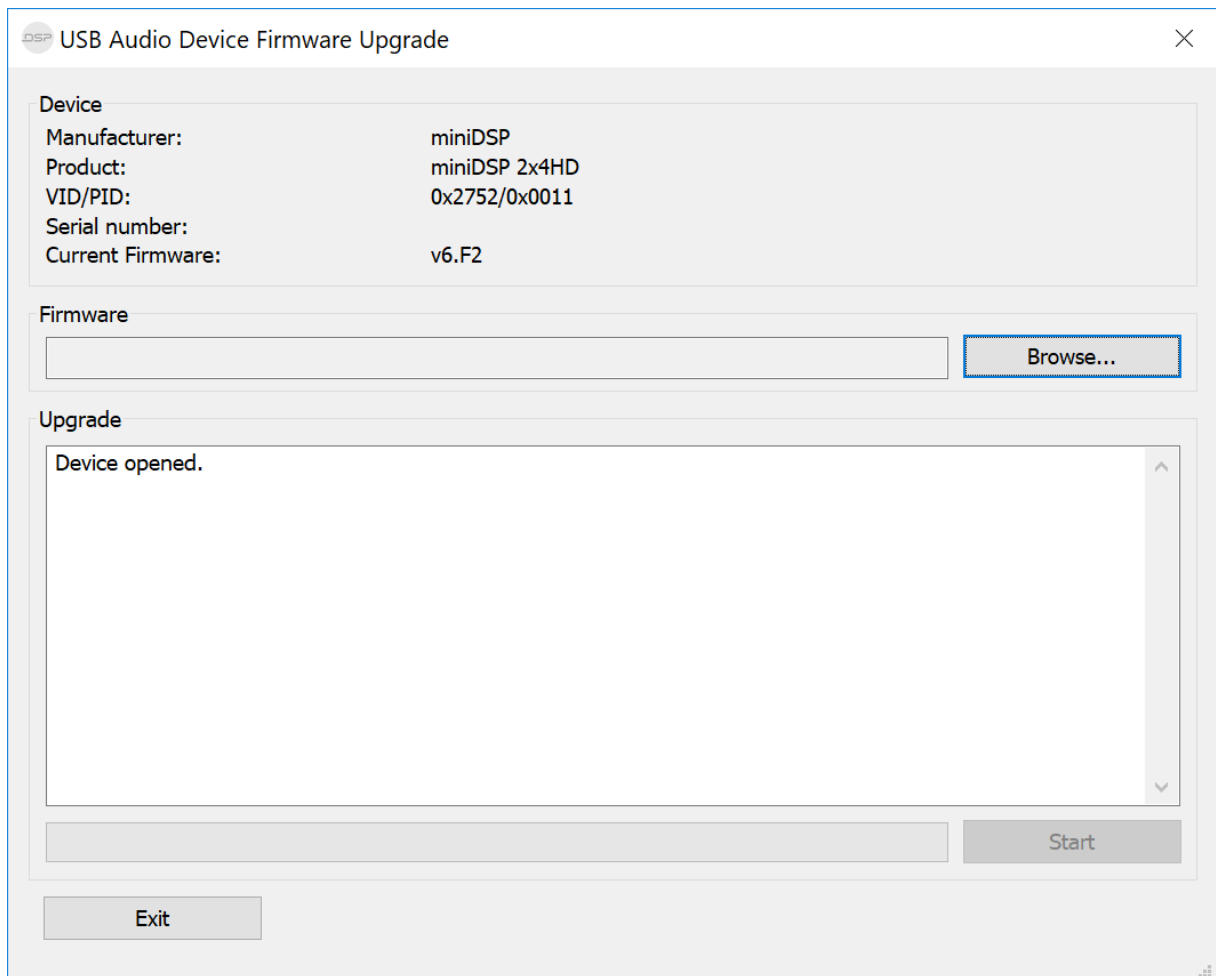
miniDSP may occasionally provide an upgrade to the 2x4 HD MCU firmware to enable new features. To upgrade the MCU firmware, first download the latest version of the 2x4 HD software package from the **User Downloads** section of the miniDSP website, then extract it on your computer (on Windows, right-click and select “Extract All...”; on Mac, double-click).



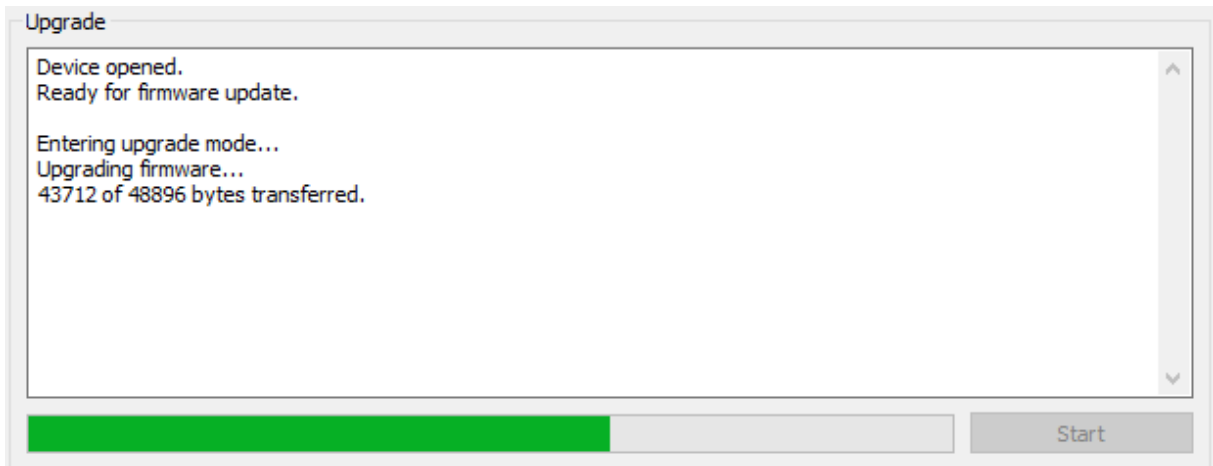
DO NOT DISCONNECT THE USB CABLE OR POWER FROM THE 2x4 HD WHILE FIRMWARE UPGRADE IS IN PROGRESS. DOING SO MAY “BRICK” YOUR 2x4 HD.

10.2.1 Windows

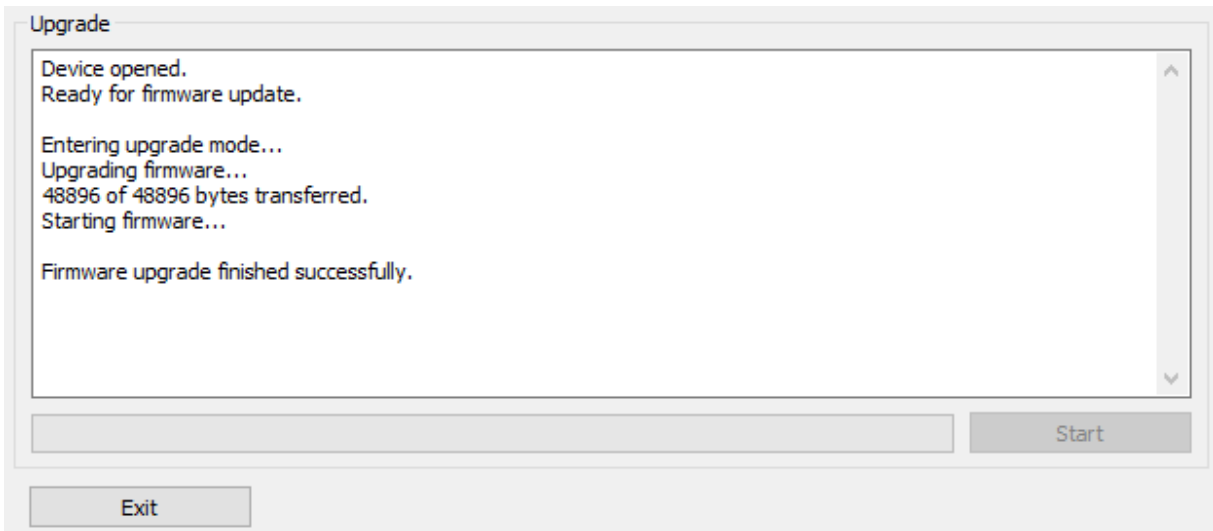
1. Connect the 2x4 HD (or Kit) to your computer via USB (if not already connected) and power it on.
2. Navigate to the **XMOS_Firmware\Firmware_Upgrade_Tools\Windows\miniDSPUAC2Dfu** folder of the software download.
3. Double-click on the **miniDSPUAC2Dfu.exe** program to run it:



- 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 “miniDSP2x4HD_XMOS_v1.7.bin.” (The version number may be different.)
- Click on the **Start** button.
- You will get a progress bar as upgrade proceeds:



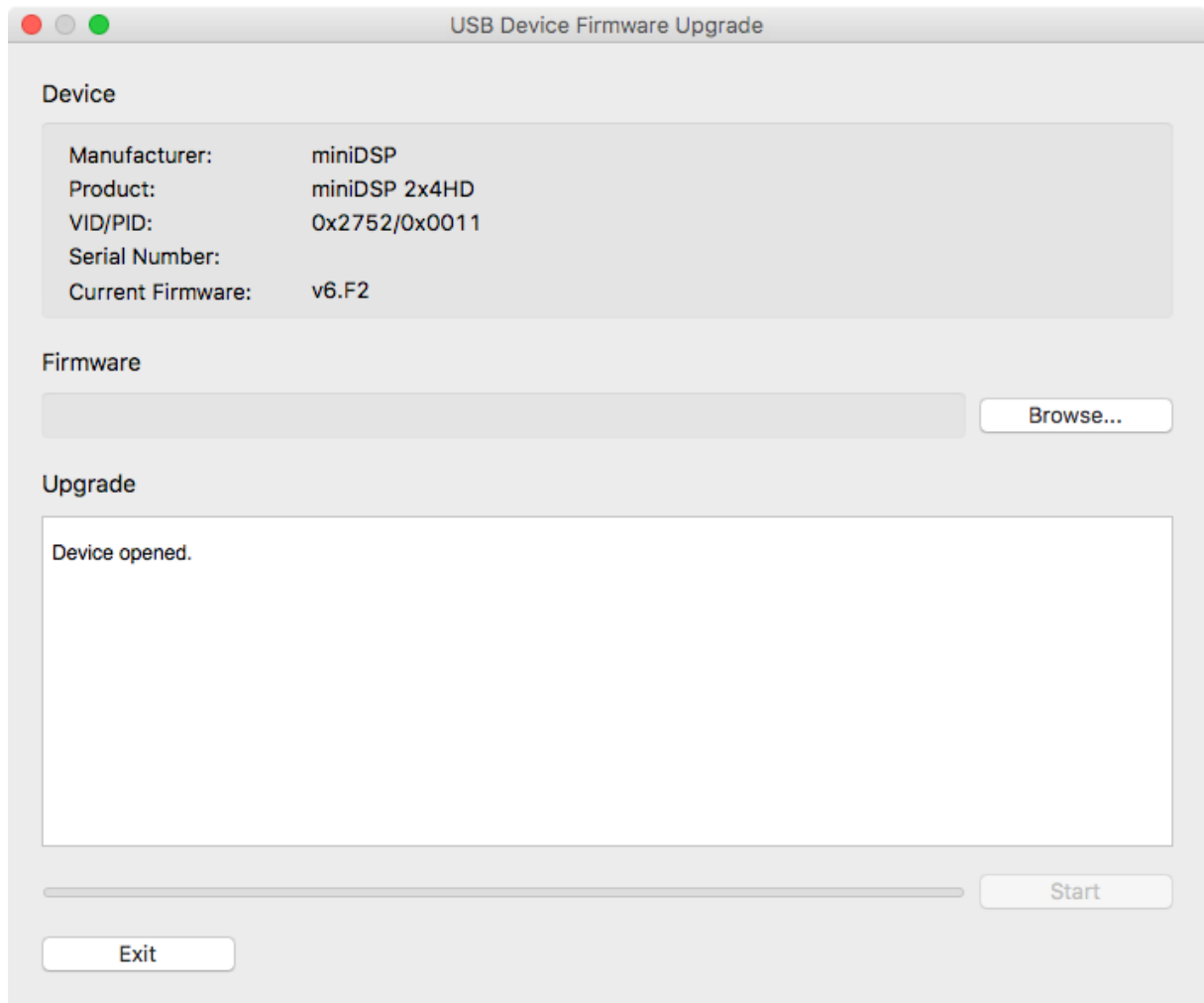
- Once the firmware upgrade completes, you will see a message that the upgrade completed successfully:



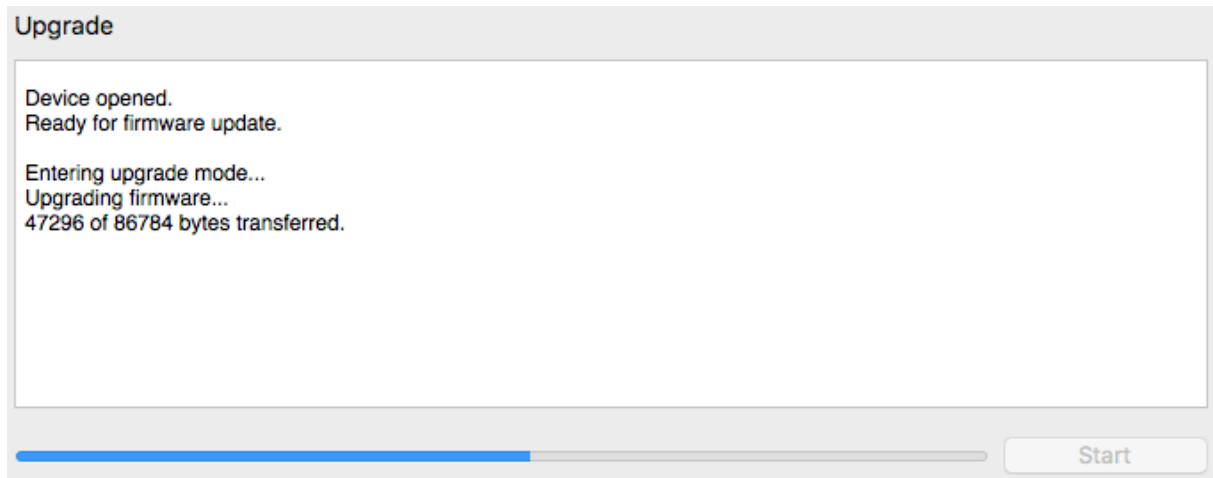
- Click on **Exit**.
- That's it! You're done. You can now use your 2x4 HD with the new functionality.

10.2.2 macOS / OS X

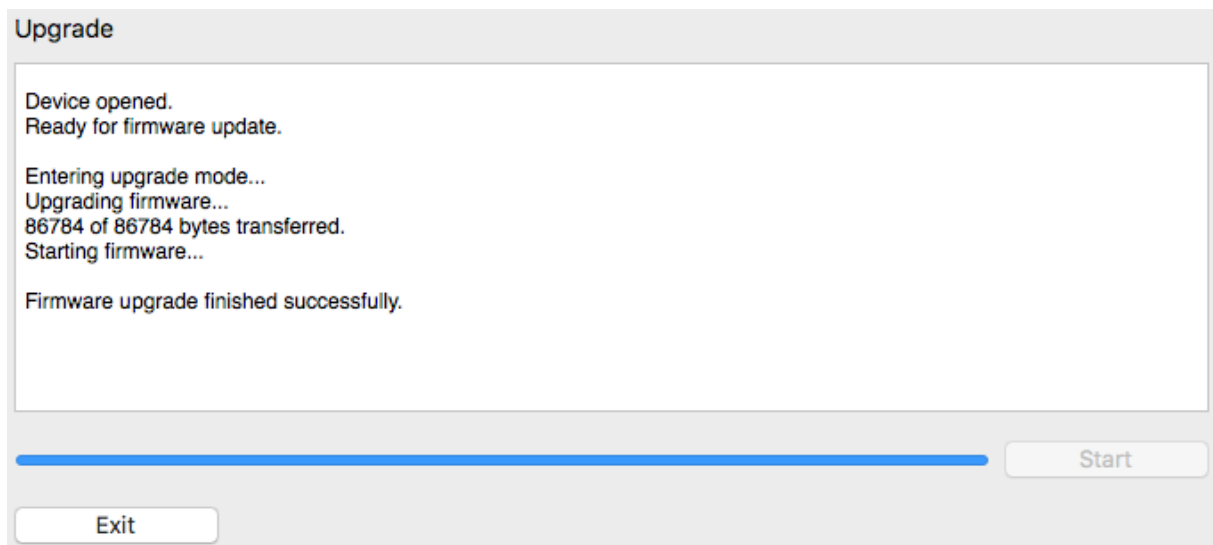
1. Connect the 2x4 HD (or Kit) to your computer via USB (if not already connected) and power it on.
2. Navigate to the **XMOS_Firmware/Firmware_Upgrade_Tools/Mac** folder of the software download.
3. Double-click on the **DFU Utility.app** program to run it:



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 “miniDSP2x4HD_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:



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



successfully:

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

10.3 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 Software Installation).
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 http://get.adobe.com/air/ .
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 routing matrix 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	a. Check the cabling from the processor to your amplifiers. b. Check that you have correctly set up the routing matrix to send the correct inputs to the correct outputs.

7	Cannot reload a configuration	a. Confirm the file format of your file (.xml). b. Confirm the version of the file.
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10.4 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, 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.