



# 10x10 HD

10-IN 10-OUT

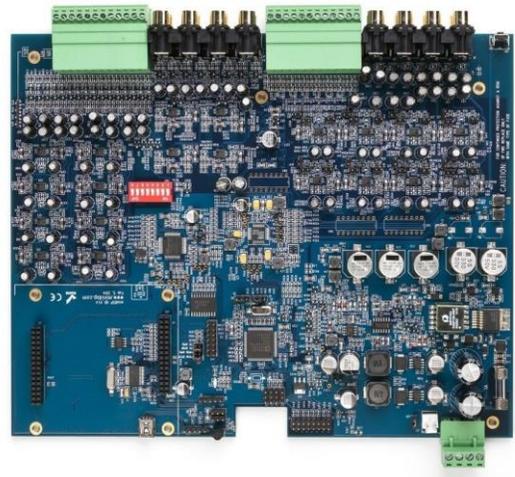
MULTICHANNEL AUDIO PROCESSOR



# 8x8 KIT

8-IN 8-OUT MULTICHANNEL

AUDIO PROCESSOR KIT BOARD



# User Manual



## Revision history

Revision	Description	Date
V1.0	Initial revision	28-06-2012
V1.1	Updated section on volume control	03-08-2012
V2.0	New version combining boxed version with kit version	30 May 2015
V2.1	New product photographs	28 Sept 2015

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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 your 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
- 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 10.4 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 10x10 Hd 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 10x10 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.

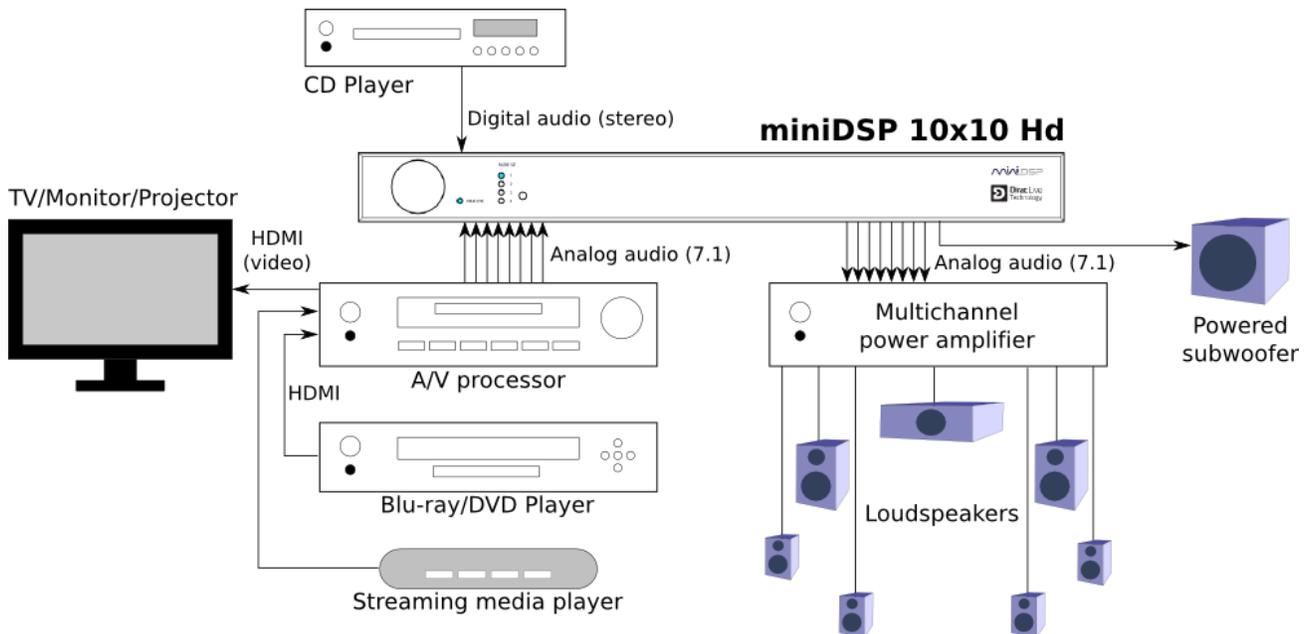
# 1 PRODUCT OVERVIEW

Thank you for choosing the miniDSP 10x10 Hd audio processor or 8x8 Kit board. The miniDSP 10x10 Hd and 8x8 Kit board are a powerful and flexible multichannel digital audio processor that is ideal for applications such as home theater and surround-sound monitoring. They feature:

- 8 analog input channels and 8 analog output channels, both balanced and unbalanced (RCA)
- Stereo AES/EBU, S/PDIF and TOSLINK digital inputs and outputs (DIGI-FP add-on required for 8x8 Kit)
- Volume control by front panel or infrared remote (VOL-FP add-on required for 8x8 Kit)
- 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
- Matrix mixer for flexible channel assignment and configuration
- Compressor/limiter and adjustable time delay on all output channels
- Real-time configuration of all processing parameters
- Four on-board presets, selectable from front panel or by infrared remote

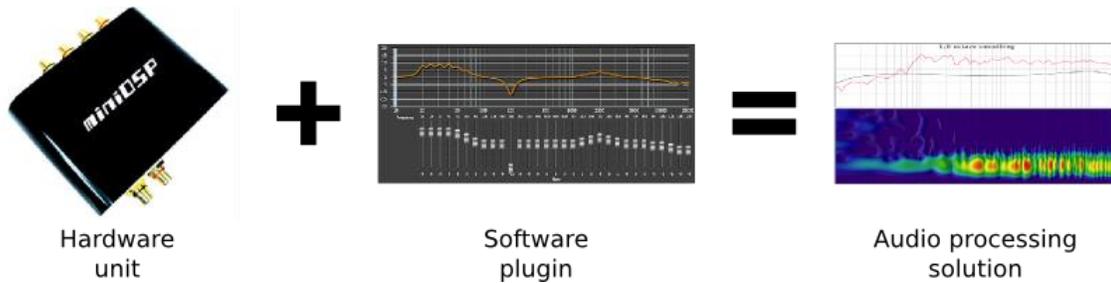
## 1.1 TYPICAL USAGE

In its most typical application, the 10x10 Hd connects to a multichannel analog source and directly drives power amplification. Its three digital inputs can be connected to stereo digital sources and mixed with the analog inputs. It is commonly used for equalization of multichannel audio for home theater and surround sound monitoring, and can also be used to implement multichannel active crossovers.



## 1.2 THE MINIDSP CONCEPT

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



### Hardware unit

In this case, the hardware unit is the miniDSP 10x10 Hd or 8x8 Kit board. See [Choosing the right miniDSP](#) for more information on miniDSP hardware.

### Software plugin

The software plugin is installed on your PC or Mac, and determines the processing that the DSP will perform. It provides a friendly user interface, and downloads instructions into the miniDSP hardware unit that tell it how to process the audio signal. See [Choosing a plugin](#) for more information on plugins.

## 1.3 CHOOSING THE RIGHT MINIDSP

The **miniDSP 10x10 Hd** is a self-contained “miniDSP in a box”. It has eight analog input channels and eight analog output channels, as well as two digital input and output channels with AES/EBU, S/PDIF, and TOSLINK connectors.



The **8x8 Kit** consists of just the audio processing circuit board without the external chassis or front and rear panel boards. The board can be integrated into any build that requires flexible yet cost-effective multichannel audio processing. Volume control and remote control can be added with the optional VOL-FP board, and digital input and output with the optional DIGI-FP board.



miniDSP make a large and growing number of hardware platforms for particular needs. If the miniDSP 10x10 Hd or 8x8 Kit is not the most suitable hardware, then other options are available:

#### miniDSP 4x10 Hd and 2x8 Kit

This platform is based on the same circuit board, but is populated for only two analog input channels. It is ideal for stereo processing applications such as room EQ and active crossovers.

#### Other platforms

miniDSP makes many other boxed and kit hardware units with analog, stereo digital, and multichannel digital (HDMI) connections. Some units feature inbuilt Class D amplification. See the [full range of miniDSP products](#).

## 1.4 CHOOSING A PLUGIN

The plugin that is usually used with the 10x10 Hd and the 8x8 Kit is the **8x8** plugin, which runs at 48 kHz sample rate. It is also possible to run the **2x8** plugin, which runs at 96 kHz. The table below summarizes the key differences when these plugins are used with the 10x10 Hd and 8x8 Kit hardware (some parameters are the same, but are included to show that they do not change with the plugin):

	2x8 plugin	8x8 plugin
Number of analog inputs	2	8
Number of analog outputs	8	8
Digital input sample rate <sup>1</sup>	20 to 216 kHz	20 to 216 kHz
Internal sample rate	96 kHz	48 kHz
Digital output sample rate <sup>1</sup>	96 kHz	48 kHz
Number of PEQ bands per block	5	6
Compressor block	No	Yes
Maximum channel delay	9 ms	15 ms

1. Applies to 10x10 Hd, or to 8x8 Kit with optional DIGI-FP board

## 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 plugin

When your order ships, your ordered plugin will be available from the [User Downloads](#) section of the miniDSP website. Download and install your plugin, as described in [Software installation](#) on page 12.

### 2. Familiarize yourself with the plugin

Before connecting your computer to the processor, it's worth familiarizing yourself with the user interface presented by the plugin. Click on and explore each of the tabs (**Input**, **Routing**, and **Output**). For more information, see [Configuring the processor](#) 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 we will reset all of the processing parameters 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

At this point, 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 miniDSP to default settings. For more information on configurations, see [Working with configurations](#) on page 33.

### 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 10x10 Hd or 8x8 Kit 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 18.

## 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). Turn the volume on your source and/or amplification down low, 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 (see [Acoustic measurement](#) starting on page 38) 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 computer connected. To do so, simply disconnect the USB cable. The processor “remembers” the configuration last set, and will continue to operate without the computer.

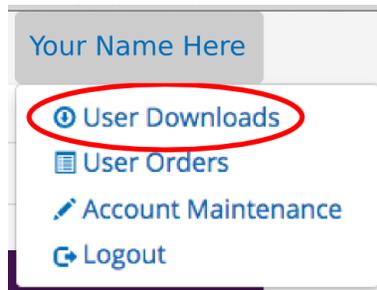
**Note:** while in offline mode, the configuration can still be modified in the plugin interface. These changes will not be downloaded to the processor until it is synchronized again. See [Working with configurations](#) on page 33.

## 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 10x10**. 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

Prior to installing the miniDSP software, 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.

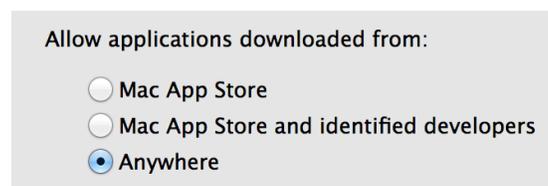
To install the miniDSP software, open the **Windows** folder of the download and double-click on the **MiniDSP\_8x8.exe** program. We recommend accepting the default installation settings. Once installation is complete, the **8x8** plugin will automatically start. Since the plugin checks for a network connection when starting up, a warning such as the following may appear. In this case, click on **Allow access**.



### 3.1.2 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 **Mac** folder of the download, and double-click on the **MiniDSP\_8x8.dmg** file to open it in a new window. Then double-click on the installer program, **Install MiniDSP-8x8.app**. Once installation is complete, the **8x8** plugin will automatically start.

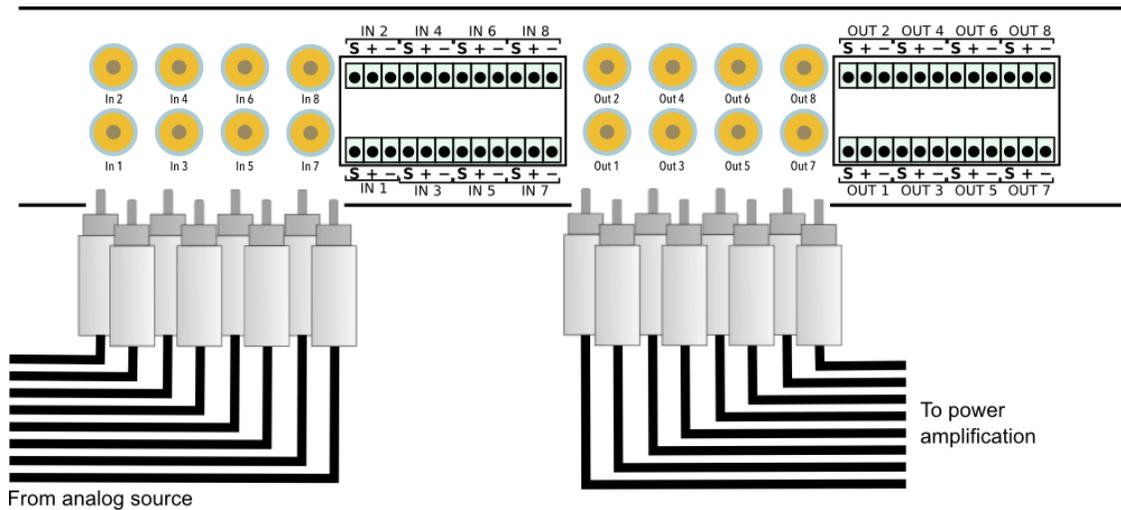
### 3.2 HARDWARE CONNECTIVITY

All connections to the 10x10 Hd are made on the rear panel. For the 8x8 Kit board, also see [Kit documentation](#).

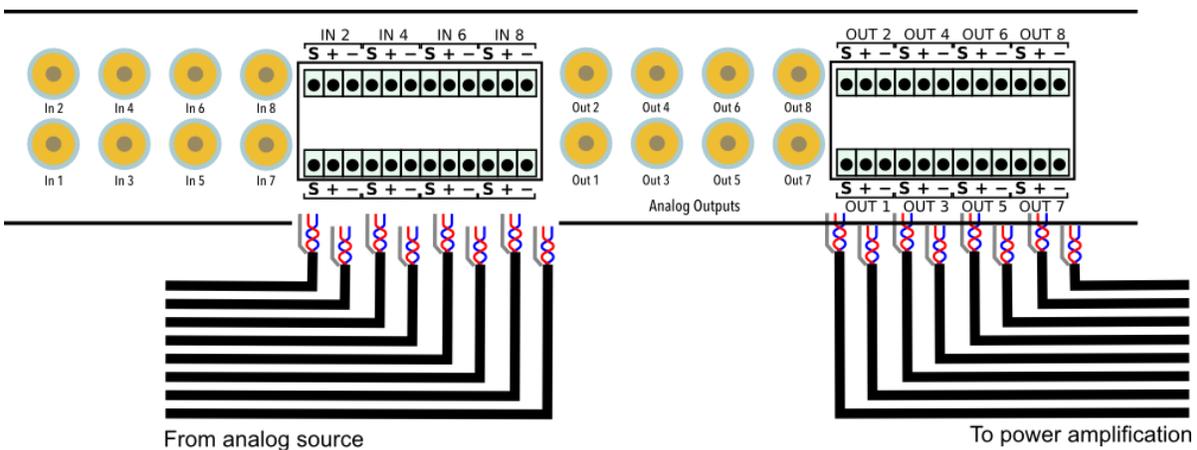
#### 3.2.1 Analog input and output

There are eight analog input channels and eight analog output channels, available in both unbalanced and balanced connections. Be sure to take careful note of the channel numbering shown in these diagrams and on the rear panel.

Unbalanced connections are made directly to the RCA jacks.



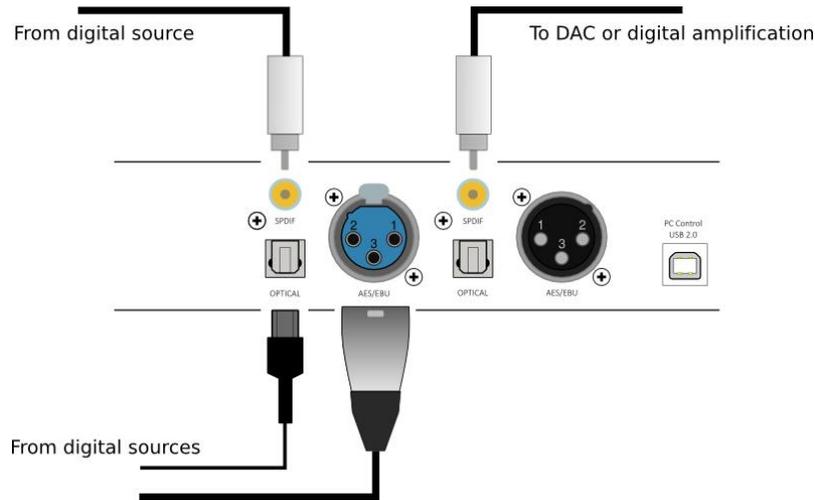
Balanced connections are made by connecting bare wire ends to the push-in Phoenix terminal blocks. For advice on using these, see [Balanced wiring tips](#) on page 46.



If using a mix of balanced and unbalanced connections, be aware that you cannot connect an input both ways. That is, for any given input channel, connect either to the RCA jack, or to the terminal block, but not both.

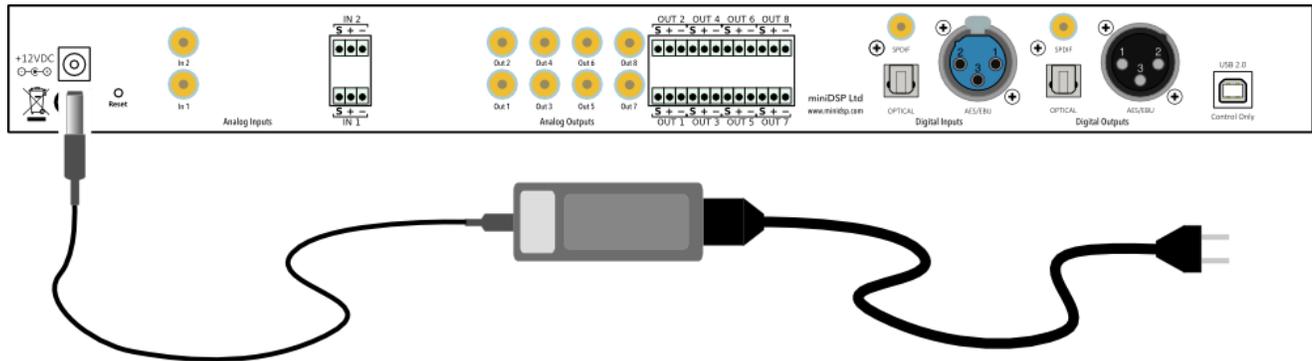
### 3.2.2 Digital input and output

Connect up to three digital sources: AES/EBU on XLR, S/PDIF on RCA, and optical (TOSLINK). Connect one of the digital outputs to your DAC or other device with a digital input. All digital outputs are simultaneously active, so multiple output connections can be made if necessary.



### 3.2.3 DC power

Fit the supplied IEC cable to the 12 VDC power supply. Plug the DC connector into the **+12VDC** socket on the rear panel of the 10x10 Hd, then plug the AC mains plug into the power outlet.

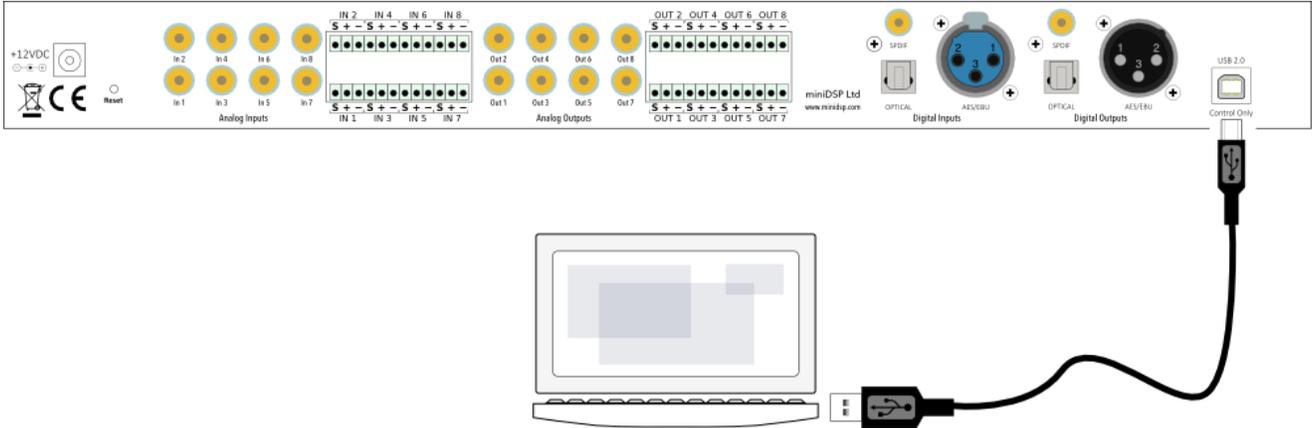


Apply power to the processor only after all analog input and output connections have been made. The processor can usually be left powered on. If powered on and off, the following sequence is recommended:

- On: Power on line-level equipment, including the 10x10 Hd, then turn on power amplification.
- Off: Turn power amplification off, then power off line-level equipment, including the 10x10 Hd.

### 3.2.4 USB

To configure the processor, connect its USB port to a USB 2.0 port on your computer using the supplied cable. Note that USB is used only for configuration—audio data cannot be streamed to the processor over USB.



### 3.3 GAIN STRUCTURE

The miniDSP audio processor provides a great deal of flexibility in matching with other equipment. In addition to providing balanced and unbalanced inputs and outputs, input sensitivity and output gain can be selected to adapt to different equipment. The maximum input and output signal levels are shown in Table 1.

	Inputs: Sensitivity jumper		Outputs: Gain switch	
	Closed	Open	Down	Up
Unbalanced	0.9V	2V	0.9V	2V
Balanced	—	8V	4V	8V

Table 1. Maximum input-output voltages

The combination of these provides a great deal of flexibility. Table 2 summarizes the net gain through the unit, depending on the choice of connection type and the input sensitivity and output gain settings.

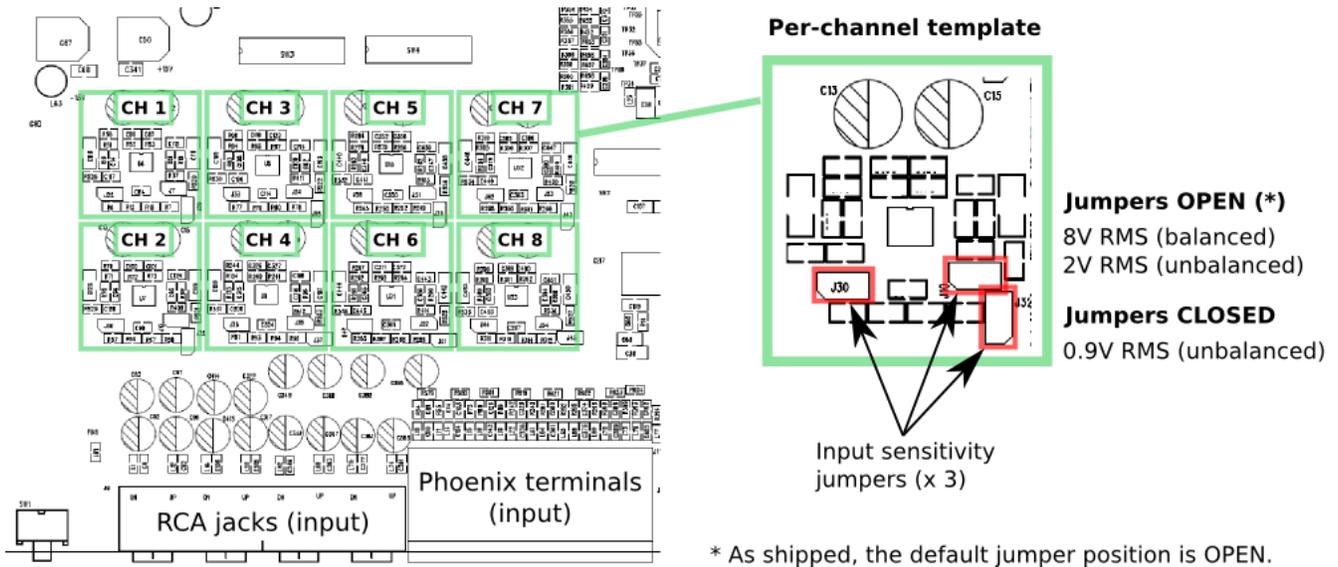
	Setting	Unbalanced output		Balanced output	
		0.9V	2V	4V	8V
Unbalanced input	0.9V	0 dB	6 dB	12 dB	18 dB
	2V	-6 dB	0 dB	6 dB	12 dB
Balanced input	8V	-18 dB	-12 dB	-6 dB	0 dB

Gain values are simplified to give exact multiples of 6dB.

Table 2. Input-output gain determined by connection type and gain structure settings

### 3.3.1 To change input sensitivity

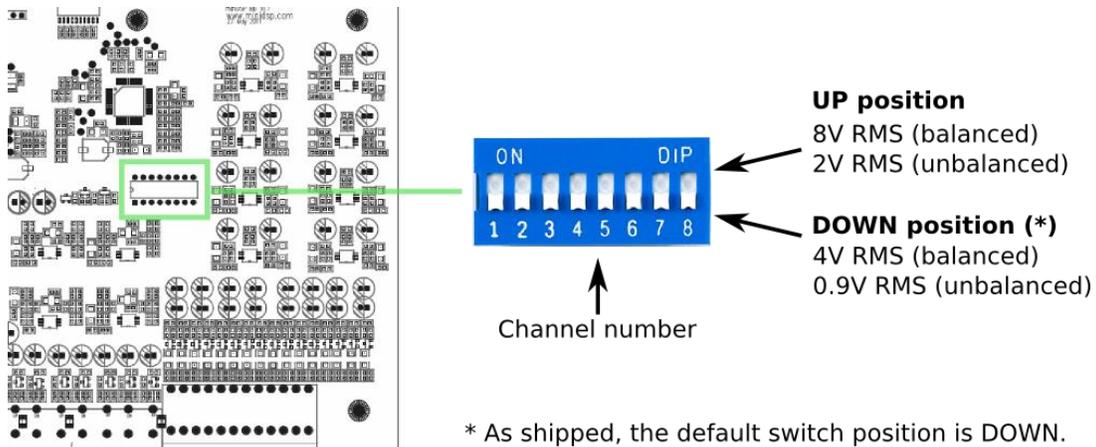
Input sensitivity can be changed for **unbalanced connection only**, by moving a set of three jumpers on each channel. Remove the top panel of the 10x10 Hd to access these jumpers. The diagram below illustrates the location of the jumper headers for each channel. For each channel that is to be changed, lift the jumpers and replace them in either the OPEN position (jumper pushed onto one pin only) or CLOSED position (jumper pushed onto both pins).



Do not make a balanced connection to the input of any channel with jumpers in the CLOSED position.

### 3.3.2 To change output gain

A set of eight DIP switches located on the main circuit board select the output gain of each channel. Remove the top panel of the 10x10 Hd to access these jumpers. The switches can be set independently for each channel, and affect output gain for both the balanced and unbalanced output connections to that channel.



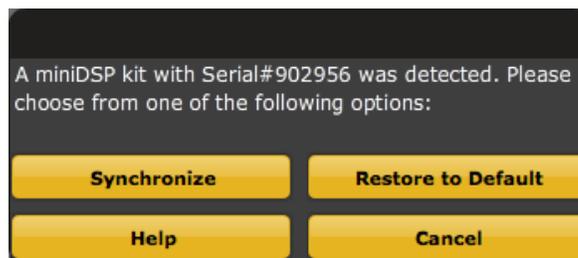
### 3.4 SYNCHRONIZING WITH THE PROCESSOR

Communication with the processor takes place over a USB connection. Note that USB is used for control purposes only. Audio data cannot be streamed to the processor over USB.

Ensure that the computer is connected to the processor by a USB 2.0 port. Then click on the **Connect&Synchronize** 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**

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 to Default**

Restore the data in the currently selected configuration to the factory defaults. When using this option, any connected output 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 ensure that you have saved the configuration to a file prior to using this option.

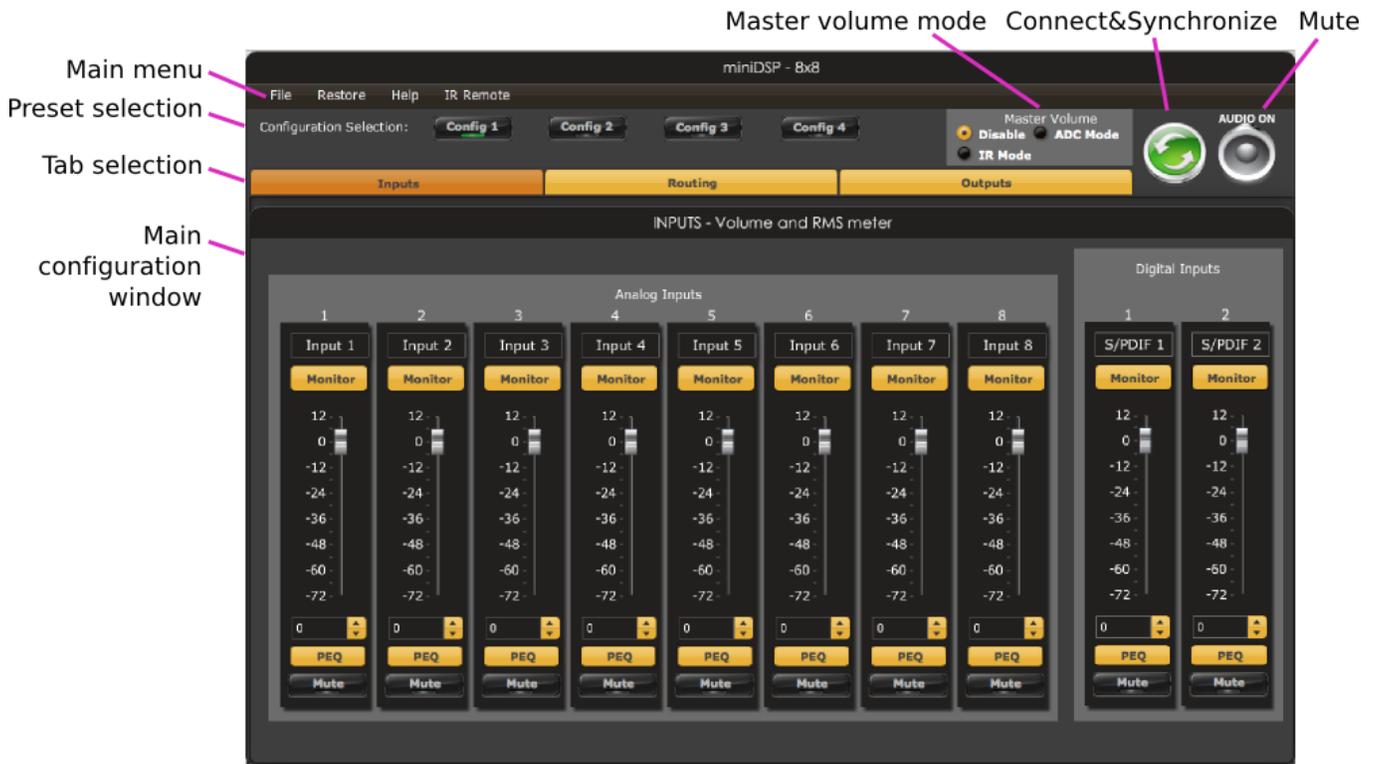
#### **Cancel**

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

## 4 CONFIGURING THE PROCESSOR

The processor is configured with the **8x8** plugin / user interface program. Once fully configured, the computer is no longer required, as source and preset selection can be done with the front panel or a remote control—see [Using the processor](#). If desired, however, the plugin can remain connected during use for real-time (“live”) control of all audio processing.

This screenshot shows the **8x8** plugin with the key areas highlighted:



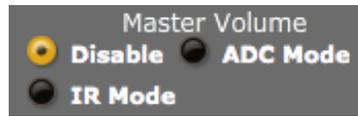
Note: during *initial* configuration of the processor, it is strongly recommended that any connected amplification be muted or powered off.

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



## 4.1 MASTER VOLUME SETTING

The **Master Volume** setting enables the processor's onboard master volume control. There are three options:



### Disable

Master volume is not enabled. This is the default setting for each configuration.

### ADC Mode

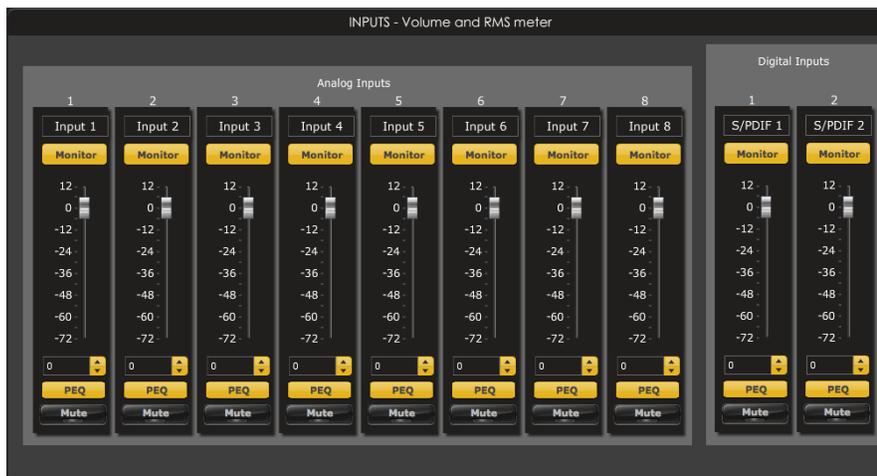
Master volume is controlled by a potentiometer connected to the correct jumper on the 8x8 Kit board. Select this option **only** if using the 8x8 Kit board and if a control potentiometer is connected as described in [Master volume control potentiometer](#).

### IR Mode

Master volume is controlled by an infrared remote. If using the 10x10 Hd, this option should always be selected. If using the 8x8 Kit board, it can be selected if the optional VOL-FP board is installed.

## 4.2 INPUT TAB

The **Input** tab displays a row of input channel control strips. There are eight analog input channels and two digital input channels (left and right signals of the selected stereo digital input):



The controls on the channel strip are as follows:

### 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** tab. To change the label, click on it, type a new label (up to eight characters), and press the Return key.

### Level meter

Click on this button to open an overlay window that displays a real time level meter. (The plugin must be in online mode to display signal levels.)

### Gain adjustment

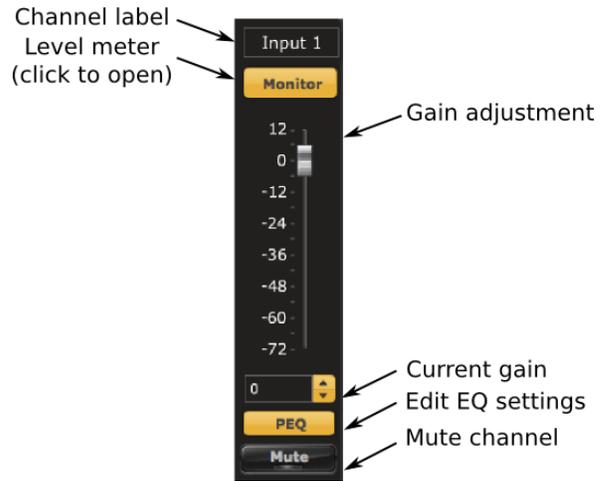
The gain of each channel can be adjusted by moving the Gain Adjustment slider, or by typing the desired gain into the Current Gain text box. The maximum gain setting is 12 dB, and the minimum gain setting is -72 dB. (0 dB, the default, is unity gain or no change in level.)

### PEQ settings

Click on this button to open the parametric EQ settings window for that channel. There are six parametric EQ filters on each input channel. For more details, see [Parametric EQ](#) on page 26.

### Mute

Press this button to mute that input channel. A visual indicator shows that the channel is muted.



## 4.3 ROUTING TAB

The **Routing** tab displays the matrix mixer, which sets up the routing from input channels to output channels. The input channels are labeled along the left, and the output channels along the top. Each input-output channel assignment is turned on and off by clicking the button corresponding to the two channels. (The labels displayed for the input and output channels are set up in the **Input** and **Output** tabs.)

If more than one input signal is routed to an output channel, those channels are mixed together (summed). For many examples of how to use the matrix mixer, see the application note [Surround sound equalization with the miniDSP 10x10 Hd](#).

The matrix mixer provides a very powerful tool for setting up various configurations. In addition, different routing setups can be set up in different presets, and then selected with a remote control.

Routing										
	Analog Out								Digital Out	
Analog In	Output 1	Output 2	Output 3	Output 4	Output 5	Output 6	Output 7	Output 8	S/PDIF 1	S/PDIF 2
Input 1	On	Off	Off							
Input 2	Off	On	Off	Off						
Input 3	Off	Off	On	Off	Off	Off	Off	Off	Off	Off
Input 4	Off	Off	Off	On	Off	Off	Off	Off	Off	Off
Input 5	Off	Off	Off	Off	On	Off	Off	Off	Off	Off
Input 6	Off	Off	Off	Off	Off	On	Off	Off	Off	Off
Input 7	Off	Off	Off	Off	Off	Off	On	Off	Off	Off
Input 8	Off	Off	Off	Off	Off	Off	Off	On	Off	Off
Digital In	Output 1	Output 2	Output 3	Output 4	Output 5	Output 6	Output 7	Output 8	S/PDIF 1	S/PDIF 2
S/PDIF 1	Off	Off	Off	Off	Off	Off	Off	Off	On	Off
S/PDIF 2	Off	Off	Off	Off	Off	Off	Off	Off	Off	On



If analog and digital inputs are switched to the same output, they are mixed together. To *switch* between analog and digital inputs, make a copy of your configuration and load it into another configuration preset. Then change the routing so that one routes only the analog input, and one routes only the digital input. Select between them with an infrared remote control.

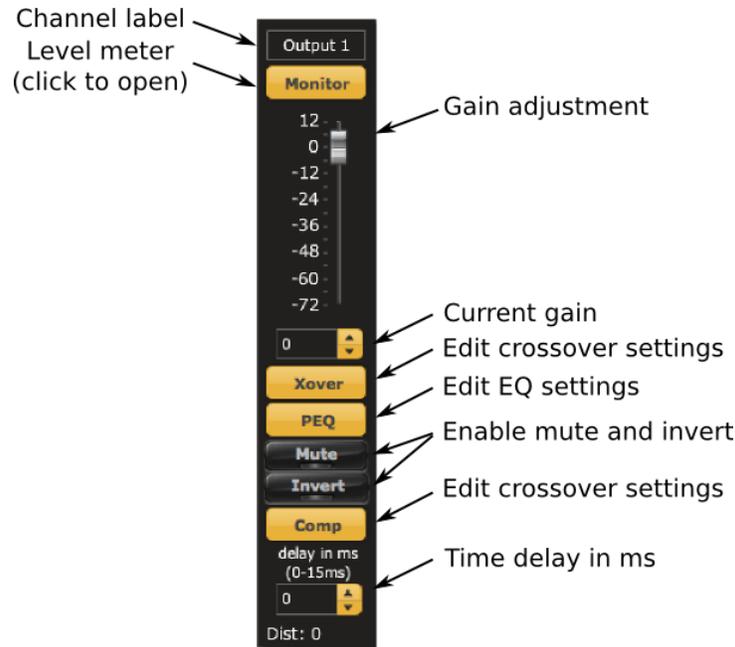
#### 4.4 OUTPUT TAB

The **Output** tab displays a row of output channel control strips. All channels are identical, except that the two digital output channels have a reduced maximum time delay (8ms instead of 15 ms).

The 'Outputs' interface is divided into two main sections: Analog Outputs (1-8) and Digital Outputs (1-2). Each channel strip includes a 'Monitor' button, a volume slider (ranging from -72 to 12 dB), a 'Dist' (Distortion) control set to 0, and a series of processing buttons: Xover, PEQ, Mute, Invert, and Comp. The Comp section includes a 'delay in ms' slider. For Analog Outputs, the delay range is 0-15ms. For Digital Outputs, the delay range is 0-8ms.

### 4.4.1 Channel strip layout

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



### 4.4.2 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** tab. To change the label, click on it, type a new label (up to eight characters), and press the Return key.

### 4.4.3 Gain control and level monitoring

#### Level meter

Click on this button to open an overlay window that displays a real time level meter for the current signal. (The plugin must be in online mode to display signal levels.)

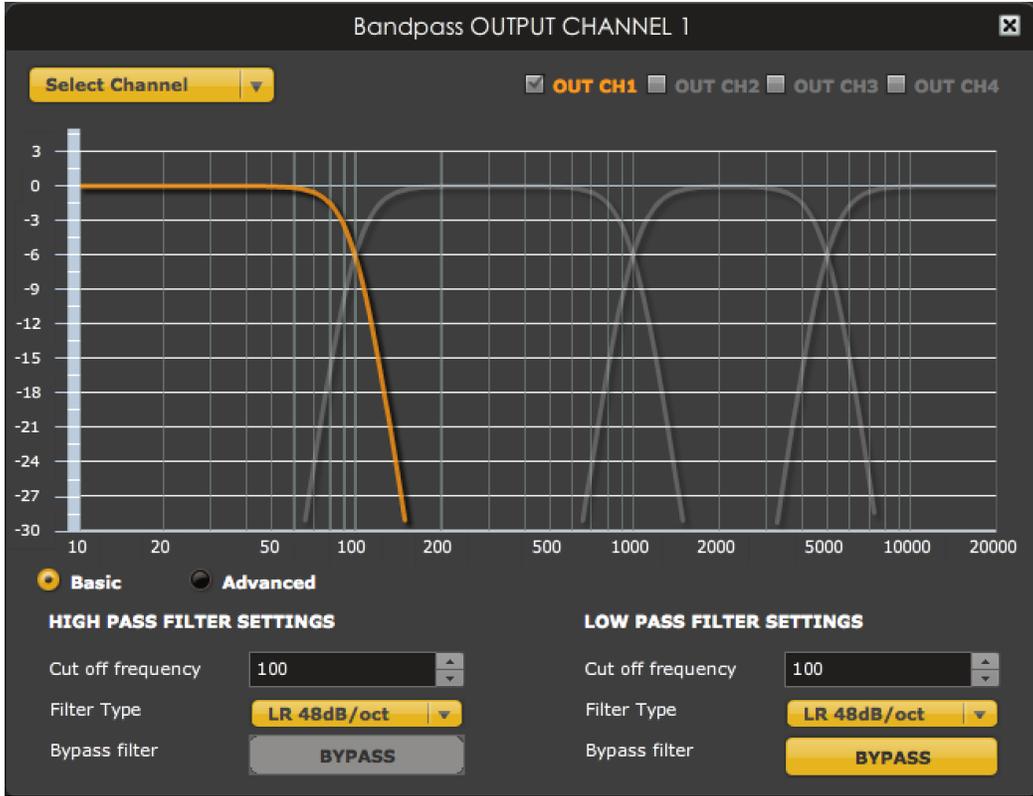
#### Gain adjustment

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.)

The level meters are useful in many situations. For example, when adding filters with boost, monitor the level meters with typical signals and maximum levels to ensure that there is no clipping. The meters can also be used during normal operation to monitor for or to help locate level or gain structure problems.

#### 4.4.4 Crossover

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

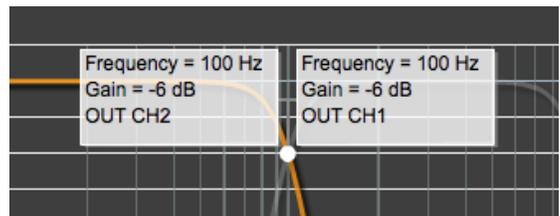


Crossovers “split” the frequency band to send to different drivers. In a two-way loudspeaker, for example, a *low pass* filter is used to remove high frequencies from the signal sent to the woofer, and a *high pass* filter is used to remove low frequencies from the signal sent to the tweeter. In a three-way 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 even disabled completely. This allows maximum flexibility in matching your processing to the acoustic characteristics of your loudspeakers.

Crossover filters on the analog output channels are displayed in two groups of four—channels 1 through 4, and channels 5 through 8. Each group is configured by default as a four-way crossover. The current channel is displayed highlighted with the other three in the group in grey. You will need to change these settings for your own system. For multichannel processing, you may wish to start by bypassing all crossover filters.

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 30.

## 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

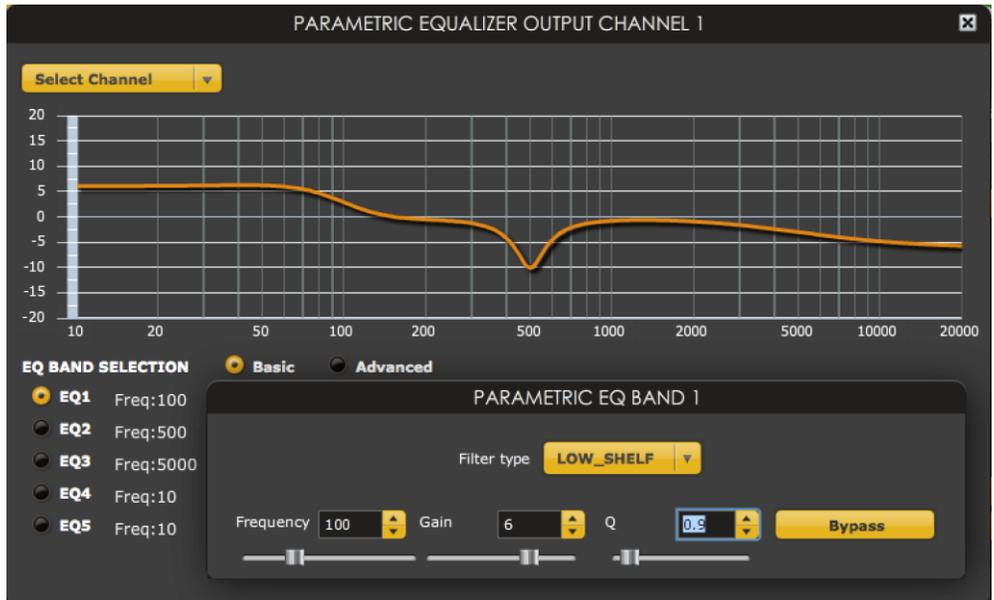
Clicking on the **Bypass** button disables or enables that high pass or low pass filter. The filter is enabled when the button is "lit".

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. To link a channel, select the other channel from the drop-down menu at the top left of the **Xover** screen, and click the **Link** checkbox.



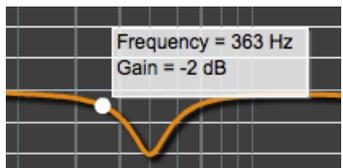
### 4.4.5 Parametric EQ

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 six 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. 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 radio buttons **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 30.

## 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 six to five.

## 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.

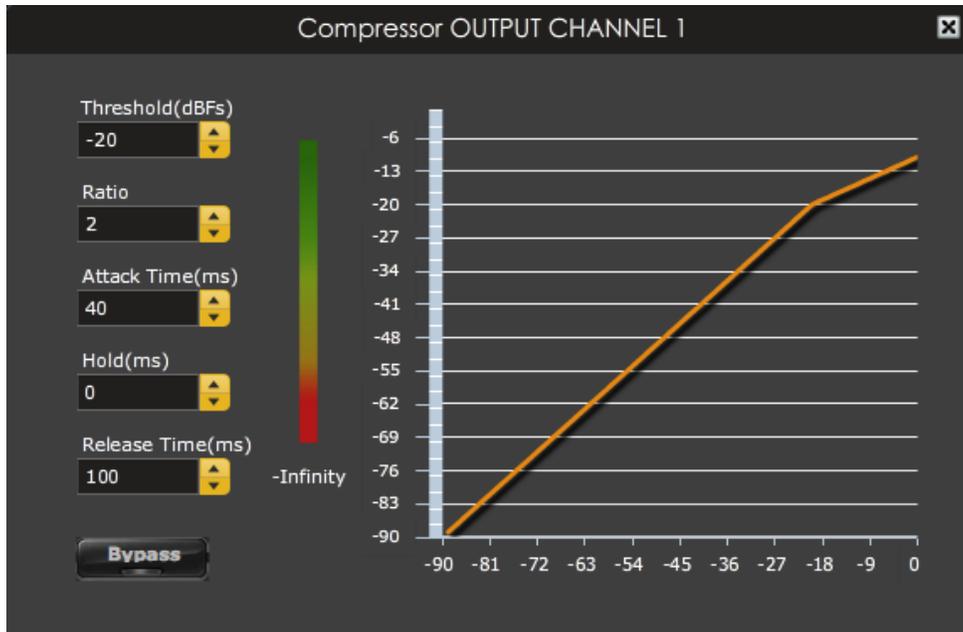
## Bypass

The **Bypass** button enables or disables a filter. The filter is enabled 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.

#### 4.4.6 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](https://en.wikipedia.org/wiki/Dynamic_range_compression).

#### 4.4.7 Invert and mute

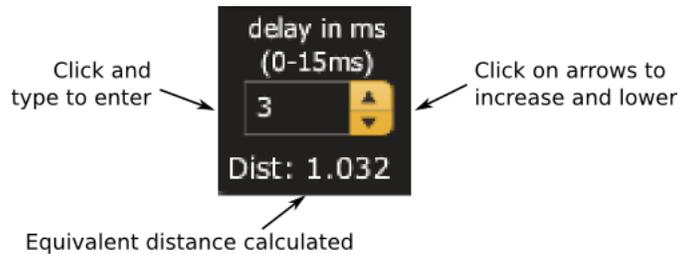
Each channel can be inverted in polarity, and individually muted. When either of these options is selected, the visual indicator on the button is "lit":



#### 4.4.8 Time delay

A delay of up to 15 ms can be applied to each analog output channel, and 8 ms to each digital 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 time delay corresponds to a distance. This distance is shown in meters below the entry box. The maximum time delay of 15 ms corresponds to a distance of approximately 5.2 meters (17 feet).



## 4.5 CUSTOM BIQUAD PROGRAMMING

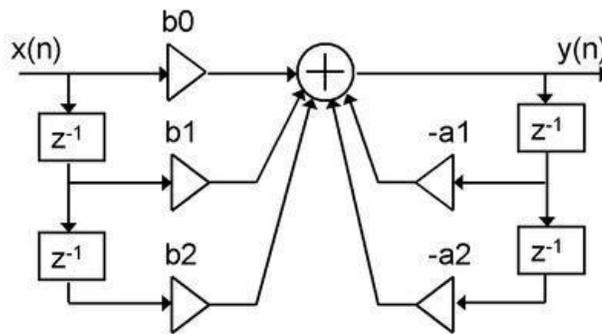
Custom biquad programming is available in the PEQ and Crossover blocks. Its purpose is to allow you to directly provide the low-level parameters aka *biquad coefficients* that control the digital filters of the processor, thus providing an almost infinite degree of flexibility.

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

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

### 4.5.2 Using custom biquad programming

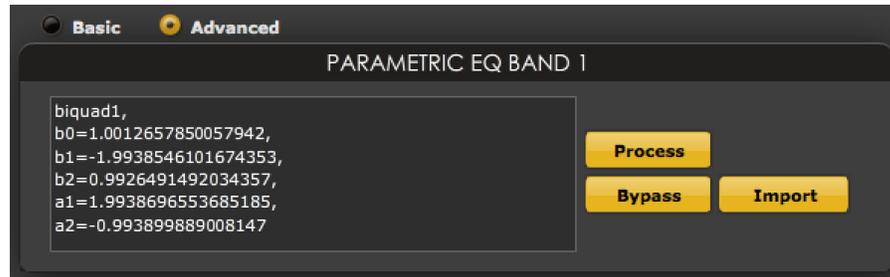
Each crossover block and PEQ filter has a selector that switches it to advanced mode:



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. Note that the design program must be set for a 48 kHz sample rate. The number of filters is limited to a maximum of six (in the **8x8** plugin, five for the **2x8** plugin).

This example illustrates the correct file format:

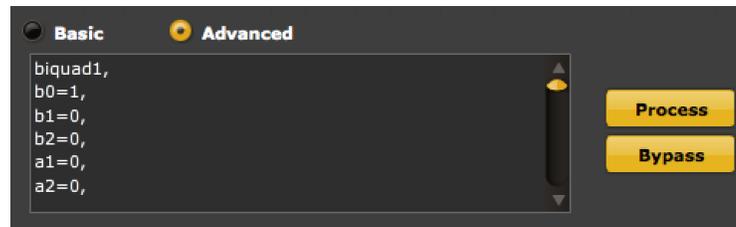
```
biquad1,
b0=0.998191200483864,
b1=-1.9950521500467384,
b2=0.996920046761057,
a1=1.9950521500467384,
a2=-0.9951112472449212,
biquad2,
b0=0.999640139948623,
b1=-1.9981670485581222,
b2=0.9985489719847982,
a1=1.9981670485581222,
a2=-0.9981891119334211,
biquad3,
...
biquad4,
...
biquad5,
...
biquad6,
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 six biquads, then only that number of biquads will be imported. For example, if importing a file with three biquads, the first three filters will be set, and the last three will not be changed. (Be careful: if the last line ends with a comma, that counts as an extra biquad.)

If the file contains more than six biquads, then an error will be reported and no filters will be changed.

### Crossover advanced mode

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



### 4.5.3 Biquad design software

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

#### 4.5.3.1 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)

#### 4.5.3.2 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](#).

#### 4.5.3.3 Active Crossover Designer (ACD)

Active Crossover Designer (ACD) by Charlie Laub performs the complete set of functions for active crossover design and generates biquad coefficients as its output. Download it here (requires Microsoft Excel):

- <http://audio.claub.net/software/ACD/ACD.html>

## 4.6 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 then be selected from the front panel or via remote control.

### 4.6.1 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.

### 4.6.2 Selecting a configuration

The active configuration is selected by one of the four buttons in the Configuration Selection area. By default, configuration 1 is selected:



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

#### The plugin is in online mode

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

#### The plugin is in offline mode

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

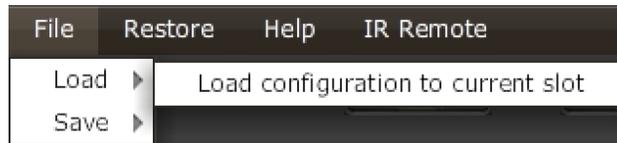
### 4.6.3 Saving and loading configurations

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 Hd be saved to a file, to ensure that the configuration is not lost if the processor is inadvertently reset.

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



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



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



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

#### 4.6.4 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.

### 4.7 KEYBOARD SHORTCUTS

The **8x8** 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:

- Output channel gain
- Crossover frequency and filter type
- PEQ filter frequency, gain, and Q

#### **Space**

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

## 5 USING THE PROCESSOR

Once configuration is complete, the computer is not required and can be disconnected. The front panel and/or an infrared remote can be used to control:

- Master volume
- Master mute (remote control only)
- Source selection
- Preset selection



(The 8x8 Kit board requires the optional add-on boards DIGI-FP and VOL-FP for these features to be enabled.)

### 5.1 STATUS INDICATORS

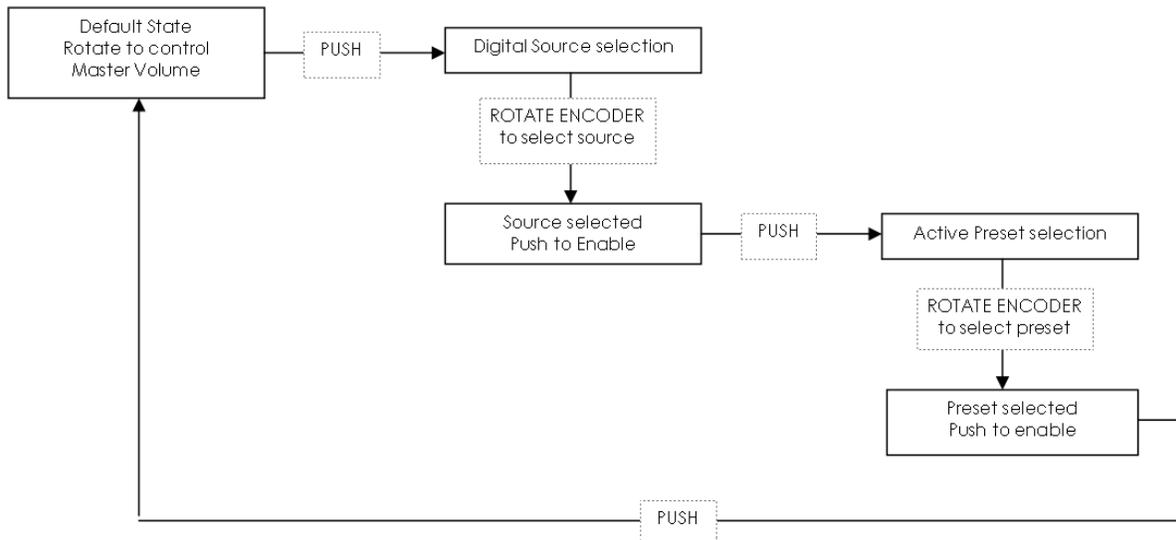
The current status of the processor is indicated by a set of LEDs:

**Source** Indicates the currently selected digital input.

**Preset** Indicates the currently selected configuration preset (1 through 4).

### 5.2 FRONT PANEL CONTROLS

The processor uses a minimalistic physical control design with a single control knob. Normally, rotating the knob changes the master volume (ensure that **IR Mode** is selected in the **Master Volume** block). Pressing the knob moves into preset and source selection mode, as shown in the following flowchart.



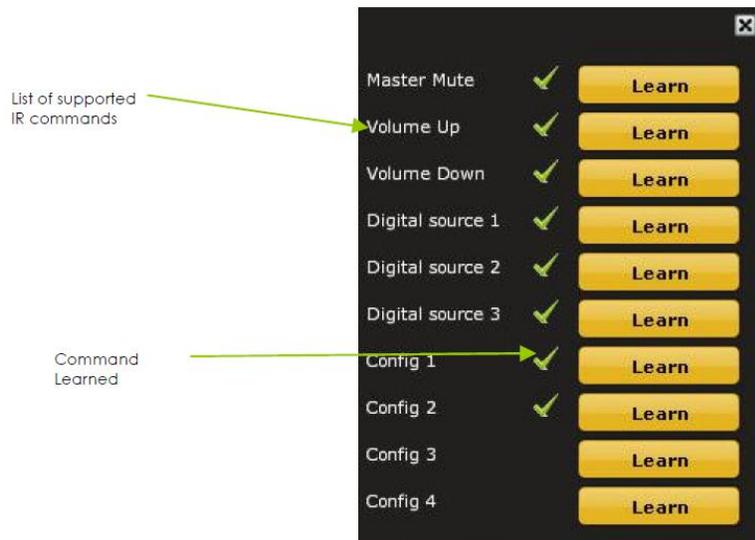
### 5.3 INFRARED REMOTE CONTROL

Many standard and programmable remote control units can be used with the 10x10 Hd. Instead of adding another remote to your collection, the 10x10 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.

Note: changing the selected configuration preset with the remote will cause the plugin to disconnect.

## 6 ACOUSTIC MEASUREMENT

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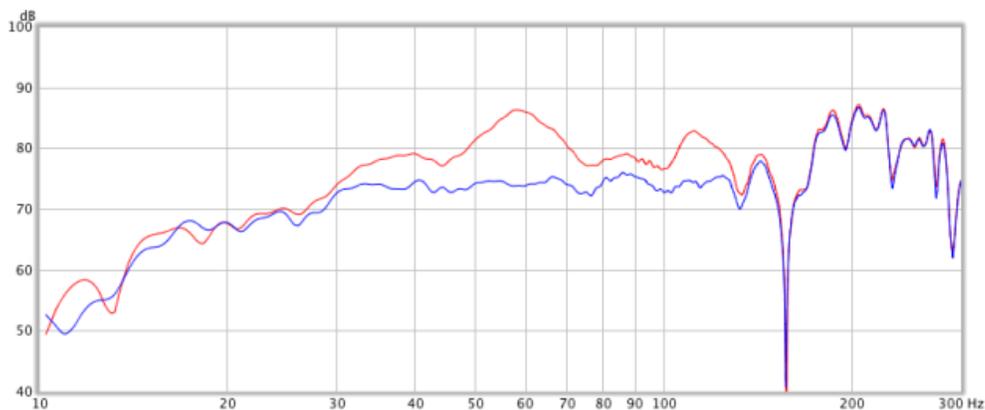
To obtain best results from your miniDSP audio processor, acoustic measurement will be required. This section provides a brief introduction.

### 6.1 WHAT IS ACOUSTIC MEASUREMENT?

An audio system has a frequency response, which includes the response of the electronics, the loudspeaker, and the listening room. In order to understand how these all combine to affect the sound heard by a listener, acoustic measurement using a suitable microphone and computer software is required.

The graph below shows a typical frequency response measurement for the bass region, using the freeware program [Room EQ Wizard](#). In this example, we are equalizing the subwoofer, so the frequency scale from left to right progresses from 10 Hz to 300 Hz. The height of the solid red line indicates the measured SPL (sound pressure level) at each frequency.

Where there are peaks or other irregularities in the response, these can be corrected electronically with parametric EQ. The blue line in the graph shows the frequency response after correction.



### 6.2 TYPES OF MEASUREMENT

Depending on your goals, there are a number of types of acoustic measurement:

**For active crossovers.** Measure each loudspeaker driver individually, at a distance of 0.5 or 1 meter (1.5 or 3 feet). Use these measurements to adjust the parametric equalizer for each driver, and to fine-tune the crossover (frequency and slope) settings.

**For subwoofer integration.** Measure the subwoofer with the microphone placed at the listening position. Typically, moving the subwoofer to find the best response without EQ will yield best performance after EQ. The REW auto-EQ function can be used to quickly generate an optimum equalization setting.

**For room correction.** Measure the response of the system with the microphone placed at the listening position. Apply equalization to make broad corrections to the frequency response and to reduce resonances. Adjustments can be confirmed by listening as well as by measurement.

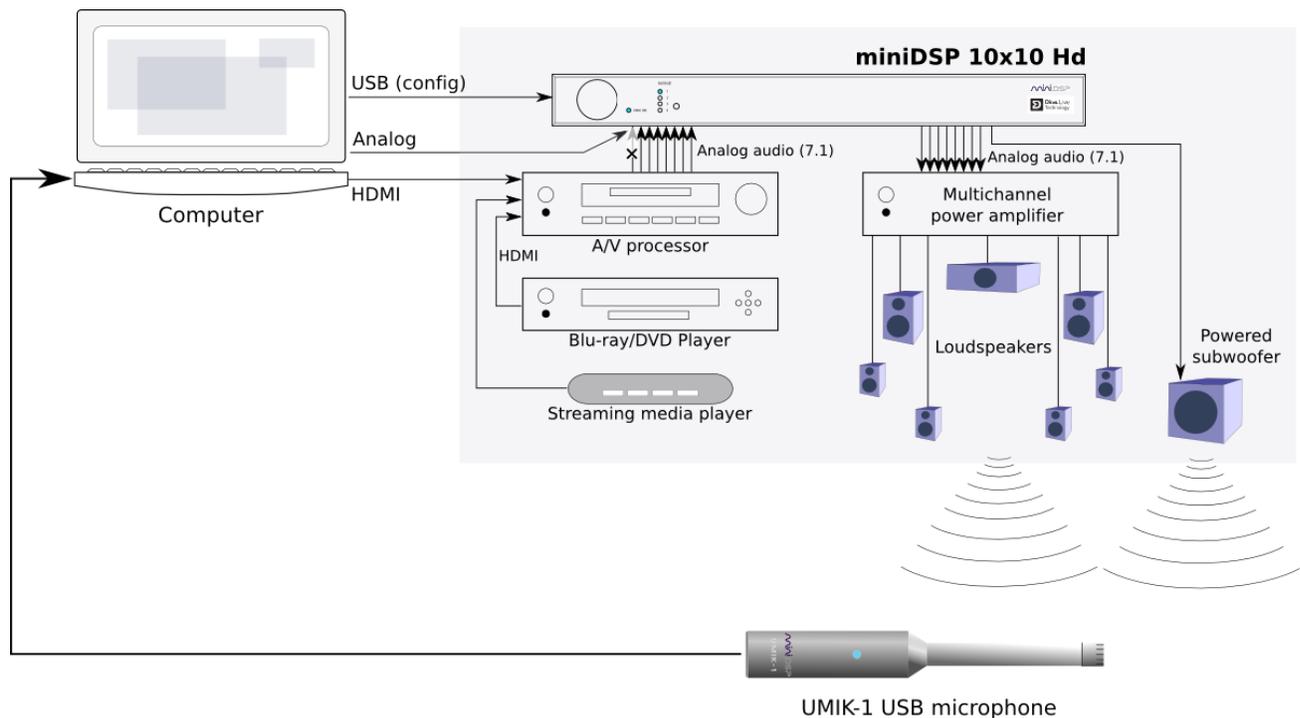
### 6.3 PERFORMING ACOUSTIC MEASUREMENTS

We suggest a miniDSP UMIK-1 calibrated USB measurement microphone. The UMIK-1 provides an extremely cost-effective acoustic measurement solution. Simply connect the UMIK-1 to a USB port on your computer.



You can also use another measurement microphone connected via a USB soundcard (for example). Make sure that the microphone you use is specified for acoustic measurement purposes. Microphones used for recording vocals and instruments do not have a flat frequency response and will give poor results in this application.

You will need to provide an audio signal from your computer. While an analog connection directly to the processor can be used, the best connection method in a typical home theater system is via an HDMI input on the A/V processor or receiver. This enables any channel to be selected from the computer without switching any cables.

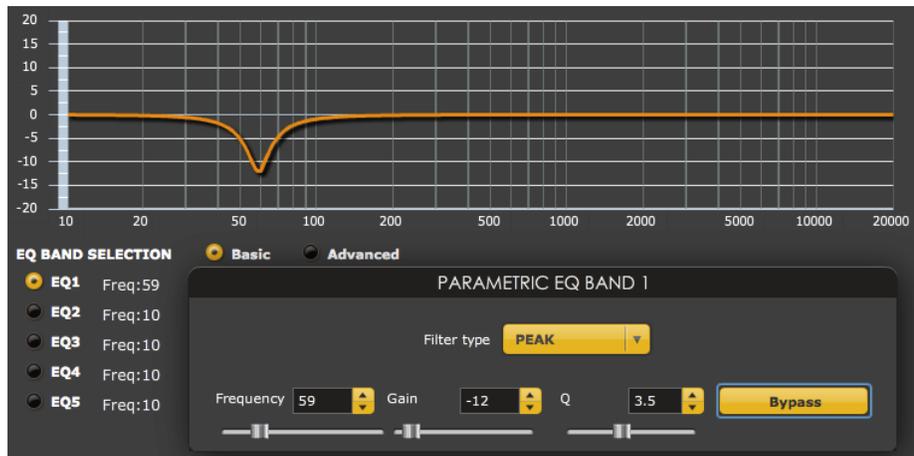


An acoustic measurement program such as [Room EQ Wizard](#) (REW) can then be used to generate a test signal that plays through the system. A number of other programs, both free and paid for, can be used, such as [FuzzMeasure Pro](#) (Mac) and [HolmImpulse](#) (Windows). It is important to use a program that is capable of accurate frequency response measurements. Programs that use a third-octave RTA (real time analyzer) are not sufficiently accurate for this purpose.

For detailed information on using an HDMI connection with Room EQ Wizard, see the app note [Using the UMIK-1 and REW with HDMI output - Windows](#) or [Using the UMIK-1 and REW with HDMI output - Mac](#).

## 6.4 ADJUSTING EQUALIZER SETTINGS

Once a frequency response plot of a channel has been obtained, the parametric equalizers (PEQ) on that channel can be used to adjust or correct the frequency response. For example, we can use the PEQ user interface directly to create filters that alter the frequency response. In the response curve shown above there is a significant peak at 59 Hz. We can largely remove this peak by applying a corresponding dip in the frequency response with these filter settings:



Dips in the frequency response can be treated with a corresponding peak filter, but care should be taken not to use too much boost or to use high-Q filters. Too much boost can cause signal clipping, and narrow dips are typically caused by acoustic cancellations in the room and cannot be removed with EQ.

Shelving filters can also be used to alter the frequency response. For example, if more weight in the bass is desired, this can be accomplished with a low-shelf filter with some boost, of 3 to 6 dB. Or, if less high treble emphasis is desired, this can be accomplished with a gentle high-shelf filter with a few dB cut at high frequencies.

More automated methods can also be used. With Room EQ Wizard (REW), for example, we can:

1. Run a frequency response measurement with REW.
2. In the REW Equalization window, generate a set of correction filters.
3. Export the generated filter to a file.
4. Open a PEQ block in the **8x8** plugin, set it to Advanced mode and import the file.

Other options are listed in the section [Biquad design software](#).

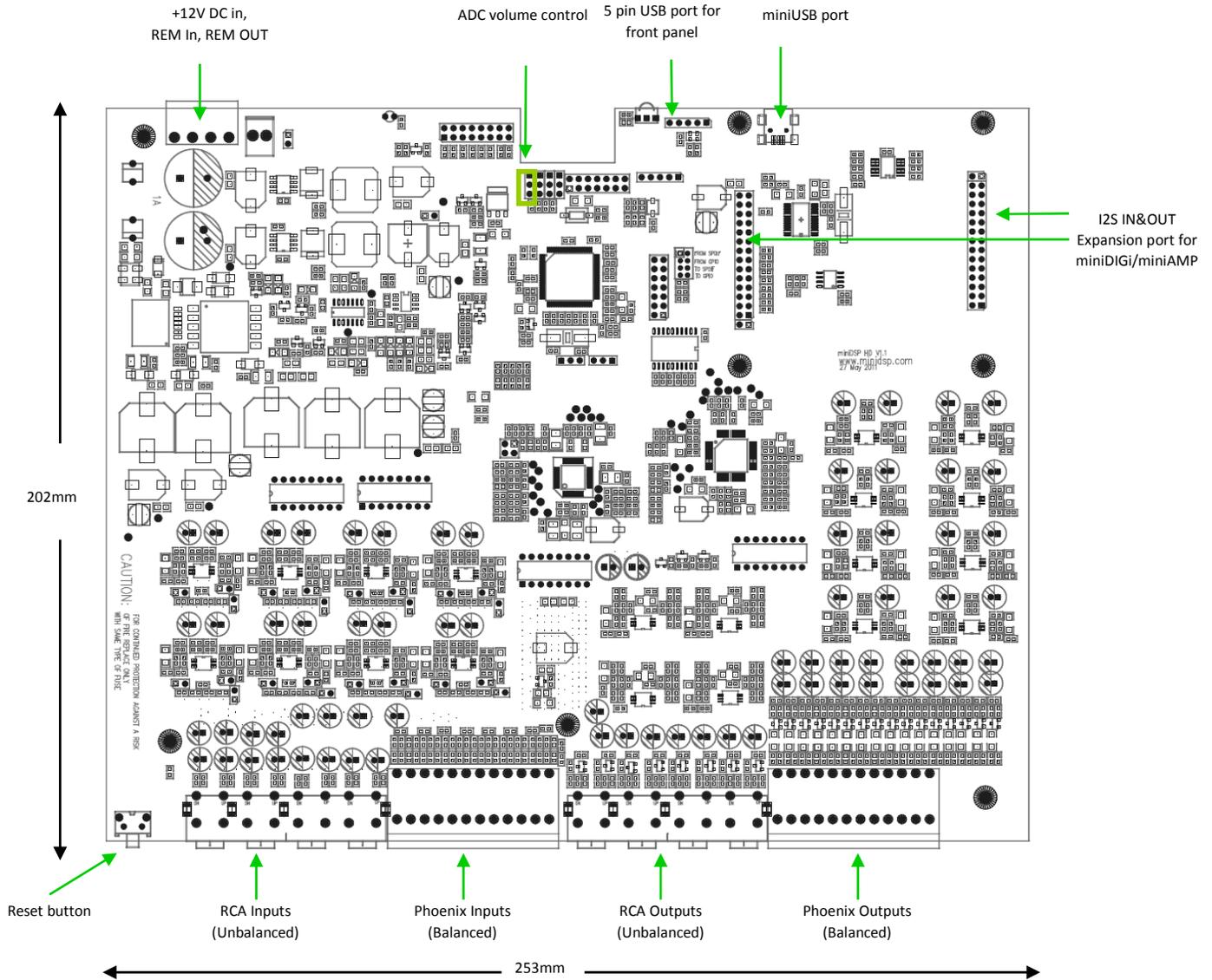
## 6.5 MORE INFORMATION

For more detailed information on these topics, please see our extensive collection of application notes on [acoustic measurement](#) and [digital crossovers](#). And always feel free to ask a question on our [forum](#)!

## 7 KIT DOCUMENTATION

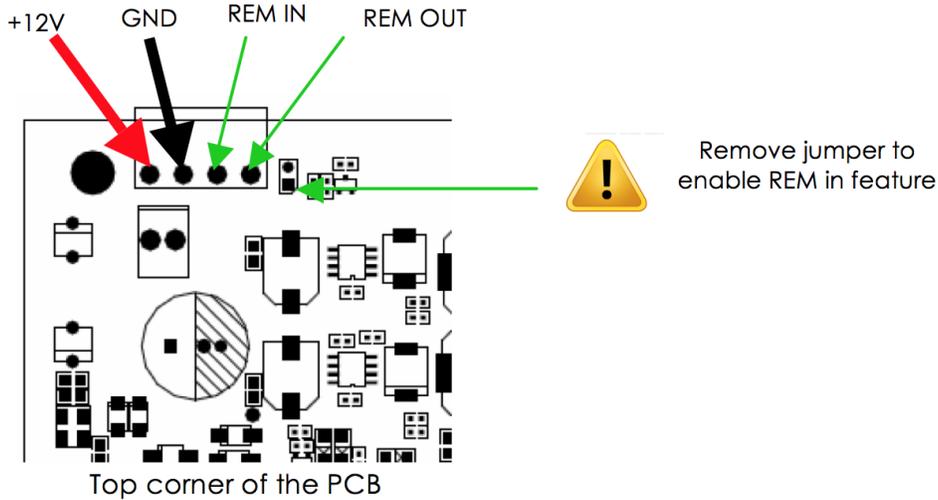
The miniDSP 8x8 Kit is a bare circuit board that can be incorporated into your own chassis and combined with other electronics components.

### 7.1 BOARD LAYOUT AND FEATURES



## 7.2 DC POWER CONNECTIVITY

The miniDSP 8x8 requires a 12V DC regulated power supply of at least 1.5A (18W). Power is supplied via a Phoenix terminal block connector. Be careful to apply the correct polarity to the terminal block connector as below. For a quick confirmation, take a look underneath the PCB for the silkscreen labels.



This Phoenix connector also has terminals for remote enable/disable. Remove the jumper as shown above to enable this feature.

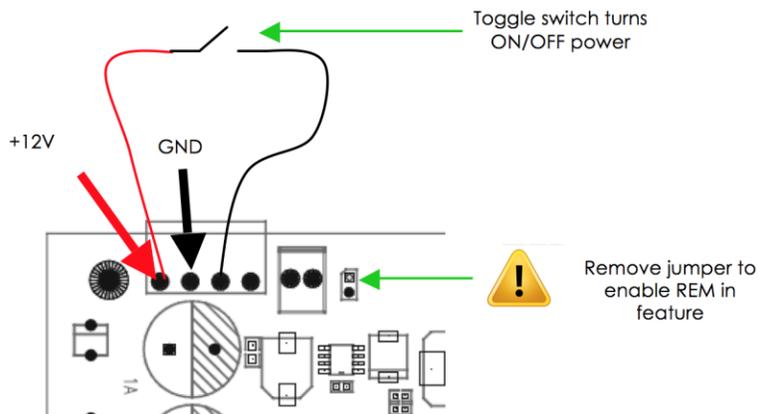
### REM IN

This input will turn the 8x8 Kit on and off. This is a useful feature for battery-operated systems, to avoid draining the battery when the 8x8 Kit is not in use. A simple example is shown below. The behavior is as follows:

- 0V < REM IN < 4V: miniDSP 8x8 is OFF
- 4V < REM IN < 12V: miniDSP 8x8 is ON

### REM OUT

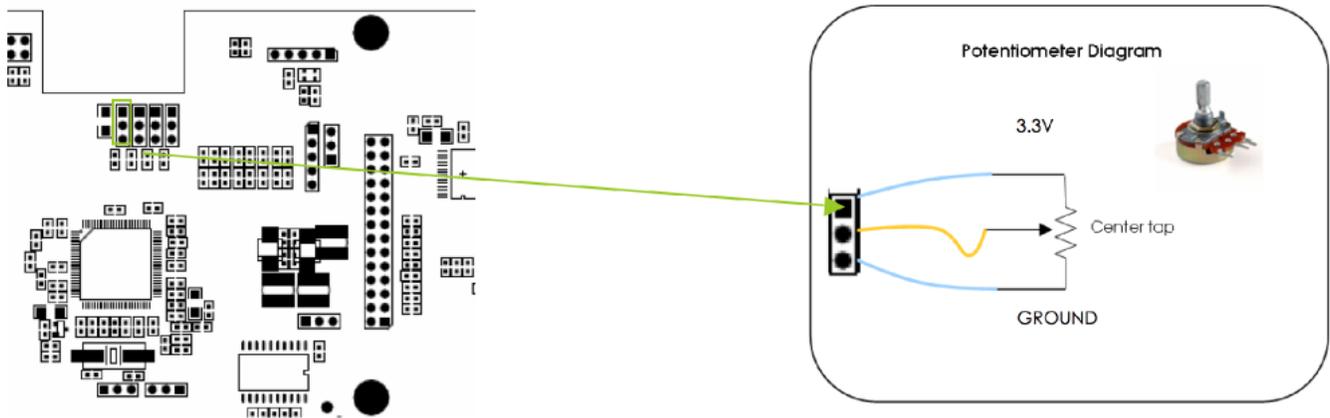
This output supplies a delayed version of REM IN, to trigger external equipment after a 2-second delay. It provides 12V @ 10mA.



### 7.3 MASTER VOLUME CONTROL POTENTIOMETER

Master volume control of all output channels (analog and digital) can be accomplished with a 10 to 100 kΩ linear potentiometer connected to the designated 3-pin header (standard 0.1"/2.54mm pitch). The wiring diagram for the potentiometer is shown in the diagram below. No attenuation (maximum volume) is applied when pin 2 of the header is at 3.3 V, while maximum attenuation (minimum volume) is applied when pin 2 is at 0 V.

For your convenience, a suitable potentiometer already wired to a 3-pin female connector can be purchased together with the kit board, or separately from the Accessories section of our [website](#).

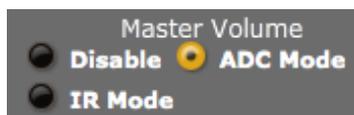


The connection to the volume header must be performed with the board powered down. If your volume control operates in reverse i.e. volume decreases as the pot is rotated clockwise, turn off connected equipment, remove power from the miniDSP board, and reverse the connector.

A single potentiometer can be connected to multiple 8x8 Kit boards. This allows master volume control of eight or more output channels. This diagram illustrates the wiring for master volume control of multiple boards:

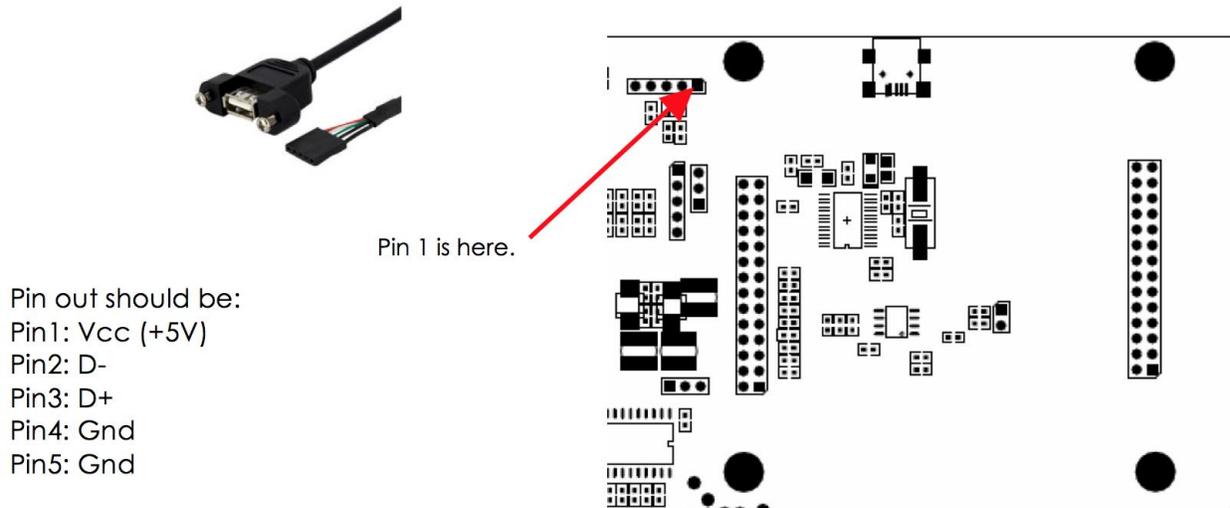


Finally, to enable master volume control, select the **ADC Mode** radio button in the **Master Volume** block.



## 7.4 FRONT PANEL USB PORT

The 8x8 Kit board has a 5-pin header (standard 0.1"/2.54mm pitch) located near its on-board USB connector, for use in builds requiring a panel mount USB connector.



This header will interface to common off-the-shelf cable assemblies (typically used in computer hardware). Although this is a standard connector, you should nonetheless confirm the pin-out with the manufacturer, or you could easily cause damage to your computer or 8x8 Kit. In most cases, pins 4 and 5 can be identified by a black wire, but once again, DO check the manufacturer's spec of the connector to prevent any short-circuits.



For your convenience, a suitable cable can be purchased together with the kit board, or separately from the Accessories section of our [website](#). Alternatively, use a DIGI-FP board, which has a USB Type B connector.

## 7.5 CONNECTION TO VOL-FP AND DIGI-FP

[VOL-FP](#) and [DIGI-FP](#) are optional add-on boards that can be purchased together with your 8x8 Kit or separately on our website. They provide volume and infrared remote control and digital input/output respectively. For information on how to connect them to the 8x8 board, please see this application note:

<http://www.minidsp.com/images/documents/DIGI-FP + VOL-FP connectivity.pdf>

## 8 ADDITIONAL INFORMATION

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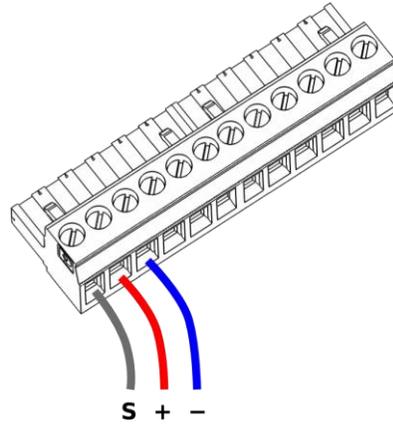
### 8.1 SPECIFICATIONS

<b>Computer connectivity</b>	Driverless USB 2.0 control interface for Windows and Mac OS X
<b>Analog inputs</b>	<p>8 x inputs, on balanced (terminal block) or unbalanced (RCA) connectors</p> <ul style="list-style-type: none"> <li>• Max balanced input voltage: 8 VRMS</li> <li>• Max unbalanced input voltage: 2.0 or 0.9 VRMS (jumper selectable)</li> </ul>
<b>Analog outputs</b>	<p>8 x outputs, on balanced (terminal block) and unbalanced (RCA) connectors</p> <ul style="list-style-type: none"> <li>• Max balanced output voltage: 8.0 or 4.0 VRMS (DIP switch selectable)</li> <li>• Max unbalanced output voltage: 2.0 or 0.9 VRMS (DIP switch selectable)</li> </ul>
<b>Digital Audio Inputs</b>	<p>Digital audio source selectable from IR remote or front panel:</p> <ul style="list-style-type: none"> <li>• AES/EBU on Neutrik 3-pin female XLR / Isolated with digital audio transformer</li> <li>• S/PDIF on RCA connector / Isolated with digital audio transformer</li> <li>• Toslink on Optical connector</li> </ul> <p>A high quality onboard Asynchronous Sample Rate Converter ensures compatibility with most sample rates, from 20–216kHz.</p>
<b>Digital Audio Outputs</b>	<p>Processed digital audio available on three outputs:</p> <ul style="list-style-type: none"> <li>• AES/EBU on Neutrik 3-pin male XLR / Isolated with digital audio transformer</li> <li>• S/PDIF on RCA connector / Isolated with digital audio transformer</li> <li>• Toslink on Optical connector</li> </ul> <p>Digital audio is always output at a sample rate of 48 kHz (with 8x8 plugin).</p>
<b>Audio resolution</b>	24-bit (32-bit internal), 48kHz sample rate (with 8x8 plugin)
<b>Storage/presets</b>	<p>All settings controllable in real time from software user interface.</p> <p>Up to 4 presets stored in local flash memory.</p>
<b>Infrared remote</b>	<p>“Learning remote” capabilities (NEC, Philips, Sony)</p> <p>Controls master volume, mute, digital input selection, preset selection.</p>
<b>Power supply</b>	Universal 12 VDC single supply, 2.1 mm center-positive (supplied)
<b>Dimensions (H x W x D)</b>	41.5 x 429 x 252 mm; 1RU height

## 8.2 BALANCED WIRING TIPS

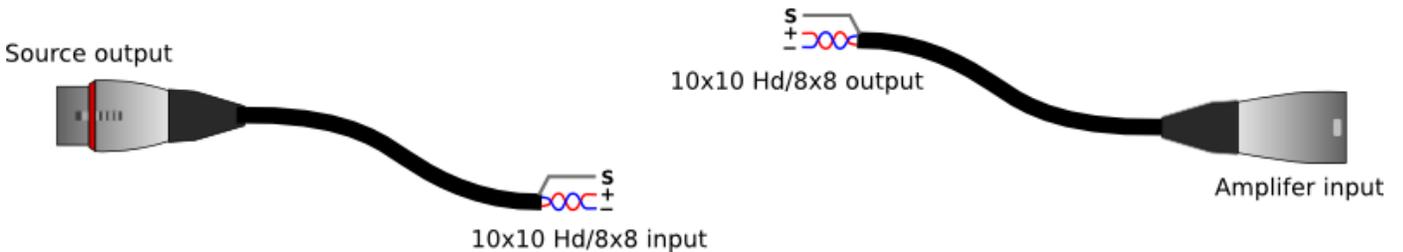
### 8.2.1 Phoenix terminal blocks

To make balanced connections, remove the terminal blocks from the rear of the processor. Connect individual wires from a shielded pair cable to each set of screw terminals as shown below. After all connections to the terminal block are secure, firmly re-insert the terminal blocks.



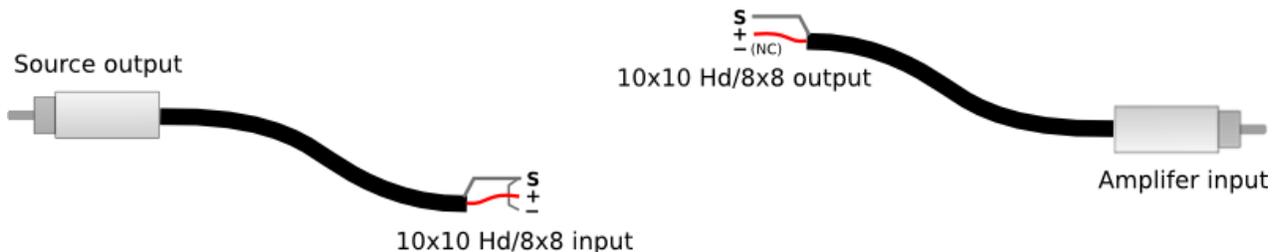
### 8.2.2 XLR adapters

If connecting the processor to equipment that uses XLR connectors, suitable cabling can often be made by cutting XLR-XLR or microphone cables in two. The two halves are then used for input and output, as shown here:



### 8.2.3 RCA adapters

In certain special cases, it can be helpful to connect a balanced input or output of the processor to an unbalanced (RCA) connection. The connection scheme is shown below. Note that the cable shield must be connected to the '-' terminal on the input side, whereas on the output side, the '-' terminal is not connected.



### 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.

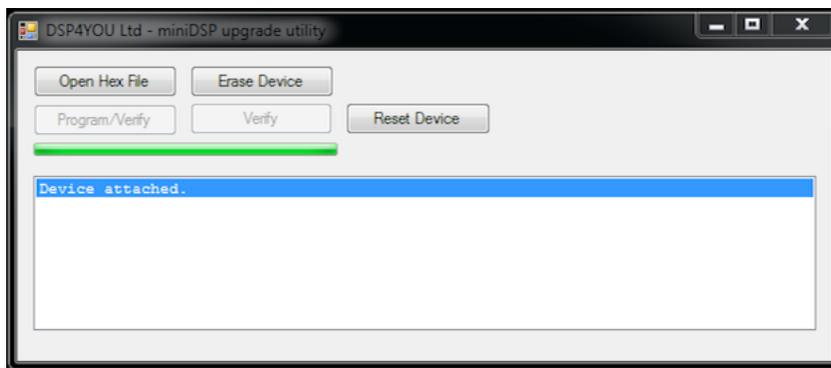
Item#	Symptoms	Troubleshooting recommendation
1	Cannot install software	a. Confirm that you downloaded and installed the required frameworks first (see <a href="#">Software Installation</a> ).
2	Software running in background but not showing	a. The Adobe Air environment may need a network connection 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 it. 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 control blocks are not enabled. f. Check that your crossover frequencies are correct e.g. that you don't have high pass and low pass frequencies incorrectly set. g. 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/s. 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 input meters and ensure that you are not overloading the inputs. If necessary, reduce the signal level from the source or change the input sensitivity jumpers. b. 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 the cabling from your source/s.</li> <li>c. 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 MCU FIRMWARE UPDATE

miniDSP may occasionally provide an update to the processor's MCU firmware to enable new features. To update the MCU firmware:

1. Download and install the latest **8x8** plugin from the **User Downloads** section of the miniDSP website.
2. Connect the processor to your computer via USB.
3. Start the **8x8** plugin (if it is not already running).
4. Click on the **Connect&Synchronize** button and select the **Synchronize** option from the dialog.
5. From the menus, select **Restore -> Upgrade Firmware**. The processor will be put into boot loader mode and the miniDSP upgrade utility will start. The status line should display "Device attached".



6. Click on the **Open Hex File** button and select the .hex file included in the download. It will have a name like **miniDSP\_2x8\_8x8\_v2.7\_release.hex**.
7. Click on the **Program/Verify** button. The status bar will indicate progress. **Do not disconnect or power off** the processor during the firmware upgrade!
8. Click on the **Reset Device** button. The processor will return to normal operating mode, and can be used as normal.



DO NOT DISCONNECT THE USB CABLE OR POWER FROM THE PROCESSOR WHILE FIRMWARE UPDATE IS IN PROGRESS. DOING SO MAY "BRICK" YOUR PROCESSOR.

## 8.5 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](http://minidsp.desk.com) with:
  - a. The specific product you are having an issue with (in this case, 10x10 Hd or 8x8 Kit).
  - b. A clear explanation of the symptoms you are seeing.
  - c. A description of troubleshooting steps (see [Troubleshooting](#) above) performed and your results.