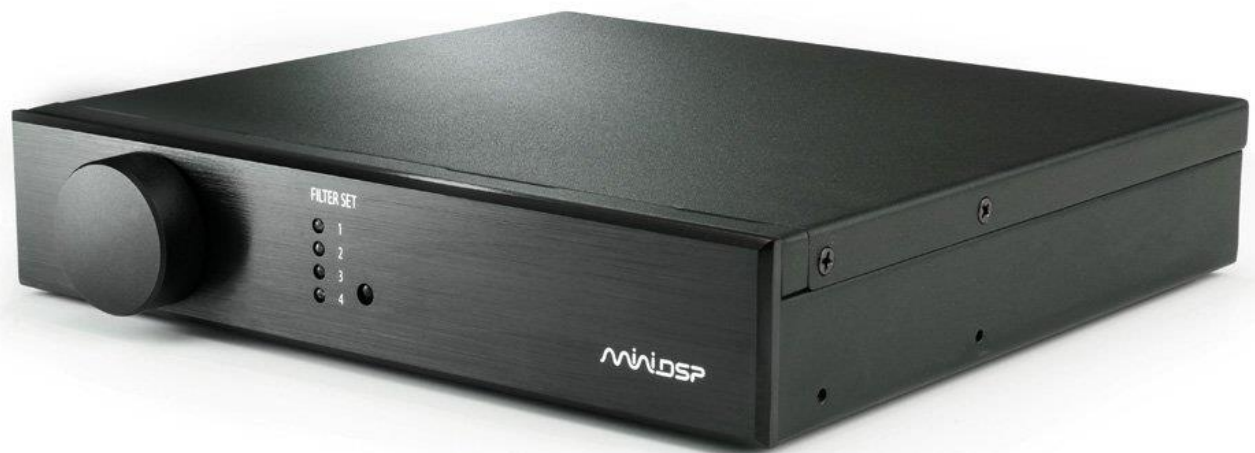# OPENDRC-DA8

2-IN 8-OUT HIGH-RESOLUTION AUDIO PROCESSOR WITH FIR/LINEAR PHASE CAPABILITY

User Manual

Revision history

Revision	Description	Date
V0.1	Preliminary version	21 August 2015
V1.0	First release version	8 September 2015

CONTENTS

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
- 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 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 OpenDRC-DA8 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 OpenDRC-DA8 audio processor. The miniDSP OpenDRC-DA8 is a powerful and flexible high-resolution[1] digital audio processor for active crossover and other audio processing applications. It features:

- Stereo digital S/PDIF input (via RCA connector) and eight analog output channels (RCA)

- Choice of 48 kHz and 96 kHz plugins

- 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, with taps per channel assignable up to a total of 9600 taps (48 kHz plugin) or 3400 taps (96 kHz plugin)

- Matrix mixer for flexible channel assignment and configuration

- Real-time configuration of all processing parameters

- Volume control by front panel or infrared remote

- Four on-board presets, selectable from front panel or by infrared remote

1.1 TYPICAL USAGE

In its most typical application, the OpenDRC-DA8 connects to a single digital source and directly drives power amplifiers in an active loudspeaker system. The onboard FIR filtering supports linear phase crossovers. Loudspeakers of up to 4-way (or three-way plus subs) are directly supported.

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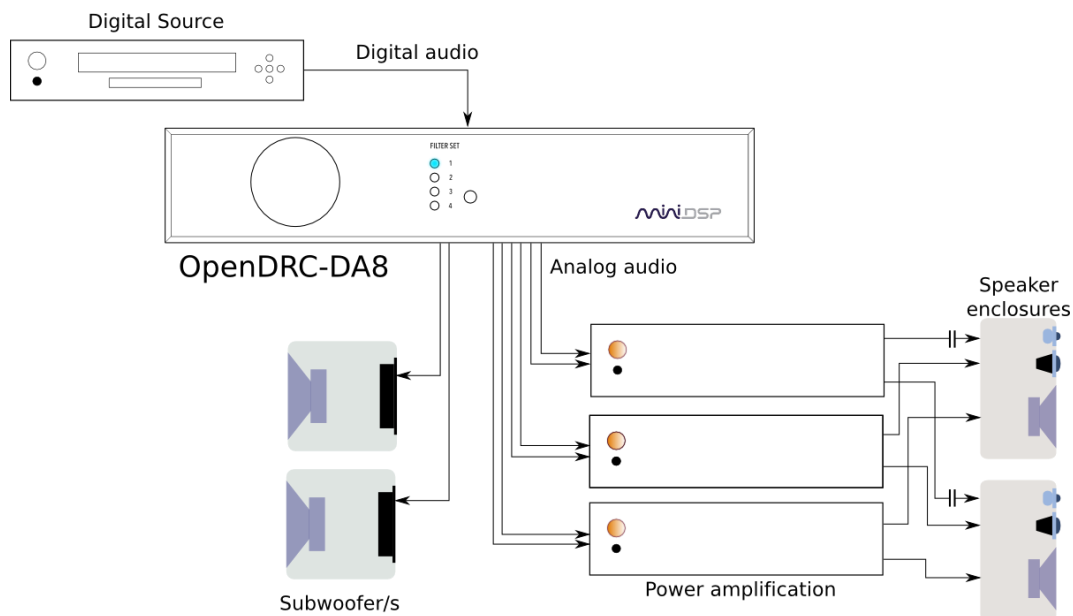
[1] With the 96 kHz plugin, the processor operates at 24-bit resolution and 96 kHz internal sample rate.

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.
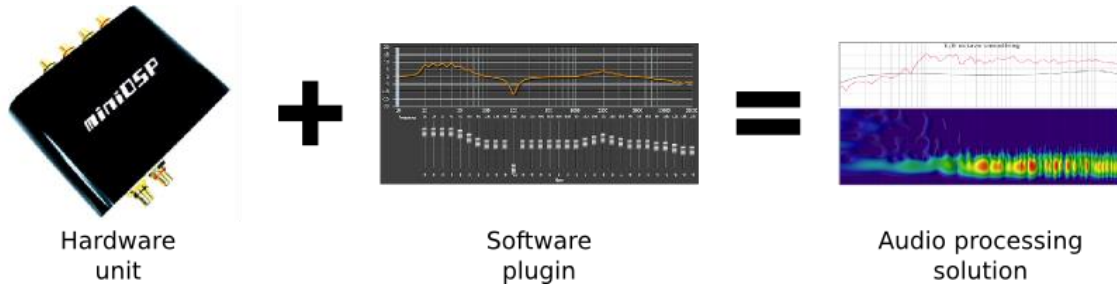
Hardware unit

> In this case, the hardware unit is the miniDSP OpenDRC-DA8.

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. Currently, the OpenDRC-DA8 supports two plugins, as described below.

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](#).

1.3 CHOOSING A PLUGIN

Two plugins are currently available for the OpenDRC-DA8: the **miniSHARC 4x8** and the **miniSHARC 4x8 96k**. The table below summarizes the key differences between these plugins when used with the OpenDRC-DA8 (some parameters are the same, but are included to show that they do not change with the plugin):

	miniSHARC 4x8 plugin	**miniSHARC 4x8 96k** plugin
Number of digital input channels[1]	2	2
Number of analog output channels	8	8
Digital input sample rate	20 to 216 kHz	20 to 216 kHz
Internal sample rate	48 kHz	96 kHz
Total number of FIR taps	9600	3400
Number of PEQ bands per block	10	10
Compressor block	Yes	Yes
Maximum per-channel time delay	3000 ms	1500 ms

*1 two channels = single stereo digital signal

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 plugin

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

2. Familiarize yourself with the plugin

> Before connecting your computer to the processor, it's worth starting the plugin and familiarizing yourself with its user interface. Click on and explore each of the tabs (**Input**, **Routing**, and **Output**). For more information, see Configuring the processor starting on page 16.

> At this time, the plugin is still in *offline* mode, so any changes you make will not be downloaded into the miniDSP hardware. Since 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 processor 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 OpenDRC-DA8 into your system. *Ensure that all power is turned off when making audio connections*. See Hardware connectivity on page 13.

miniDSP

7. Go into online mode

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

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

For more information, see Synchronizing with the processor on page 15.

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 OpenDRC-DA8 down low (view the Master Volume display in the plugin while turning the knob counter-clockwise) and start playing music or a pink noise test signal. Gradually increase the volume until your 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 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

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:
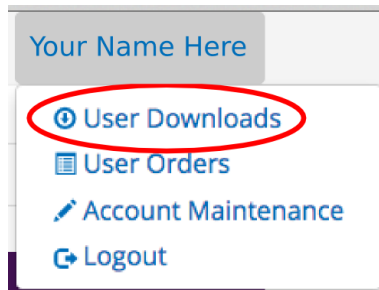
Navigate to the **OpenDRC Plug-ins** section of User Downloads, then download the zip file (or files) under the heading **miniSHARC 4x8 48k** and/or **miniSHARC 4x8 96k**. Double-click on it to unzip it. Then follow the installation procedure below according to your computer type.

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.
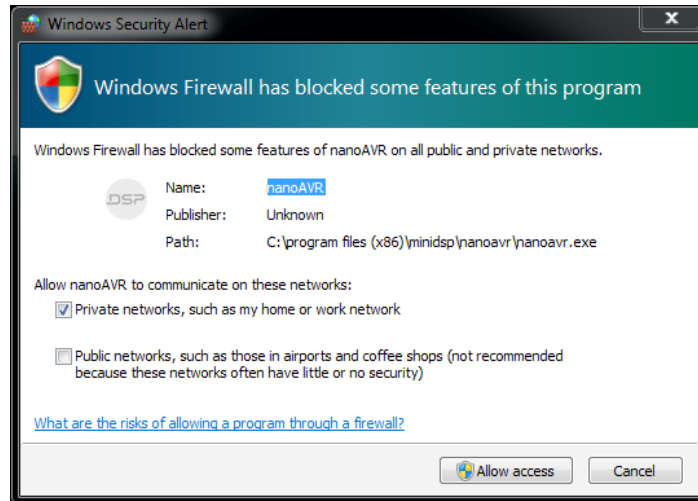
3.1.1   Windows

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.
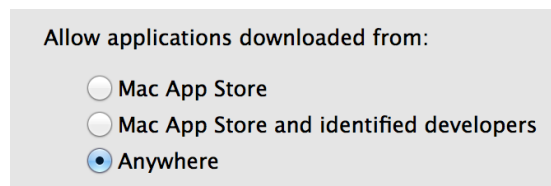
To install the miniDSP software, open the **Windows** folder of the download and double-click on the **miniSHARC_4x8.exe** or **miniSHARC_4x8_96k.exe** program. We recommend accepting the default installation settings. Once installation is complete, the plugin will automatically start. Since the plugin 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 **miniSHARC-4x8.dmg** of **miniSHARC-4x8_96k.dmg** file to open it in a new window. Then double-click on the installer program, **Install miniSHARC 4x8.app** or **Install miniSHARC 4x8 96k.app**. Once installation is complete, the plugin will automatically start.

3.2 HARDWARE CONNECTIVITY

All connections to the OpenDRC-DA8 are made on the rear panel.

3.2.1 Digital input

Connect a single digital source to the S/PDIF coax (RCA) connector.
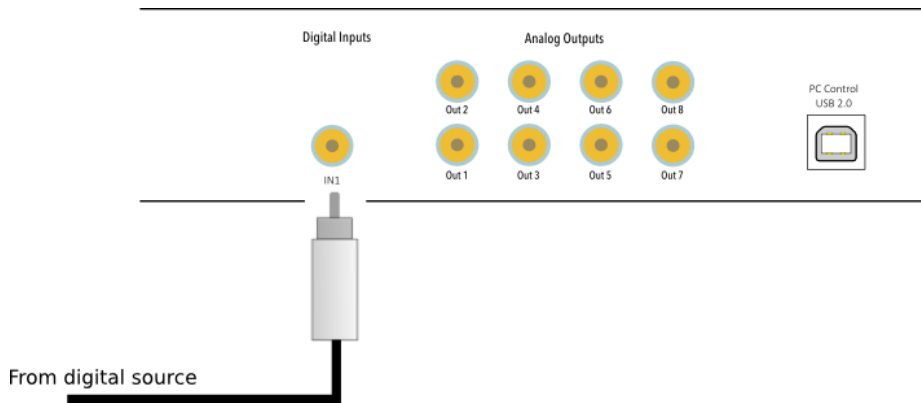
Note: the digital input accepts only a stereo PCM digital signal. It does not accept encoded or multichannel digital audio (such as Dolby Digital or DTS).

3.2.2 Analog outputs

There are eight analog output channels. Unbalanced connections are made directly from the RCA jacks to power amplifiers. Be sure to take careful note of the channel numbering shown in this diagram and on the rear panel.
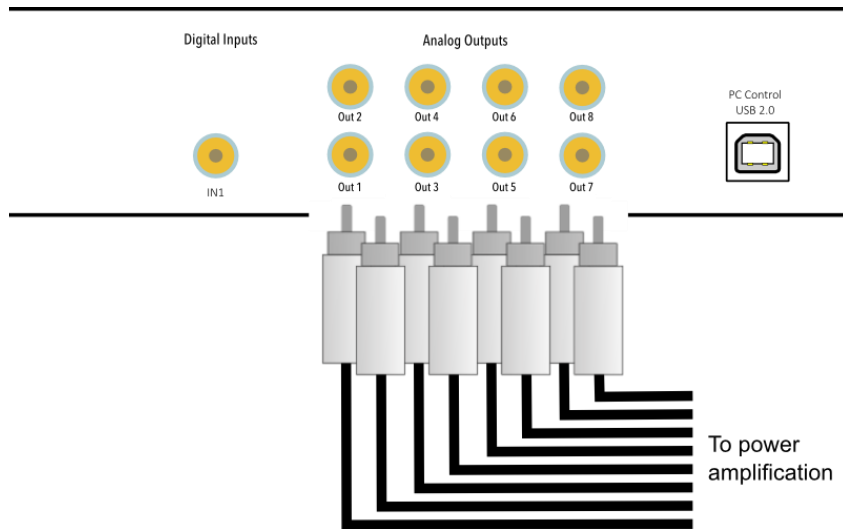
3.2.3   DC power

The supplied 5 VDC power supply includes a set of interchangeable power pins. Fit the correct pins for your country. Connect the DC plug to the 5 VDC power socket.
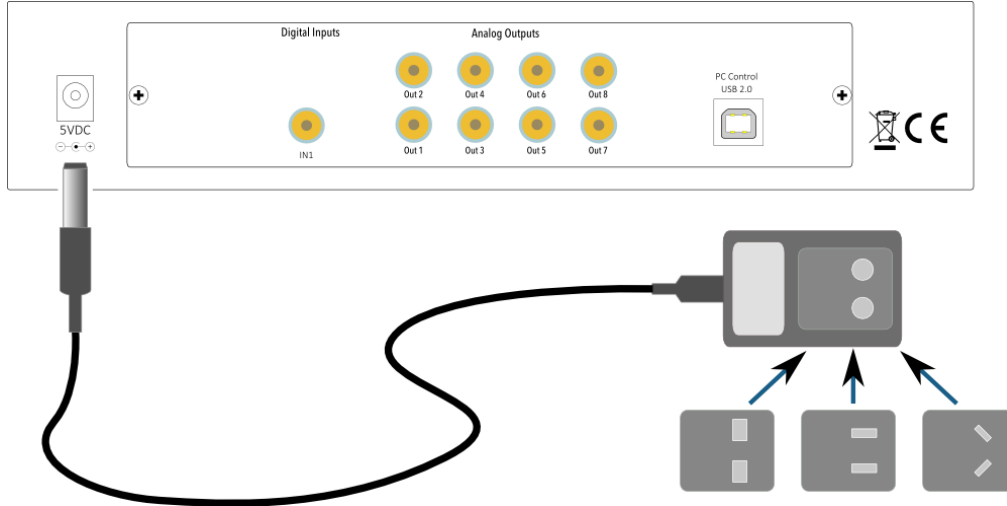

 Apply power to the processor only after all input and output connections have been made. The processor can usually be left powered on. If powered on and off, the following sequence is recommended:

- On: Power on line-level equipment, including the OpenDRC-DA8, then turn on power amplification.
- Off: Turn power amplification off, then power off line-level equipment, including the OpenDRC-DA8.

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.
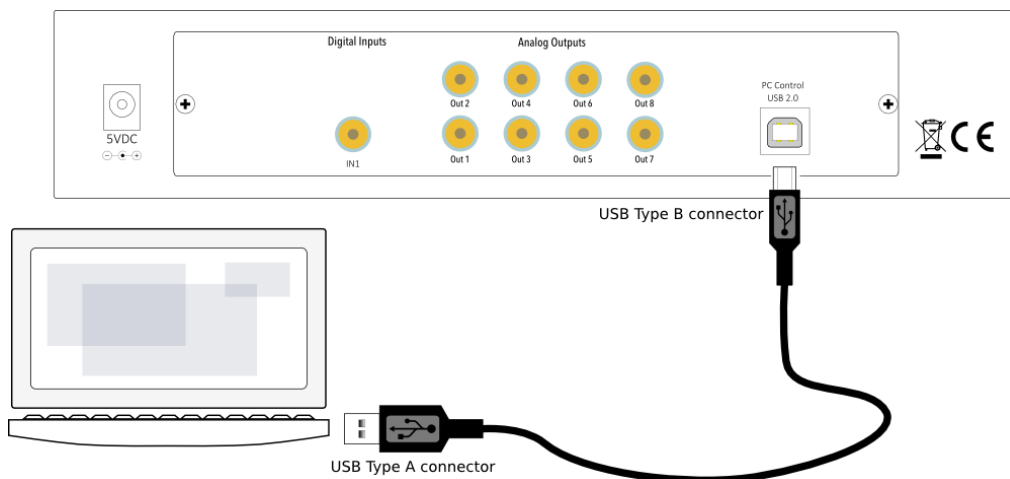
3.3 SYNCHRONIZING WITH THE PROCESSOR

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:
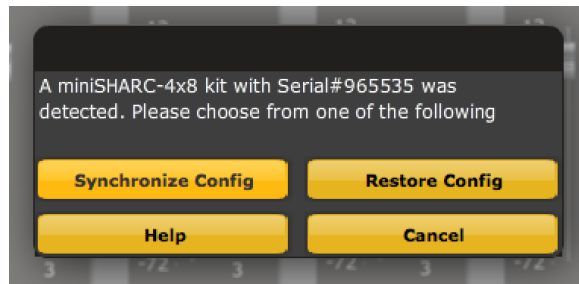
The options are:

Synchronize 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.

Cancel

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

4 CONFIGURING THE PROCESSOR

The OpenDRC-DA8 processor is configured with the **miniSHARC 4x8** or **miniSHARC 4x8 96k** 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 OpenDRC-DA8. If desired, however, the plugin can remain connected during use for real-time ("live") control of all audio processing.

This screenshot shows the **miniSHARC 4x8** plugin with the key areas highlighted:
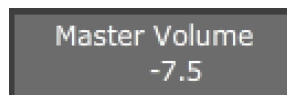

⚠️ During *initial* configuration of the processor, it is strongly recommended that any connected amplification be muted or powered off.

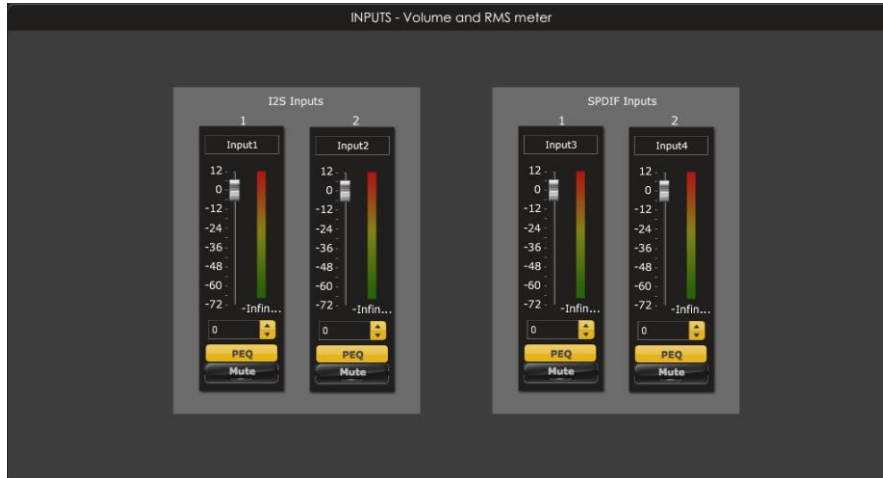The **Mute** button disables all audio output:

The Master volume display shows the current volume setting:

4.1 INPUT TAB

The **Input** tab displays a row of input channel control strips. When the **miniSHARC 4x8** plugin is used with the OpenDRC-DA8, only the two rightmost channels (labeled **SPDIF Input**) receive signal:

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

> 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 <u>Parametric EQ</u> on page 23.

Mute

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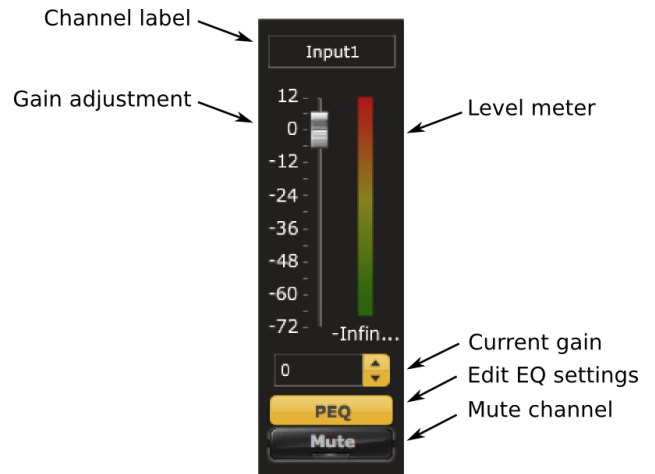
4.2 ROUTING TAB

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.)

When the **miniSHARC 4x8** and **miniSHARC 48 96k** plugins are used with the OpenDRC-DA8, only the SPDIF inputs are active. We recommend that the I2S inputs be unrouted, as shown here:

To make the routing matrix easier to view, the labels of the input and output channels can be changed in the Input and Output tabs, resulting in:

The above example is a typical routing matrix for 4-way loudspeaker. If more than one input signal is routed to an output channel, those channels are mixed together (summed). Here is another common routing matrix, for a stereo 3-way crossover and a pair of summed subwoofer outputs:

![miniDSP logo]

4.3 OUTPUT TAB

The **Output** tab displays a row of output channel control strips. All channels are identical.
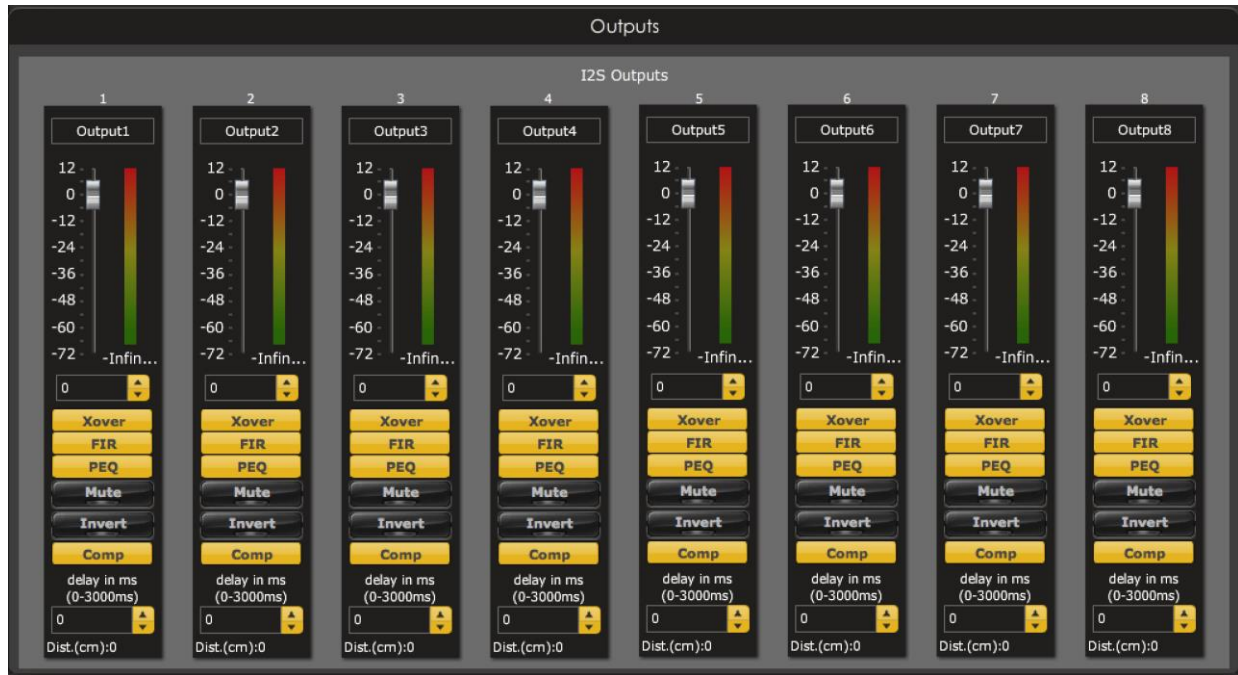
4.3.1 Channel strip layout

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

4.3.2 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** tab. To change the label, click on it, type a new label (up to eight characters), and press the Return key.
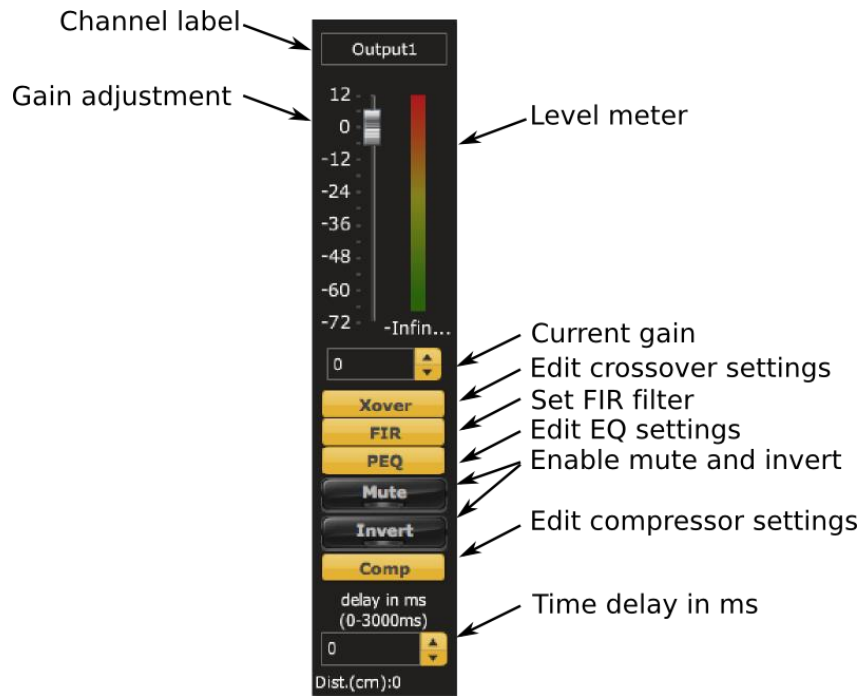
4.3.3 Gain control and level monitoring

Level meter

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.)

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.

4.3.4   Crossover

Each output channel has independent high pass and low pass filters. Click on the **Xover** button to open the crossover settings window:
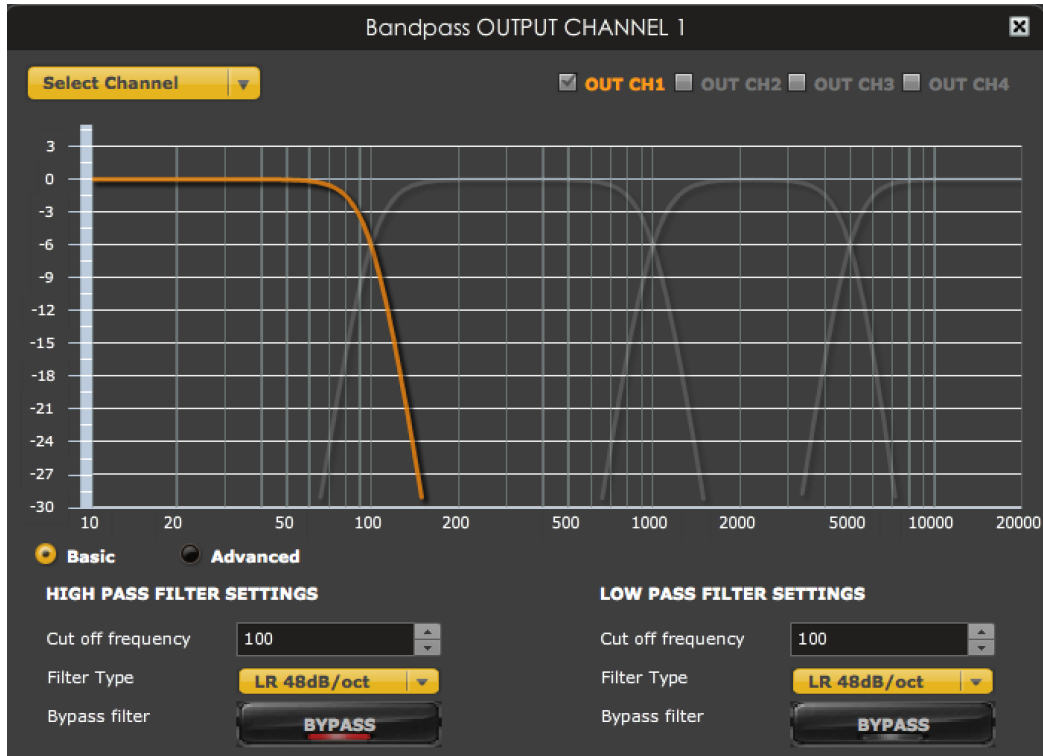
Crossovers "split" the frequency band to send to different drivers. In a two-way loudspeaker, for example, a *low pass* filter is used to remove high frequencies from the signal sent to the woofer, and a *high pass* filter is used to remove low frequencies from the signal sent to the tweeter. In a three-way 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-excursion.

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

Crossover filters are displayed in two groups of four—channels 1 through 4, and channels 5 though 8. Each group is configured by default as a four-way crossover. The current channel is displayed in orange, with the other three in the group in grey.

Hovering the mouse over the curve brings up an overlay showing the frequency and the attenuation at that frequency.
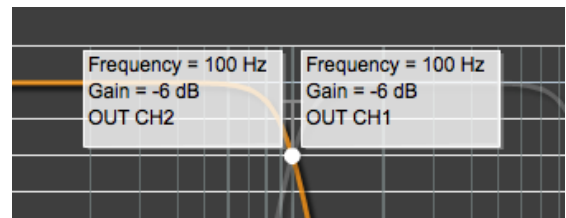
Basic/Advanced

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 27.

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.

4.3.5 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 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.

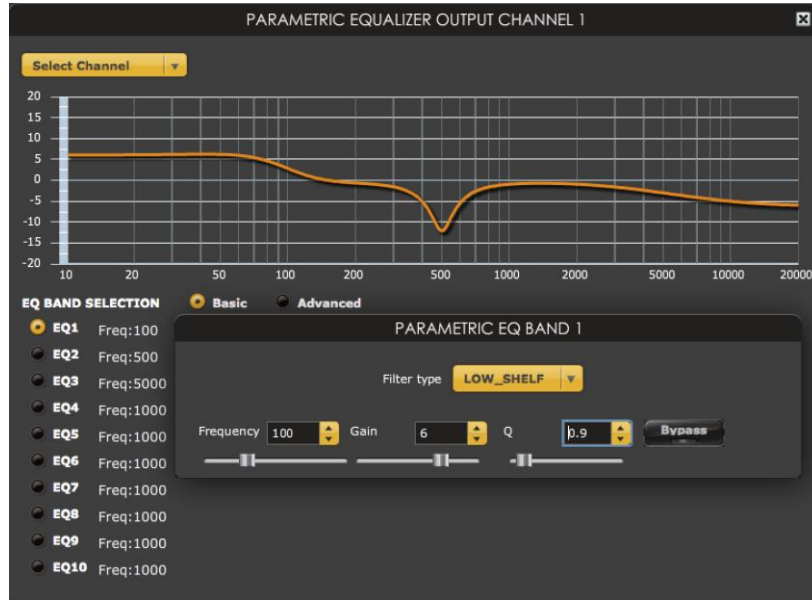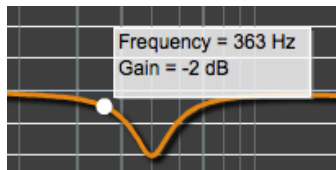Hovering the mouse over the curve brings up an overlay showing the frequency and the gain at that frequency.

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 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 <u>Custom biquad programming</u> on page 27.

Filter type

Selects the type of filter:

PEAK   Create a dip or a peak in the frequency response.

LOW_SHELF   Reduce or increase part of the frequency spectrum *below* a given frequency.

HIGH_SHELF   Reduce or increase part of the frequency spectrum *above* a given frequency.

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

Frequency

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

Gain

For the PEAK and SUB_EQ filter types, this is the gain in dB at the center frequency. For the HIGH_SHELF and LOW_SHELF filter types, this is the gain in dB reached at high or low frequencies respectively. A filter has no effect if its gain is set to 0 dB. Gain can be adjusted in increments of 0.1 dB up to +/- 16 dB.

Q

Q controls the "sharpness" of the filter. For the PEAK and SUB_EQ filter types, lower Q gives a broader peak or dip, while higher Q gives a narrower peak or dip. For the HIGH_SHELF and LOW_SHELF filter types, Q controls how quickly the filter transitions from no gain to maximum gain.
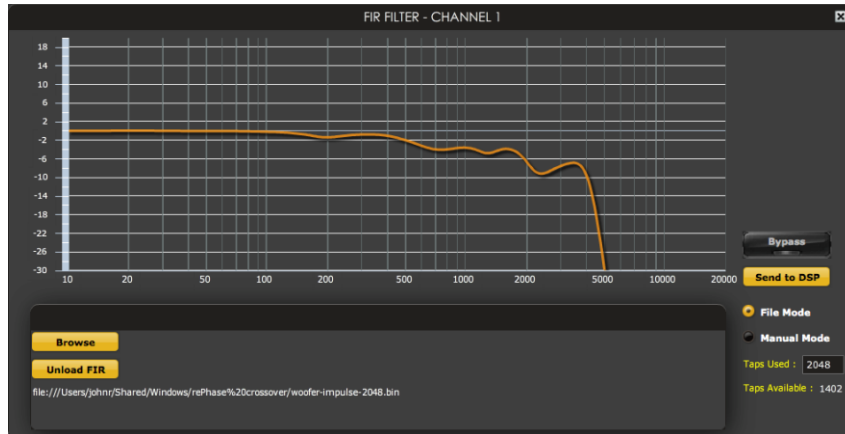
Bypass

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

4.3.6   FIR

Each output channel has a variable-tap FIR filter. Click on the **FIR** button to open the FIR filter settings window:

FIR filtering is a powerful feature that allows very complex filters to be constructed (with the aid of suitable design software). These filters can correct for amplitude only ("linear phase filters"), phase only, or a combination or both. FIR filtering is described in detail in <u>FIR filtering and design</u>, starting on page 30.

4.3.7   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":

4.3.8   Time delay

A delay of up to 1500 or 3000 ms (depending on the plugin) can be applied to each output channel. To set the delay, click in the delay entry box for a channel. The delay value can be entered numerically, and the up and down arrows can be used to change the delay in small (0.02 ms) increments.

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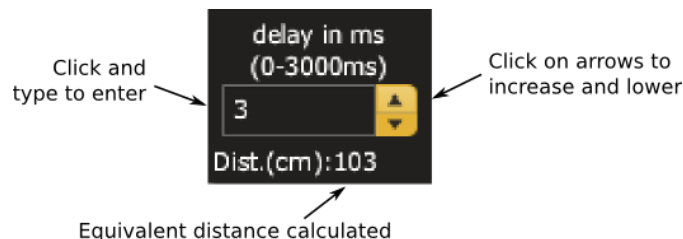
4.3.9 Compressor

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 abov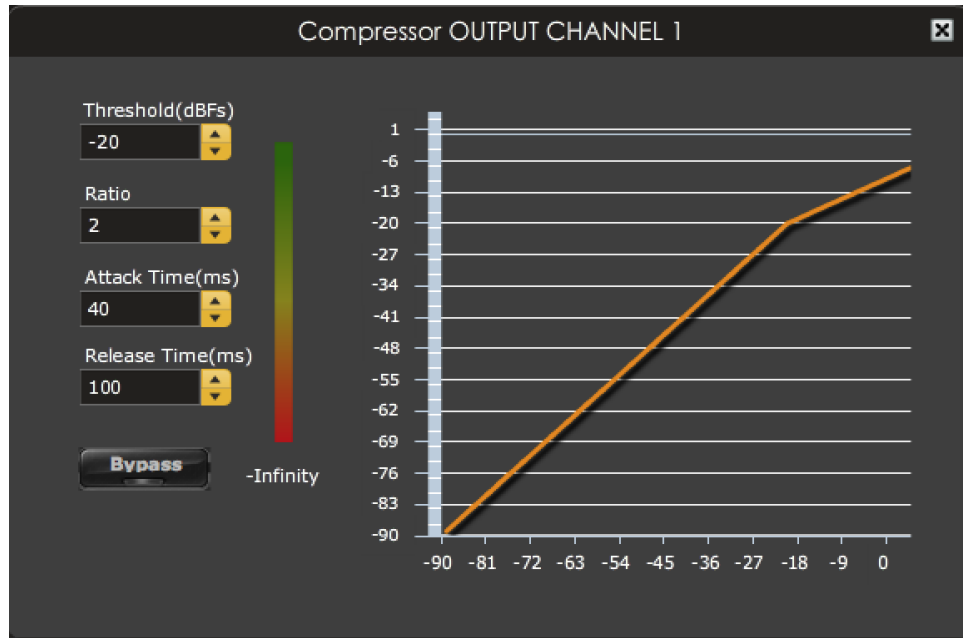e the threshold, according to the **Ratio** parameter. This can be used to limit the power delivered to speakers and thus reduce the risk of damage from overdriving.

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.

4.4 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.

4.4.1 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:

$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}$$
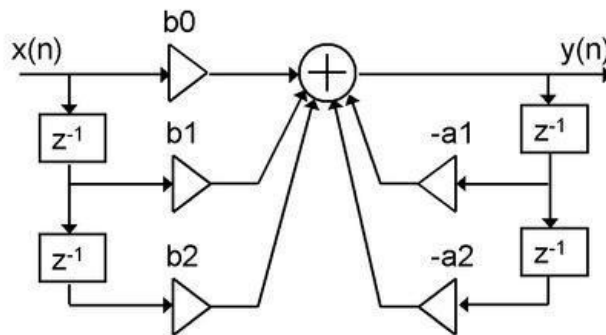
A single biquad like this can perform a great many functions, including all of the functions of a single parametric EQ filter, one 6 or 12 dB/octave high pass or low pass filter, and more. Biquads are combined in series (cascaded) to create more complex filters. The function that each biquad performs is determined by just five numbers: *a1*, *a2*, *a0*, *b1*, and *b2*. These numbers are called the *coefficients*.

4.4.2 Using custom biquad programming

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

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

Parametric EQ advanced mode

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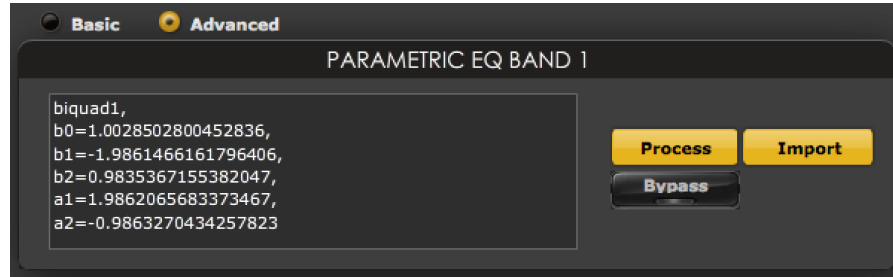
Parametric EQ file import (REW integration)

Multiple biquads in the parametric EQ block can be set at once by importing a coefficient file. This file can be generated by Room EQ Wizard (REW) or by other programs. The design program must be set for a **48 kHz** sample rate if using the **miniSHARC 4x8** plugin, and a **96 kHz** sample rate if using the **miniSHARC 4x8 96k** plugin. 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,
...
biquad5,
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. (Be careful: 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 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.
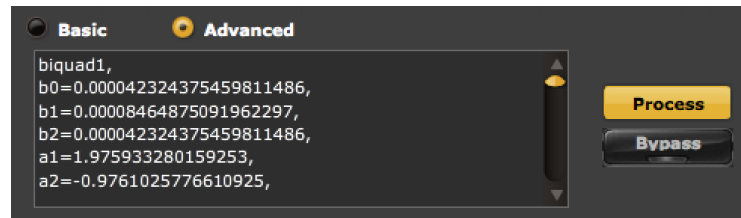
4.4.3 Biquad design software

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

4.4.3.1 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

4.4.3.2 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.

4.4.3.3 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

4.5   FIR FILTERING AND DESIGN

FIR filtering is a powerful and advanced feature of the OpenDRC-DA8. It allows construction of complex arbitrary equaliztion and crossover filters with independent control of amplitude and phase. The parameters of each FIR filter are set in the **FIR** settings window:
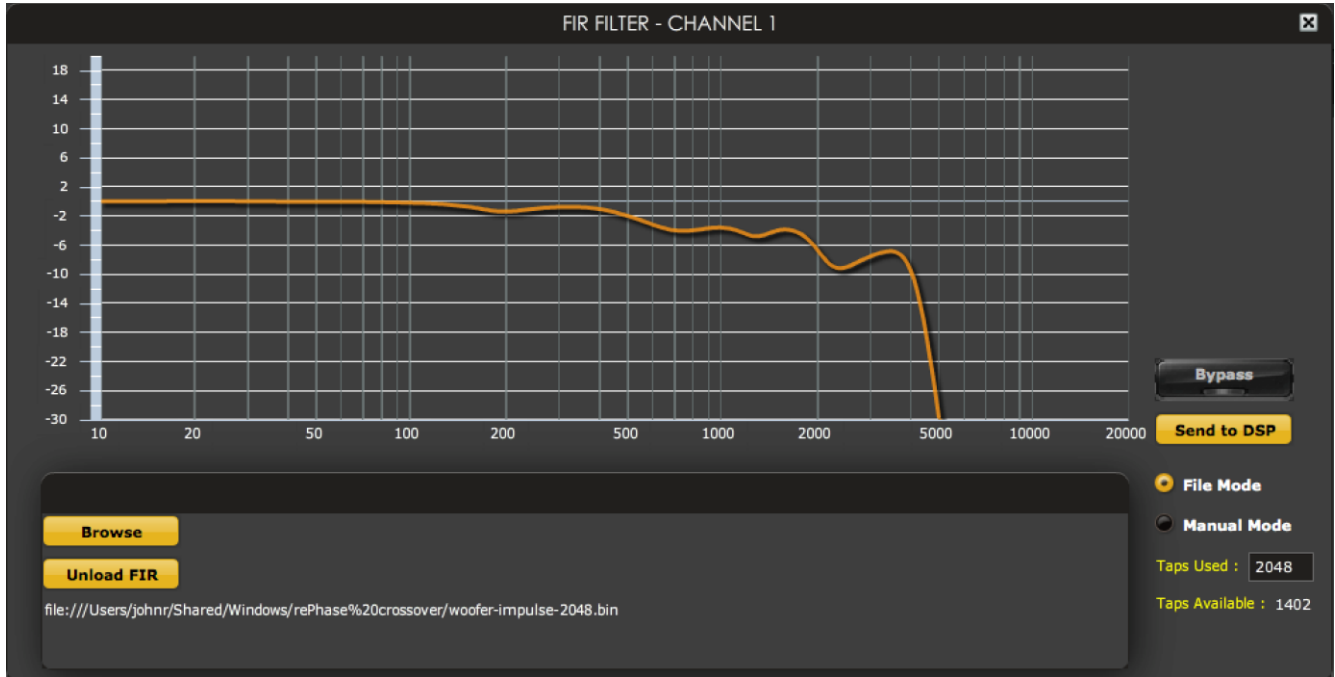
Browse   Opens a filter browser to select a file containing FIR filter coefficients. (See FIR filter file format below).

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

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

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

File Mode / Manual Mode

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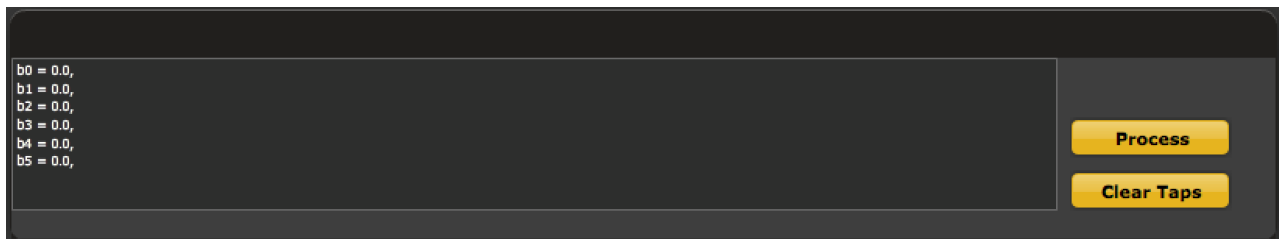
4.5.1   FIR filtering overview

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 OpenDRC-DA8 can compute 9400 taps when run at 48 kHz, and 3400 taps when run at 96 kHz. These taps can be distributed as you wish across the eight 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 : 2048

Taps Available : 1402
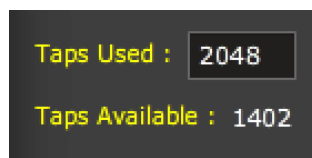
4.5.2   FIR filter design software

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.

4.5.3   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,
```

4.5.4   Loading filter coefficients

In **File Mode**:

1.   Click **Browse**, navigate to the file containing the filter coefficients, and open it. A dialog will appear confirming the number of coefficients loaded.

2.   Confirm that the response curve is as you expect.

3.   Press **Send to DSP**. This will write the coefficients into the DSP's memory.

4.   To clear the filter coefficients, click **Unload FIR** and then **Send to DSP**.

In **Manual Mode**:

1.   Cut and paste the coefficients from the text output of the design program.

2.   Press the **Process** button.

3.   Confirm that the frequency response graph is as you expect.

4.   Press **Send to DSP**. This will write the coefficients into the DSP's memory.

5.   To clear the filter coefficients, click **Clear Taps** and then **Send to DSP**.

 If, after selecting a filter file or setting coefficients, the frequency response graph does not change as expected, make sure that the Bypass button is turned off.
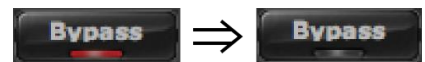
 If you clear the filter taps, make sure that you also bypass the filter, otherwise there will be no audio through that channel.
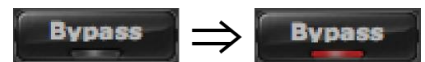
4.6 WORKING WITH CONFIGURATIONS

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:

Configuration Selection: Config 1 Config 2 Config 3 Config 4

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

The plugin is in online mode

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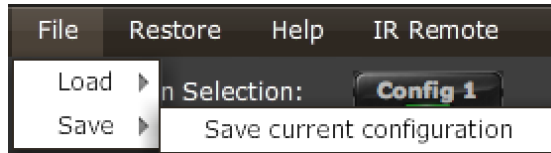
The plugin is in offline mode

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

4.6.3 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**.
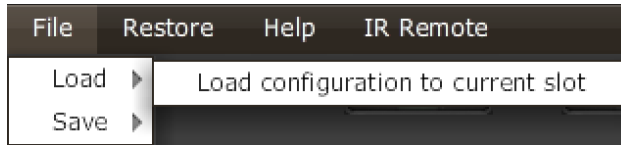
If the plugin is in online mode, the new configuration data will be downloaded to the processor immediately. If the plugin is in offline mode, the configuration will be loaded into the user interface only, and will be downloaded to the processor the next time it is synchronized.

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.

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, **t**he 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 **miniSHARC 4x8** 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 OPENDRC-DA8

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)
- Preset selection
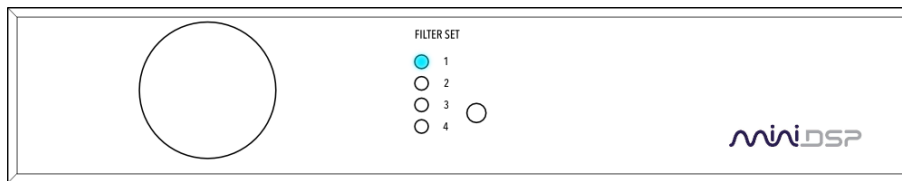
5.1 STATUS INDICATORS

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

To change the volume

> Rotate the control knob clockwise to increase the volume, and counter-clockwise to decrease it. If the computer is connected and the plugin is in online mode, the Master Volume display in the plugin updates accordingly. (This is not necessary for normal operation, but can help with troubleshooting if needed.)

**To change the selected filter set**

> Briefly press the control knob. The **Filter Set** LED blinks quickly. Rotate the control knob until the desired filter set LED is blinking. Press the control knob again, and the selected LED will remain steady.

5.3 INFRARED REMOTE CONTROL

Many standard and programmable remote control units can be used with the OpenDRC-DA8. By default, the OpenDRC-DA8 recognizes the Apple Remote, with the key assignments shown at right.

(The commands **Config Inc** and **Config Dec** change the selected configuration preset up or down. For example, if the currently selected preset is **1**, pressing **Config Inc** changes it to **2**.)
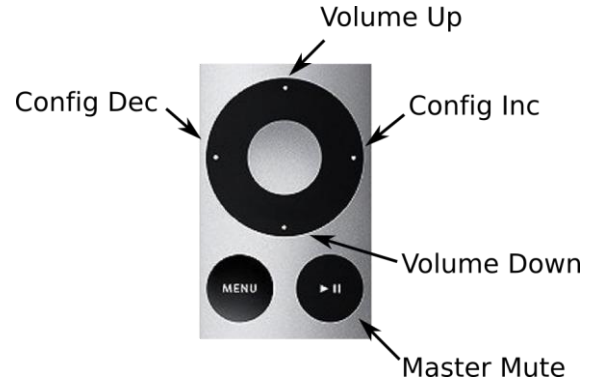
In addition, the OpenDRC-DA8 can "learn" the control codes of your current remote if it supports one of the following remote control codes:

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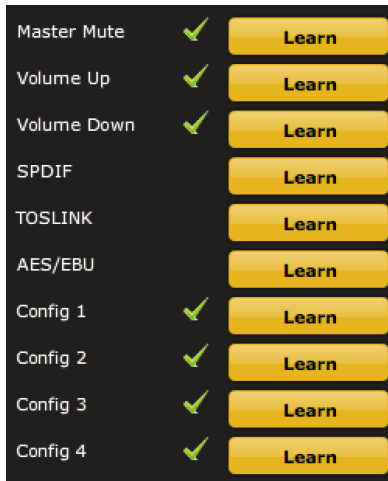
To "unlearn" a command, press the **Learn** button and wait for the plugin to time out.

6 ADDITIONAL INFORMATION

6.1 SPECIFICATIONS

Computer connectivity   Driverless USB 2.0 control interface for Windows and Mac OS X

Digital audio Input   S/PDIF on RCA connector / Isolated with digital audio transformer

A high quality onboard Asynchronous Sample Rate Converter ensures compatibility with most sample rates, from 20–216kHz.

Analog audio outputs   8 x outputs on unbalanced (RCA) connectors, 2.0V RMS full scale output

Audio resolution   24-bit input and output, 48 kHz internal sample rate with the **miniSHARC 4x8** plugin, 96 kHz internal sample rate with the **miniSHARC 4x8 96k** plugin.

Audio processing   32-bit floating-point processor

Storage/presets   All settings controllable in real time from software user interface.

Up to 4 presets stored in local flash memory.

Infrared remote   "Learning remote" capabilities (NEC, Philips, Sony)

Controls master volume, mute, digital input selection, preset selection.

Power supply   5 VDC single supply @ 600mA, 2.1 mm center-positive

Dimensions (H x W x D)   41.5 x 214.5 x 206 mm

6.2 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 matrix mixer is sending the correct inputs to the correct outputs.
5	No signal showing on input meters	a. Check the cabling from your source. b. Check that your source is playing audio and that it is not muted or have volume control turned down. c. Check that the plugin is synchronized with the hardware unit.
6	Audio sounds distorted	a. Check the output meter and ensure that you are not overloading the outputs. If necessary, reduce the output gain and/or the amount of boost in the EQ blocks.
6	Audio is coming through the wrong outputs	a. Check the cabling from the processor to your amplifiers. b. Check that you have correctly set up the matrix mixer 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.

6.3 MCU FIRMWARE UPDATE

miniDSP may occasionally provide an update to the processor's MCU firmware to enable new features. Currently, firmware upgrade is supported from the Windows platform only.

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

To update the MCU firmware:

1. Download and install the latest **miniSHARC 4x8** plugin from the **User Downloads** section of the miniDSP website.
2. Connect the processor to your computer via USB.
3. Start the **miniSHARC 4x8** 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".
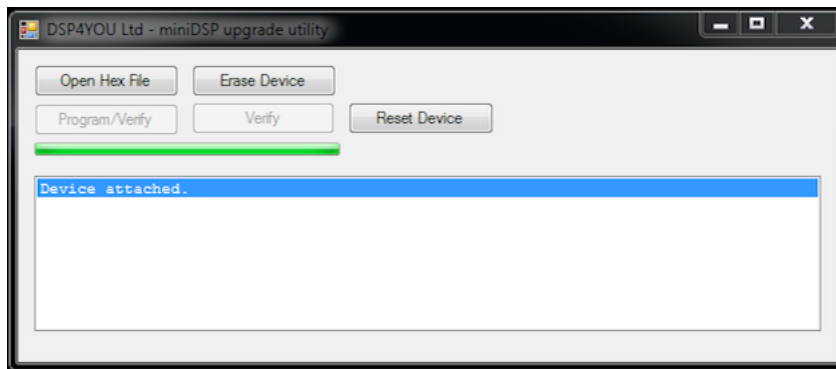
6. Click on the **Open Hex File** button and select the .hex file included in the download. It will have a name like **ENC_miniSHARC_ver2.17.hex**.
7. Click on the **Program/Verify** button. The status bar will indicate progress. **Do not disconnect or power off** the processor during the firmware upgrade!
8. After the status indicates that the verify has completed successfully, click on the **Reset Device** button, and then quit the upgrade utility.
9. Return to the **miniSHARC 4x8** plugin. (If there is a dialog informing "Connection to DSP closed," click on **OK**.) Click on the **Connect** button, and then on **About this product** from the **Help** menu to verify the new firmware version.

6.4 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, OpenDRC-DA8).

    b. A clear explanation of the symptoms you are seeing.

    c. A description of troubleshooting steps (see Troubleshooting above) performed and your results.