

# HARMONY DSP 8x12

8-IN 12-OUT ADVANCED IN-CAR AUDIO PROCESSOR AND POWER AMPLIFIER
WITH DIRAC LIVE® ACOUSTIC CORRECTION

# **User Manual**





## **Revision history**

Revision	Description	Date
0.1	First draft	10 November 2020
1.0	First public version	30 November 2020
1.1	Minor updates and corrections	16 December 2020
1.2	Adobe AIR and Flash no longer needed	1 February 2021
1.3	Added Application Guide, Big Sur supported	5 August 2021
1.4	Additional information on Line Out mode	3 February 2022
1.5	Minor installation update	18 May 2022

## **SUPPORTED OS**







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## **IMPORTANT INFORMATION**

Please read the following information before use. In case of any questions, please contact miniDSP via the support portal at <a href="mailto:support.minidsp.com">support.minidsp.com</a>.

## **System Requirements**

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

#### Windows

- Microsoft• <sup>®</sup> Windows<sup>®</sup> 10 or 11, latest version with all updates
- At least a dual core i3, i5, or i7 processor
- At least 2 GB RAM (4 GB preferred)
- Two free USB 2.0 ports
- Internet connection

#### macOS

- macOS 10.14 Mojave or later, latest version with all updates
- At least a dual core i3, i5, or i7 processor, or an ARM processor (M1/Pro/Max)
- At least 2 GB RAM (4 GB preferred)
- Two free USB 2.0 ports
- Internet connection

## **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 <u>application notes</u> 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.

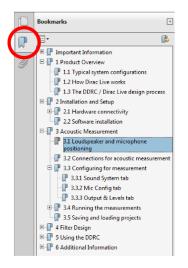
## **CE Mark Statement**

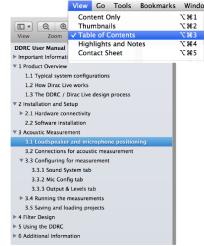
The Harmony DSP 8x12 has passed the test performed according to European Standard EN 55022 Class B.

#### 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.

For reading on the computer, we have included hyperlinked cross-references throughout the manual. In addition, a table of contents is embedded in the PDF file. Use the View menu (Preview on Mac) or the bookmarks sidebar (Adobe reader on Mac and Windows) to view this table of contents.





Adobe Reader on Windows

Preview on Mac



## 1 PRODUCT OVERVIEW

Thank you for choosing the miniDSP Harmony DSP 8x12 advanced in-car audio processor and power amplifier. The Harmony DSP 8x12 features stereo optical and 6-channel USB digital inputs, 12 speaker- or line-level analog outputs, a full eight channels of Dirac Live room correction and miniDSP's powerful audio processing on each output channel.

By use of the onboard matrix mixers, the Harmony DSP 8x12 is adaptable to many configurations, ranging from a simple stereo correction and EQ system, through to integration of one or more subwoofers, through to a complete active multichannel surround system. Low-noise digital-input line/power drivers ensure pristine audio quality in any vehicle environment.

Four complete processing configurations are stored on-board and can be selected from the OLED wired remote or by infrared remote control.

### 1.1 TYPICAL USAGE

Figure 1 illustrates a straightforward installation, with two or four channels supplied by the source device and four speakers being driven by a power amplifier. The included miniDSP UMIK-1 is used to perform Dirac Live calibration.

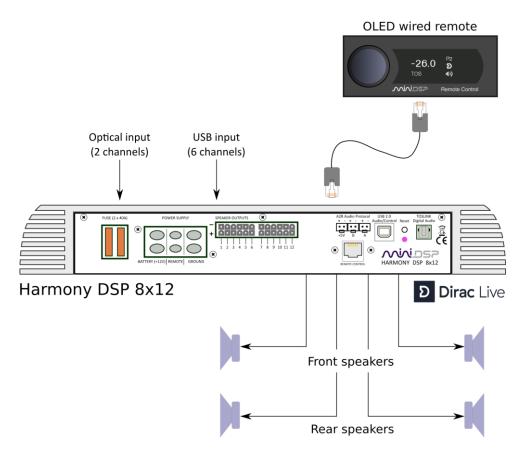


Figure 1. Basic system setup with four-channel acoustic correction



The Harmony DSP 8x12 also supports more advanced configurations. With 12 powered output channels, implementing an in-vehicle active speaker system has never been easier, and the Dirac Live correction ensures optimum response all around. Some features include:

- Comprehensive bass management function to synthesize the subwoofer feed from speaker channels.
- Up to 8 channels of Dirac Live correction with flexible assignment to output channels.
- Rear channel synthesis for stereo sources.
- Multiple onboard configurations for different situations (with/without passengers, competition etc.).

Figure 2 illustrates a more complex installation that uses more of the features of the Harmony DSP 8x12.

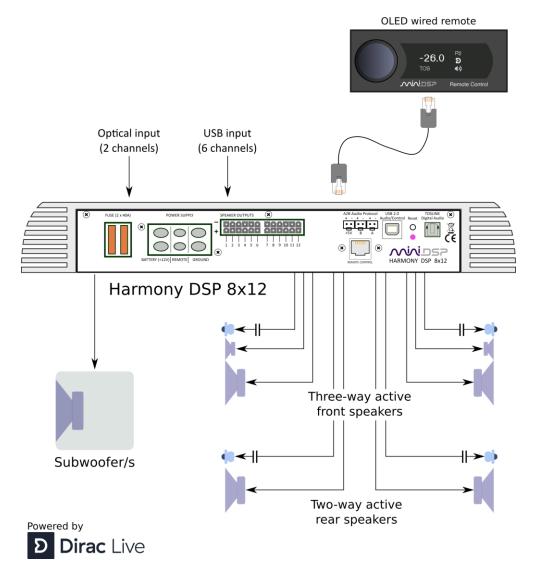


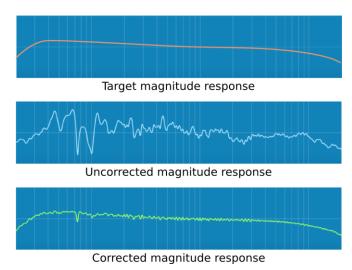
Figure 2. Advanced system configuration with subwoofer(s) and active speakers



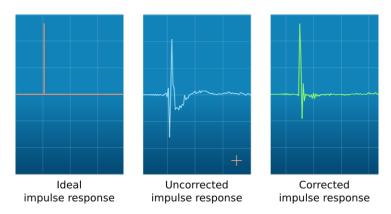
## 1.2 DIRAC LIVE

The Harmony DSP 8x12 executes Dirac Live® digital room correction, from <u>Dirac Research</u>. This mixed-phase filtering technology will minimize the effects of cabin modes and resonances, adjust frequency response to optimum, and improve imaging, dynamics and clarity.

To accomplish its remarkable improvement in listening quality, the **DiracLive** application steps you through the procedure for taking measurements around your listening area. Dirac Live® employs a sophisticated analysis algorithm to make the optimal correction across the *whole* listening area, not just at a single point. The user has full control over the target response. Measurements are taken with a calibrated acoustic measurement microphone, the miniDSP UMIK-1.



In addition to correcting frequency response, Dirac Live® corrects the system's *impulse response*, which reflects how the system responds to a sharp transient such as a drumbeat. Reflections, diffraction, resonances, misaligned drivers, and so on, all combine to smear out the transient. Correcting the impulse response makes the speaker behave much more like an ideal speaker.





Dirac Live calibration is described in the separate miniDSP Dirac Live User Manual.



## **2** SOFTWARE INSTALLATION

The Harmony DSP 8x12 is configured with software running on a PC or Mac. There are two sets of software to download and install, from <u>dirac.com</u> and from <u>miniDSP.com</u>.

### 2.1 WINDOWS

This section describes software download and installation for Windows 10 or Windows 11.



The software described in this section runs on Windows 10 or Windows 11 **only**. Other versions of Windows are not supported by the current version of Dirac Live.

## 2.1.1 Download and install the DiracLive application

Download the **DiracLive** application for Windows from <a href="https://www.dirac.com/live/downloads/">https://www.dirac.com/live/downloads/</a>.

Double-click on the downloaded installer to run it. *Be sure to accept the default installation settings*. Do not run the application yet.

#### 2.1.2 Download and install the miniDSP software

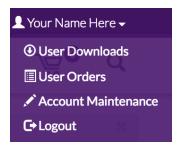
#### 2.1.2.1 Download from the miniDSP website

If you purchased your processor directly from miniDSP, your software will be available from the <u>User Downloads</u> section of the miniDSP website when your order ships. To access the download, you will need to be logged into the website with the account you created when purchasing.

If you purchased your processor from a miniDSP dealer, you will receive a coupon together with the product. Redeem this coupon at the link below:

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

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



Navigate to the C-DSP plugins section and download the zip file under the heading Harmony 8x12 DSP Plugin.

After downloading, unzip the file (right-click and select "Extract All..."). The unzipped download has a name like **Harmony\_DSP\_8x12\_v1\_13\_FW\_v1\_5**. (The version numbers embedded in the folder name may be different.)



#### 2.1.2.2 Possible Windows installation issues

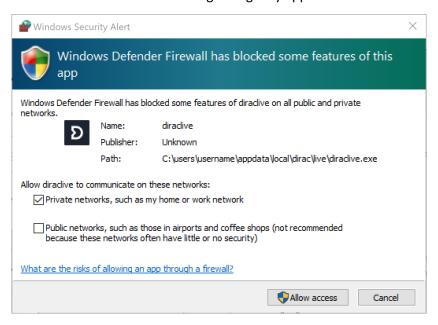
The miniDSP software requires that a number of other frameworks be installed for it to work. These packages should be installed automatically, but you can manually install them if you receive an error message that required software is missing.

- Microsoft .NET framework (version 3.5 or later)
- Microsoft Visual C++ 2010 Redistributable Package: for x86 (32-bit Windows) or x64 (64-bit Windows).

## 2.1.2.3 Install the plugin

- 1. Navigate to the **Plugins** folder of the software download and then to the **Windows** folder.
- Double-click on the plugin installer to run it. It will be named C-DSP-8x12-DL\_v1\_13.exe. (The version number embedded in the filename may change.) We recommend that you accept the default installation settings.

**Note**: the first time you run the C-DSP 8x12 DL plugin and the DiracLive application, you may see a Windows Firewall warning such as the one below. Ensure that "Private networks..." is checked and "Public networks..." is not checked. Then click on "Allow access." This warning dialog may appear more than once.





### 2.2 MACOS

This section describes software download and installation for macOS 10.14 Mojave and later.



This software described in this section runs on macOS 10.14 Mojave or later **only**. Other versions of OS X / macOS are not supported by the current version of Dirac Live.

## 2.2.1 Download and install the DiracLive application

Download the **DiracLive** application for macOS from <a href="https://www.dirac.com/live/downloads/">https://www.dirac.com/live/downloads/</a>.

Double-click on the downloaded file to unzip it. Then double-click on the unzipped installer file to run it. *Be sure to accept the default installation settings*. Do not run the application yet.

#### 2.2.2 Download and install the miniDSP software

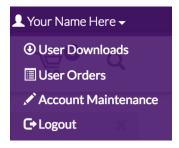
### 2.2.2.1 Download from the miniDSP website

If you purchased your processor directly from miniDSP, your software will be available from the <u>User Downloads</u> section of the miniDSP website when your order ships. To access the download, you will need to be logged into the website with the account you created when purchasing.

If you purchased your processor from a miniDSP dealer, you will receive a coupon together with the product. Redeem this coupon at the link below:

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

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



Navigate to the C-DSP plugins section and download the zip file under the heading Harmony 8x12 DSP Plugin.

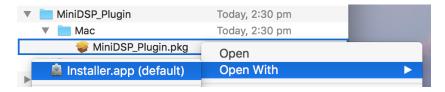
After downloading, unzip the file (double- click on it). The unzipped download has a name like Harmony\_DSP\_8x12\_v1\_13\_FW\_v1\_5. (The version numbers embedded in the folder name may be different.)

## 2.2.2.2 Possible Mac installation issues

If double-clicking on the installer brings up a message that the installer cannot run, use this alternate method:



- 1. Right-click on the installer (or click while holding the Control key).
- 2. Move the mouse over the "Open With" item and then click on "Installer (default)." (The name "MiniDSP\_Plugin" will be replaced with the actual plugin name.)



3. The following window will appear. Click on "Open."



#### 2.2.2.3 Install the plugin

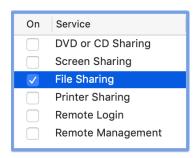
- 3. Navigate to the **Plugins** folder of the software download and then to the **Mac** folder.
- 4. The installer is named **C-DSP-8x12-DL\_v1\_13.pkg**. (The version number embedded in the filename may change.) To run it, double-click on it, or right-click and open as described above. We recommend that you accept the default installation settings.
- 5. To run the plugin, locate it in the Applications -> miniDSP folder and double-click on it. To make it easier to run in future, right-click on its dock icon and select Options -> Keep in Dock.

#### 2.2.2.4 Enable file sharing for device discovery

To enable device discovery, open System Preferences, go to Sharing, then enable File Sharing as shown at right.

#### Notes:

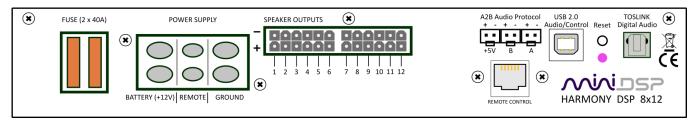
- a) This step is not always necessary and may depend on your Mac's configuration or your home network setup.
- b) If you wish, you can turn File Sharing off again after completing your Dirac Live calibration.





## 3 HARDWARE CONNECTIVITY

Connections to the Harmony DSP 8x12 are made on the rear panel.



## 3.1 Mounting

Mount the amplifier inside the vehicle, typically in the boot, trunk or cargo area. Do not place it in the engine bay, near hot components (exhaust) or where it may be exposed to weather, dust, fluids or corrosive substances.

The amplifier needs to have adequate space around it so air can flow around it for cooling. Don't cover it with seating or carpet, or wedge it between anything, for example. Don't place anything on top of it or place it in a closed console with no ventilation.

## 3.2 USB

The USB 2.0 (Type B) port is used for:

- Configuration and calibration with the miniDSP plugin and with DiracLive. Connect to an available USB port on your computer.
- Audio playback of up to 6 channels at 44.1 or 48 kHz. Connect to an available USB port on a computer, tablet, phone, streamer or Android-based head unit. In some cases you will need an appropriate adapter.

## 3.3 TOSLINK DIGITAL AUDIO

The TOSLINK input accepts stereo PCM audio at sample rates from 44.1 to 192 kHz.

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

## 3.4 A2B

The A2B ports are for an optional future enhancement (via firmware upgrade).

## 3.5 REMOTE CONTROL

Connect the OLED wired remote with the supplied RJ-11 phone cable.



### 3.6 POWER SUPPLY AND REMOTE TRIGGER

The power supply and trigger connections accept bare wire connections.

- 1. Connect a nominal 12V DC supply (10 16 VDC range is acceptable) to the terminals marked "Battery (+12V)" and "Ground". Use the supplied hex key to tighten the connecting wire securely.
- 2. Connect a remote trigger source to the Terminal labeled "Remote". Use the smaller supplied hex key to make this connection. The Harmony will power on when the voltage on this connection is greater than 4 VDC.

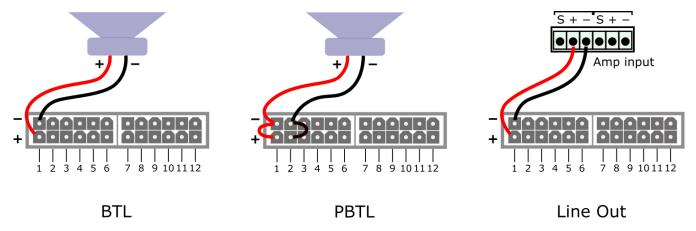
If no remote trigger source is available, a mechanical switch can be used to connect Remote to +12V DC.

## 3.7 ANALOG OUTPUTS

There are twelve analog output channels provided on two connector blocks. To simplify connection, mating connectors with labelled pigtails are provided with the Harmony unit.<sup>1</sup>

There are three modes for the outputs. This is set for each output channel using the plugin.

- 1. BTL (bridge-tied load) mode. Connect the + and terminals to a single speaker or speaker driver. Note: be sure not to connect the negative output to ground, as both positive and negative outputs are "hot."
- 2. PBTL (parallel bridge-tied load) mode. Connect the + and terminals of the odd-numbered channel to the positive speaker terminal. Connect the + and terminals of the corresponding even-numbered channel to the negative speaker terminal. Note: be sure not to connect any output to ground.
- 3. Line Out mode. In this mode the output is a **balanced** line-level output with full-scale output of 5 VRMS, for driving an external amplifier or subwoofer that has balanced inputs. **Caution**: See *Notes on Line Out mode* on the next page.



To set the output mode, see page 27.

<sup>&</sup>lt;sup>1</sup> For additional or custom cable looms, use Molex 5557 series or equivalent. Housing PN: 39012120, Crimp terminal PN: 457503111 <a href="https://www.molex.com/molex/products/part-detail/crimp">https://www.molex.com/molex/products/part-detail/crimp</a> terminals/0457503111



### 3.7.1 Notes on Line Out mode

In Line Out mode, the output signal is a balanced signal suitable for driving a balanced input on an amplifier or subwoofer. The maximum level is 5 Vrms.

Note the following important points if Line Out mode is used:

- 1. The Line Out signal is not pop-free on every power up. A connected amplifier or subwoofer must therefore be muted whenever the Harmony DSP 8x12 is powered on.
- 2. Do not connect an output channel in Line Out mode directly to an unbalanced external amplifier. This will cause a short circuit from the negative output of the Harmony to ground. The Harmony will shut down to protect itself if this occurs.
- 3. If it is necessary to connect the Harmony DSP 8x12 to an amplifier's unbalanced input, use an audio isolation transformer between them.
- 4. If noise is picked up on a channel in Line Out mode:
  - 1. Connect using a shielded twisted-pair cable.
  - 2. Ensure that the cable does not run close to the 12V supply cable or to any other power wiring in the vehicle.

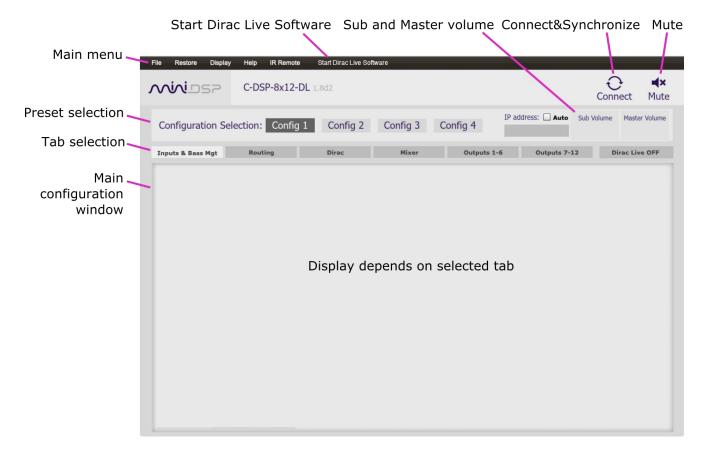


## 4 Plugin Overview

The overall power of the Harmony DSP 8x12 comes not only from its comprehensive I/O but its merge of Dirac Live processing with miniDSP's powerful audio processing.

## 4.1 USER INTERFACE

This screenshot shows the **C-DSP 8x12 DL** plugin with the key areas highlighted:





We recommend that you do a "straight-through" Dirac Live calibration before attempting more advanced configuration with the plugin. This way you can become familiar with Dirac Live calibration and learn how the calibration algorithms work in your vehicle.



The **C-DSP 8x12 DL** plugin works with both the Harmony DSP 8x12 and the C-DSP 8x12 DL processors. Some features of the plugin will be enabled or disabled according to the specific processor that is connected.



#### 4.2 SYNCHRONIZING WITH THE PROCESSOR

Communication with the Harmony DSP 8x12 for configuration and control takes place over USB. Ensure that the computer is connected to the processor via a USB 2.0 port. Then click on the **Connect** button:







The first time you connect, or if you have made any changes to any data in the plugin 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 output equipment should be muted or powered off until you have set the configuration to a working state. *Note that the configuration data currently stored in the processor will be deleted.* 

#### Cancel

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



## 4.3 GLOBAL CONTROLS

#### 4.3.1 Start Dirac Live Software

The **Start Dirac Live Software** button (in the menu bar at the top of the window) starts the separate **DiracLive** application. When it is pressed, the C-DSP 8x12 DL plugin will disconnect from the Harmony DSP 8x12 processor and then start up **DiracLive**.

Start Dirac Live Software

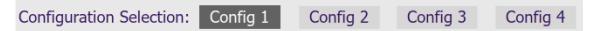


You **must** start the DiracLive application from within the C-DSP 8x12 DL plugin using the **Start Dirac Live Software** button. If you open the DiracLive application by itself, it will not be able to detect the Harmony DSP 8x12 processor.

## 4.3.2 Configuration/preset selection

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

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



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



#### 4.3.3 Tab selection

For the selected configuration, this row of tabs selects the parameters to be display in the main display area.



#### 4.3.4 Master mute

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



#### 4.3.5 IP Address and Auto

Reserved for future use.



#### 4.3.6 Subwoofer and Master volume

The Harmony DSP 8x12 has two modes of volume control operation:

#### Master volume mode

In this mode, the wired remote and infrared remote control the master volume i.e. all output channels. Master volume can be varied from -72 to 0 dB.

#### Subwoofer volume mode

In this mode, the wired remote and infrared remote control the volume of just a selected set of channels. This is typically used to control subwoofer volume but can also be used in other ways – for example, to control rear channel speaker level. Subwoofer volume can be varied from -72 to +12 dB.

To set which channels are controlled in subwoofer volume mode, see page 38.

The current values for master and subwoofer volume are displayed in the plugin while connected:



These can be edited directly by typing the desired value and pressing the Tab or Enter key. Note that Subwoofer volume is *relative* to master volume – that is, when master volume is changed, the actual signal level from the subwoofer changes by the same amount as the speakers.



When performing measurements or calibration, check first that Subwoofer volume is set to your "normal" level. Typically, setting Subwoofer volume to 0 dB is a good choice. Then you can increase it by up to 12 dB.

#### 4.3.7 Dirac Live on/off

The **Dirac Live** button turns Dirac Live processing on and off. This can also be done with a remote control or from within the DiracLive application:





#### 4.4 SIGNAL FLOW

To understand how the Harmony DSP 8x12 combines Dirac Live with miniDSP's audio processing, refer to Figure 3. Each of the blue labels at the top of the diagram corresponds to a tab accessible in the plugin user interface.



#### From left to right:

- Eight digital input channels are passed through bass management, to extract a subwoofer channel (optional)
- A matrix mixer routes nine channels through to eight channels of Dirac Live processing.
- The signals from Dirac Live are routed through an 8x12 matrix mixer to the 12 output channels.
- Each output channel is processed with crossovers, parametric EQ, and gain and delay.

The signal processing is configured by two different programs – the **DiracLive** application for Dirac Live and the **C-DSP 8x12 DL plugin** for the processing on the input and output side of Dirac Live. Each program loads its configuration into one of four presets or "slots". When a preset is selected, the configurations from both programs are loaded into the working DSP memory.

Because of the way these two "halves" of the Harmony DSP 8x12 work together, it's usual to configure the system from the outputs back to the inputs.

For the sections configured by the C-DSP 8x12 DL plugin, acoustic measurement with a separate program like Room EQ Wizard (REW) will give the best results. (See the <u>application notes</u> on our website for guidance.)

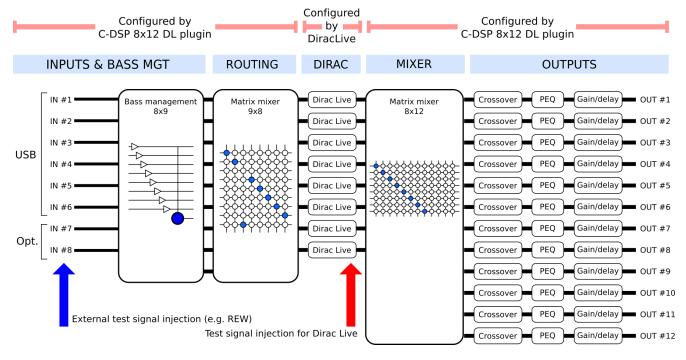
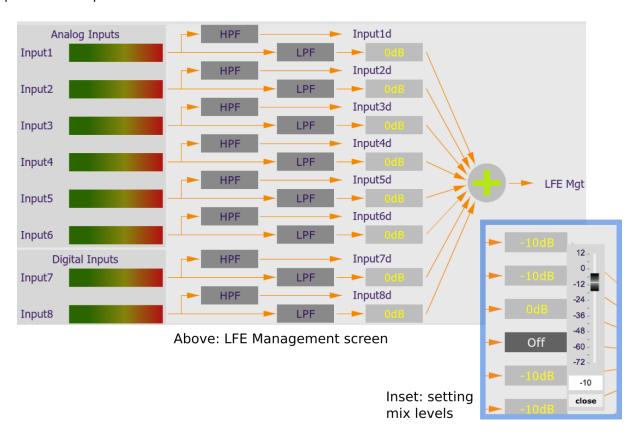


Figure 3. Signal flow through the Harmony DSP 8x12

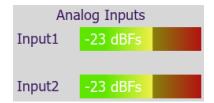


## 4.5 INPUTS & BASS MGT TAB

This tab is used to generate a subwoofer feed from the speaker channels. The Harmony DSP 8x12 provides superb control over subwoofer integration, including fine-grain frequency settings, accurate time delay, and steep crossover slopes.



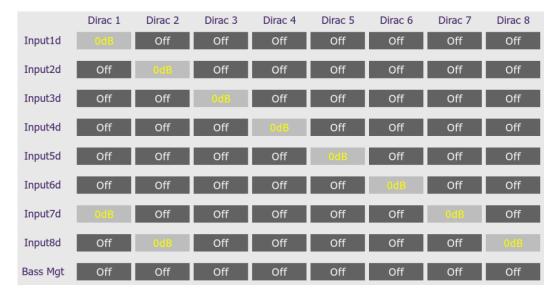
The level meters at the left display the current input signal level. This can be very useful for debugging and troubleshooting. The plugin must be in online mode to display the signal level.





## 4.6 ROUTING TAB

This tab mixes or routes the input channels and the Bass Mgt signal and sends them to the Dirac Live processing algorithm. The input channels are labeled along the left, and the output channels are labeled along the top. There are 9 input channels and 8 output channels. Here is the default setting:

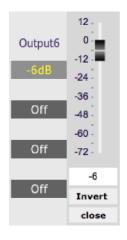


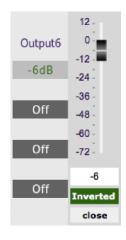
At each cross-point of the matrix, turn signal routing on and off by clicking. When routing is on for that cross-point, the level at which the input channel (labels along the left) is mixed into the corresponding output channel (labels along the top) is displayed in yellow. Any number of input channels can be mixed to each output channel.

Note: To rename an input channel, click on its label and type in a new name (maximum of eight characters).

To change the gain of the mixed signal at a cross-point, right-click on the cross-point to pop up the gain control. Move the slider – or alternatively, type the value directly in to the entry box below it – *press the Enter key*, then click the **close** button. The gain can be set in 1 dB steps between -72 and +12 dB.

Each cross-point can also have the signal inverted by clicking on the **Invert** button. When the mixed signal is inverted, the button will be displayed in green and its label will change to "Inverted". The lettering of the cross-point will be highlighted in green.







### 4.7 DIRAC LIVE TAB

This tab displays the gains and delays of the Dirac Live filters loaded into the Harmony DSP 8x12. (The plugin must be online to display them.) These gains and delays are calculated by the DiracLive application during its Optimize phase and cannot be changed by the user – they are "read only." Here is an example:



The **Input** and **Output** columns display the current signal level at the inputs and outputs of the Dirac Live processing block, while Gain and Delay display the gains and delays calculated during Dirac Live calibration. Note that the displayed gains and delays are applied even when Dirac Live filtering is turned off.



In order to pass audio through without gain and levels adjustment, such as when measuring with Room EQ Wizard for an active crossover, you will need to leave an empty slot on the **Filter Export** tab of the DiracLive application.

The button is used to turn Dirac Live processing on and off.

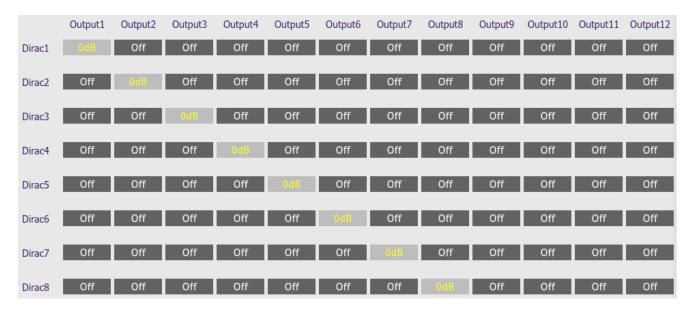




## 4.8 MIXER TAB

The **Mixer** tab displays the matrix mixer, which sets up routing and mixing from Dirac Live channels to output channels. The Dirac Live channels are labeled along the left and the output channels along the top.

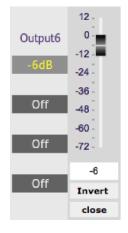
This is the default Mixer tab:

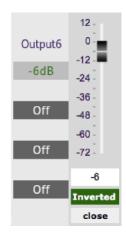


When a cross-point is turned on, the Dirac Live signal (at the left) is routed to the output channel (at the top). In the default routing, Dirac1 is routed to Output1, Dirac2 to Output2, and so on.

This matrix mixer is typically used to route audio to multiple output channels for the purpose of implementing active speakers. It is also used when directing the subwoofer signal to multiple subwoofer output channels.

As with the Routing tab, each cross-point has adjustable level and an Invert option.

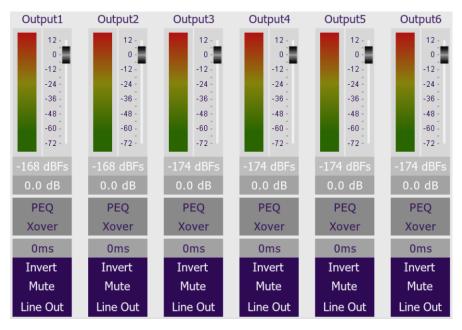




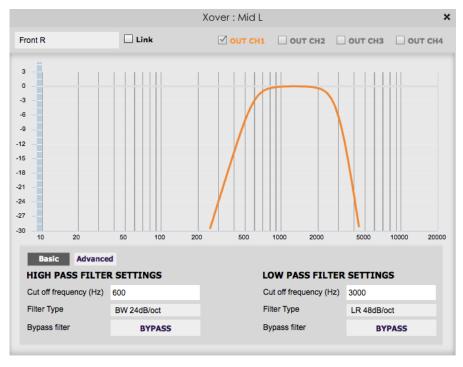


#### 4.9 OUTPUT TABS

There are two output tabs: one for channels 1 to 6 and one for channels 7 to 12. Each displays a row of six output channel control strips. All output channels are identical. This screenshot shows channels 1 to 6:



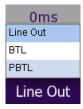
Each output channel has a complete "strip" of controls with customizable label, level meter, gain adjustment, signal processing (see the Plugin Reference starting on page 41 for full details), and output mode (Line, BTL, PBTL) selection. Signal processing functions are accessed by clicking on the relevant button – for example, the crossover settings screen is opened by clicking on the **Xover** button:





#### 4.10 OUTPUT MODE SELECTION

The output mode is shown at the bottom of each channel when the plugin is in online mode. The screenshot on the previous page shows the factory default setting, with all channels in Line Out mode. To change the mode, click on the button. Odd-numbered channels can be selected between Line Out, BTL or PBTL:

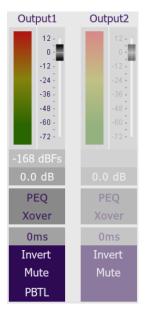


Even-numbered channels can be selected between Line Out or BTL:



Note that output mode is independent of the selected preset configuration.

Because PBTL mode applies to a pair of channels, the corresponding even-numbered channel is greyed out on the Output and Mixer tabs when PBTL mode is selected:







Turn off the power to the Harmony DSP 8x12 after changing output mode, and then re-apply power. You can do this by taking the Remote trigger input low and then high again, or by removing power entirely and then re-applying it. Be sure to follow the instructions provided by the plugin when changing modes. Be sure to read *Notes on Line Out mode* on page 16.



#### 4.1 APPLICATION GUIDE

This section describes some examples of audio processing configurations that can be set up using the plugin. They start from simple (no configuration required) through to more advanced. Each example is not necessarily a complete configuration but describes part of the signal processing flow. In many systems, you will combine various of these examples for your needs – for example, bass management combined with active crossover.

## 4.1.1 Straight through

In this example, no configuration in the plugin is required, as the flow of processing is directly through from inputs to outputs.

Figure 4 illustrates the signal flow and processing in dark blue, overlaid on the signal flow diagram from Figure 3. In this case, the signals flow directly from each input to the corresponding output, with processing only being done in the Dirac Live signal processing blocks in the middle. You can simply leave the plugin in its default configuration and proceed directly to Dirac Live calibration.



Figure 4. Straight-through processing in the Harmony DSP



If you have not used Dirac Live before, we suggest that you perform a Dirac Live calibration without configuring any additional processing with the plugin. Once you have learned how Dirac Live calibration works, then attempt a more advanced configuration using the plugin.



## 4.1.2 Routing the stereo digital input

Typically, the stereo digital input will play through the front left and right speakers. You can combine this with any of the other types of routing explained in this section – for example, subwoofer integration, active speakers, and so on.

Figure 5 illustrates the routing of the stereo digital input. In the Routing tab, input channels 7 and 8 are routed to Dirac channels 1 and 2.

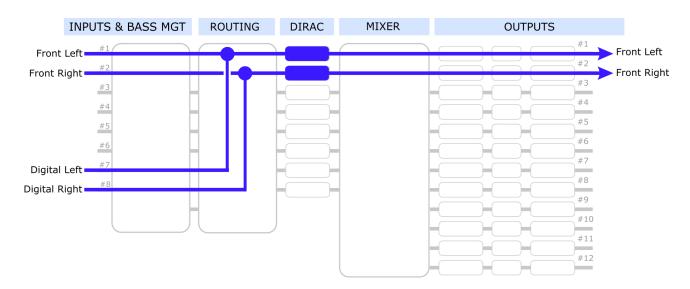


Figure 5. Routing the stereo digital input

Use the Routing tab to implement this routing:





## 4.1.3 Bass management

Figure 6 illustrates the signal flow and processing for bass management in the Harmony DSP 8x12. For simplicity, the diagram shows only the front left and front right input channels, but usually you would do the same for the rest of the input channels as well.

The Bass Management processing:

- Applies high pass filters ("HPF") to the signal sent to Dirac Live and then to the speakers. This takes the low-frequency load away from the speakers.
- Applies low pass filters ("LPF") to extract low frequencies and sums these signals to generate the Bass Mgt signal. In the Routing tab, this is sent to the subwoofer channel for Dirac Live correction.

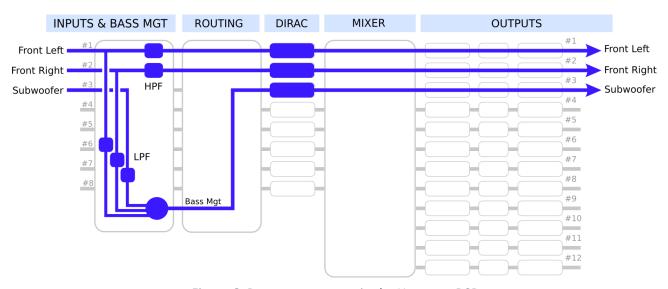


Figure 6. Bass management in the Harmony DSP

The important thing to note here is that the Subwoofer input does not feed directly through to Dirac Live – instead, it is low pass filtered and summed with the low frequencies from the speaker channels (Front Left and Front Right in the example), then routed via the Bass Mgt signal.

Use the Routing tab to route the Bass Mgt signal to the Dirac channel that will be used for the subwoofer. It will look something like this:





To use this configuration, do a Dirac Live calibration first on the speaker and subwoofer channels. Then set up the Inputs & Bass Mgt tab with the high pass and low pass filters for the subwoofer crossover. For example, in the HPF block:

HIGH PASS FILTER SETTINGS		
Cut off frequency (Hz) 80		
Filter Type	BW 24dB/oct	
Bypass filter	BYPASS	

And in the LPF block:

LOW PASS FILTER SETTINGS			
Cut off frequency (Hz)	80		
Filter Type	BW 24dB/oct		
Bypass filter	BYPASS		

You should perform acoustic measurements to check and fine-tune your subwoofer crossover settings, to ensure a smooth response through the crossover. You may need to adjust the filter type, frequency or slope to get the best integration. After you have set up the plugin, save your configuration to a file.



### 4.1.4 Dual subwoofers

Multiple subs can be connected to separate outputs of the Harmony DSP. This enables you to use the output channel processing to fine-tune the combined response of all subs. Use the Mixer tab to route the processed subwoofer signal to two or more output channels as shown in Figure 7.

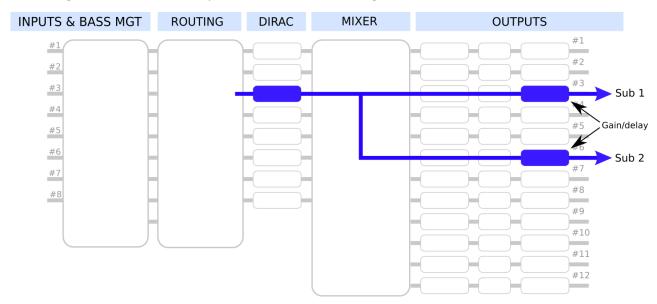


Figure 7. Processing flow example for multiple subwoofers

It is important to note that there is only one Dirac Live channel for all the subs. This way, Dirac Live can correct for the combined response of **all** subwoofers playing at the same time. Set up the Mixer tab as shown in this example:



You can adjust individual subwoofers on the output channels *before* running Dirac Live calibration. For example, changing the delay between the subs can improve the combined response, either by removing a null or by reducing the variation between seats. This application note on our website will give you some ideas on how to tune with multiple subs (it was written for a room, but Methods B and C will apply in a vehicle):

• Tuning multiple subwoofers with miniDSP



## 4.1.5 Rear channel synthesis

The **Invert** button in the matrix mixers is useful in applications such as generating a difference signal for rear fill. This must be done in the Routing tab, not the Mixer tab. This allows Dirac Live to correct the response of each individual speaker (correcting the response of two speakers at the same time does not work well). The signal flow is illustrated in Figure 8.

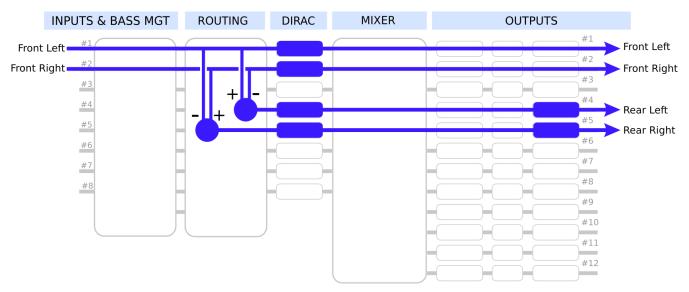


Figure 8. Processing flow for rear channel synthesis

The basic strategy is as follows:

1. Create L–R and R–L signals as shown in the routing matrix below.



- 1. Add a delay of up to 20 ms to the rear channel outputs (experiment to see what sounds best to you).
- 2. Add high pass and low pass filters to the rear channels to limit bandwidth (experiment with this).

However, there is some interaction with Dirac Live. When performing a Dirac Live calibration, Dirac Live will attempt to compensate for the delays on the rear output channels. The high pass and low pass filters will also make calibration more difficult. Therefore, we suggest you follow this procedure each time you run a calibration:

- 1. Turn off high pass and low pass filters on the rear output channels.
- 2. Zero the delays on the rear output channels.
- 3. Run the Dirac Live calibration.
- 4. Set the delay and filters on the rear output channels again. Experiment to find out what sounds best.



## 4.1.6 Active speaker crossover

The Harmony DSP can implement active speaker crossovers of any complexity up to the number of available output channels. An example processing flow is illustrated in Figure 9.

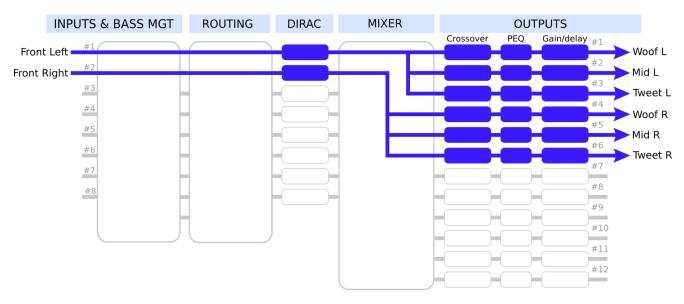
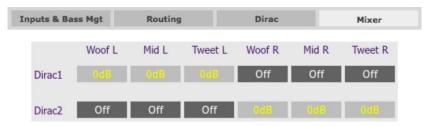


Figure 9. Processing flow for active speakers (three-way example)

Each speaker has a single Dirac Live channel. The Mixer tab routes each channel from Dirac Live to three output channels. It looks like this:



You will need to use REW to measure each driver, correct it, and implement crossover filters. The procedure is similar to that described in our app notes for a two-way speaker <u>at this link</u>. For example, the midrange crossover might look like this:



Your system may be more complex than this. For example, you may have active rear channels as well, and use bass management to drive one or more subwoofers. Set up the crossovers for all channels first, then proceed to perform the Dirac Live calibration and fine-tune your target curve.



## A note on active crossovers and Dirac Live

The connection scheme shown in Figure 9 is our recommendation for implementing an active crossover and combining it with Dirac Live. Because of the number of Dirac Live channels available, you may be tempted to try an arrangement where each individual driver is corrected by a single Dirac Live channel. The configuration would look something like this:

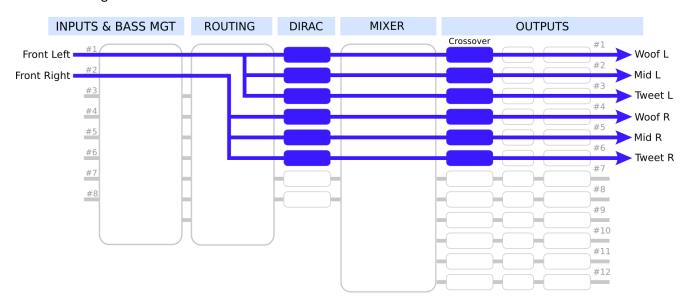


Figure 10. Alternative processing flow for active speakers (not recommended)

This arrangement is tempting as it seems that Dirac Live will adjust for the delays between the drivers and flatten the response of each driver individually. However, we don't recommend this configuration. Apart from being more difficult to set up:

- Dirac Live will not be able to correct for the phase shift through the crossover(s). In other words, impulse response correction will not be working as expected.
- Dirac Live will not be able to correct the *combined* response of the drivers in the crossover region.

We therefore recommend that you use the output channel processing to implement the active crossover (including basic driver correction and delays between drivers) and allow Dirac Live to work on the combined response of two or three active-crossed drivers.



## **DIRAC LIVE CALIBRATION**

Dirac Live calibration with the C-DSP 8x12 DL is described in the separate miniDSP Dirac Live User Manual.



Be sure to start the DiracLive application from within the C-DSP 8x12 DL plugin using the Start Dirac Live Software button. If you open the DiracLive application by itself, it will not be able to detect the Harmony DSP 8x12 processor.

## MINIDSP

## **DIRAC LIVE**

**USER MANUAL** 

FOR DIRAC LIVE-ENABLED MINIDSP PROCESSORS

#### STEREO PLATFORMS

**SHD SERIES** 

DDRC-24

DDRC-22D

#### **MULTICHANNEL PLATFORMS**

DDRC-88A, DDRC-88D

C-DSP 8x12 DL, HARMONY DSP 8x12

#### SUPPORTED OS



Windows 10



macOS Mojave



macOS Catalina



# **6** REMOTE CONTROL

The OLED wired remote can be used to control:

- Master volume
- Subwoofer volume
- Volume mode selection
- Master mute
- Preset selection
- Dirac Live on/off



## 6.1 STATUS INDICATORS

The OLED display shows the currently selected preset, master volume, and the status of mute and Dirac Live.

### 6.2 OPERATION OF THE OLED WIRED REMOTE

#### To change the volume

Rotate the control knob clockwise to increase the volume, and counterclockwise to decrease it. Either the master or subwoofer channel volume will change, depending on the current volume control mode.

#### To mute and unmute

Press and hold the control knob.

#### To change the volume control mode

Quickly press the control knob, then rotate until either "Master" or "Sub" is displayed. Either press the control knob again or wait for a second. The displayed volume will either be the master volume, or the additional gain/attenuation applied to the designated subwoofer channels.

### To change the preset

Quickly press the control knob twice, then rotate until the desired preset number is displayed. Either press the control knob again or wait for a second. The display will show "Pls wait..." while the new preset is being loaded.

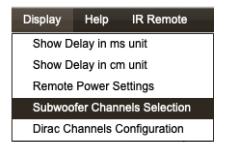
### To enable or disable Dirac Live processing

Quickly press the control knob three times. The display will change to show either "Dirac On" or "Dirac Off" in large letters. Rotate the knob until the desired selection (on or off) is displayed, then press the control knob or wait for a second.

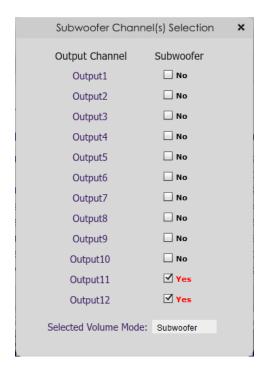


### 6.3 CHANNEL SELECTION FOR SUBWOOFER VOLUME CONTROL

Any number of channels can be selected for subwoofer volume control mode. From within the C-DSP 8x12 DL plugin, drop down the **Display** menu and select **Subwoofer Channels Selection**.



The selection dialog appears:



- 1. Check the channels to be controlled in subwoofer volume control mode.
- 2. If desired, use the dropdown labeled **Selected Volume Mode** to select the desired mode.



### 6.4 Using the miniDSP remote

The miniDSP remote control (v2) controls the following runtime functions. The infrared receiver for the remote is located in the OLED wired remote.



#### Standby

Has no effect with the Harmony DSP 8x12.



#### Mute

Mute and unmute audio output.



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

#### Media control



Has no effect with the Harmony DSP 8x12.



#### **Dirac Live**

Enable or disable Dirac Live filtering. Dirac Live filtering will be effective only on presets for which Dirac Live filters have been loaded.



### **Source selection**

Has no effect with the Harmony DSP 8x12.

(1)

### Preset (1 through 4)

Switch to the selected preset. It takes a few seconds for the preset selection to complete, while the processor loads the new filters from its flash memory into the DSP.





## 6.5 LEARNING THIRD-PARTY REMOTE CODES

The Harmony DSP 8x12 can "learn" the control codes of a third-party remote if it supports one of the following remote control codes:

- Apple
- NEC
- Sony
- Philips RC6

To initiate the learning process, start the C-DSP 8x12 DL plugin and click on the Connect button. Once connected, 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. Repeat for all commands:





To "unlearn" a command, press the **Learn** button and wait for the plugin to time out. Note that you cannot "learn" the miniDSP remote.

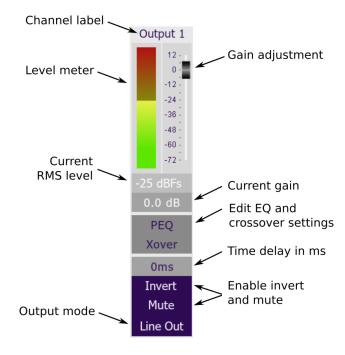


# 7 Plugin Reference

## 7.1 OUTPUT CHANNEL PROCESSING

#### 7.1.1 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.



### 7.1.2 Level meter and gain control

#### Level meter, current RMS level

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

#### **Gain adjustment**

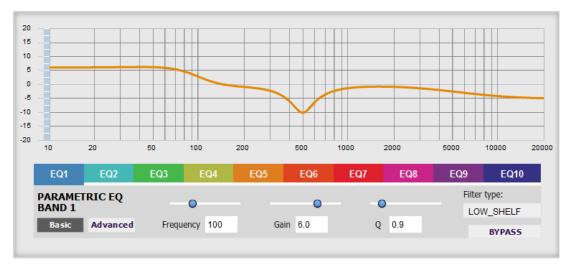
The gain of each channel can be adjusted by moving the Gain Adjustment slider, or by typing the desired gain into the Current Gain text box. The maximum gain setting is 12 dB, and the minimum gain setting is –72 dB. (0 dB, the default, is unity gain or no change in level.)

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.



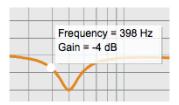
### 7.1.3 Parametric EQ (PEQ)

Parametric equalization (PEQ) is a flexible type of equalization filter. It can be used to correct for errors in driver response when implementing an active speaker system. Click on the PEQ button to open the parametric equalizer settings window:



There are 10 parametric EQ filters on each input and output channel. The window displays a frequency response graph showing the combined response of all enabled filters on that channel. For example, the screenshot above shows a response curve created with a low-shelf boost filter at 100 Hz, a dip at 500 Hz, and a high-shelf cut filter at 5000 Hz.

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



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





#### **EQ** band selection

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

#### Basic/Advanced

By default, each filter is in basic mode and shows the controls described below. Advanced mode enables custom biquad programming for almost infinite flexibility in filter implementation. This is described in Custom biquad programming on page 47.

#### Filter type

Selects the type of filter:

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

**LOW\_SHELF** Reduce or increase part of the frequency spectrum *below* a given frequency.

**HIGH\_SHELF** Reduce or increase part of the frequency spectrum *above* a given frequency.

**ALL\_PASS** Create a phase shift across the frequency band. This can be useful for correcting phase

issues and for simulating analog crossovers.

#### **Frequency**

For the PEAK filter type, this is the center frequency of the peak or dip. For the HIGH\_SHELF and LOW\_SHELF filter types, this is the frequency at which the gain is half of the set value. For the ALL\_PASS filter type, this is the center frequency of the phase shift.

#### Gain

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

#### Q

Q controls the "sharpness" of the filter. For the PEAK filter type, lower Q gives a broader peak or dip, while higher Q gives a narrower peak or dip. For the HIGH\_SHELF and LOW\_SHELF filter types, Q controls how quickly the filter transitions from no gain to maximum gain. For the ALL\_PASS filter type, higher Q gives a steeper phase transition.

#### **Bypass**

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





### 7.1.4 Crossover (Xover)

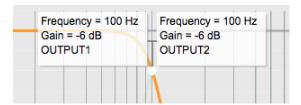
Each output channel has independent high pass and low pass crossover filters. Click on the **Xover** button to open the crossover settings window:



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.

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 47.

#### **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, 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**

The **Bypass** button enables or disables a crossover filter. The filter is enabled if the button says "BYPASS" and disabled if the button says "BYPASSED" (see screenshot below).



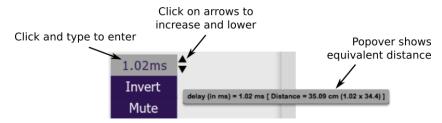
Each channel can be linked to one other channel. When a channel is linked to another, the crossover settings of that channel are mirrored to the other. Typically, the corresponding drivers on the left and right channels are linked: left and right tweeter, left and right woofer, and so on. To link a channel,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





### 7.1.5 Time delay

A delay of up to 20 ms can be applied to each output channel. To set the delay, click in the delay entry box for a channel. The delay value can be entered numerically by typing in the entry box. The up and down arrows can be used to change the delay in small increments.

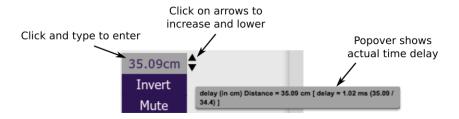


The time delay corresponds to a distance. This distance is shown in cm when the cursor is hovered over the time delay box. The maximum time delay of 20 ms corresponds to a distance of approximately 6.9 meters (about 22.5 feet).

The delay can also be displayed and entered directly in terms of its equivalent distance in cm. To do so, drop down the **Display** menu and select **Show Delay in cm unit**.

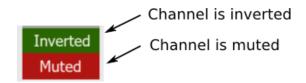


In this case, hovering the cursor over the entry box will display the actual time delay:



#### 7.1.6 Invert and mute

Each output channel can be inverted in polarity, and individually muted. When either of these options is selected, the display changes color and the label of the button reflects the current state.





### 7.2 CUSTOM BIQUAD PROGRAMMING

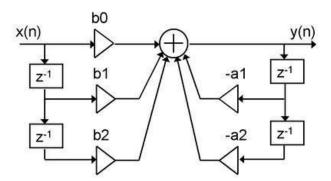
Custom biquad programming is available in the **PEQ** and **Xover** (crossover) blocks. Its purpose is to allow you to directly provide the low-level parameters aka *biquad coefficients* that control the digital filters of the processor, thus providing an almost infinite degree of flexibility.

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

### 7.2.1 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, b0, b1, and b2. These numbers are called the *coefficients*.

### 7.2.2 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 PEQ 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 PEQ block can be set at once by importing a coefficient file. (Click on the **IMPORT** button.) 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 for the **C-DSP 8x12 DL** 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,
biquad3,
biquad4,
. . .
biquad10,
b0=1.0010192374642126,
b1=-1.9950555192569264,
b2=0.9940580112181501,
a1=1.995060938714333,
a2=-0.9950718292249559
```

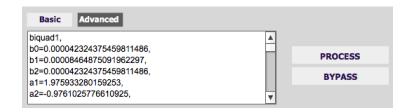
Note that the last line must not have a comma at the end. If the file has less than ten biquads, then only that number of biquads will be imported. For example, if importing a file with six biquads, the first six filters will be set, and the last four will not be changed. (Be careful: if the last line 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 **Xover** (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.





## 7.2.3 Biquad design software

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

### 7.2.3.1 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

Be sure to set the sample rate to 48 kHz on any worksheet that you use.

### 7.2.3.2 Room EQ Wizard (REW)

Room EQ Wizard (REW) 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 <u>Auto EQ with Room EQ Wizard</u>. When calculating the EQ in REW, choose "C-DSP 8x12 DL" in the Equaliser section of the EQ window. (Be sure not to choose "C-DSP 8x12," as that has a different internal sample rate.)



### 7.3 Working with configurations

The data that controls the audio processing Harmony DSP 8x12 is called a *configuration*. Four configuration presets are stored in internal memory. Presets can be selected with the wired remote panel or with an infrared remote.

### 7.3.1 Online and offline mode

Initially, the plugin is in *offline* mode. When the **Connect** button is clicked, 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:





Online mode

Offline mode

## 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 on your computer only. The next time the plugin is synchronized to the processor, the parameters will be downloaded to the processor (provided the **Synchronize Config** button is selected).



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

### 7.3.2 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 time that configuration was synchronized to the processor, then a dialog will appear asking you if you want to synchronize the configuration.

#### The plugin is in offline mode

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



### 7.3.3 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 defaults.

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



Note: the Harmony DSP 8x12 does not have an SD card slot, so while the menu option "Save all configuration to SD card" can be used to save a file, it cannot be reloaded into the Harmony.

To load a configuration from a saved file, drop down the **File** menu and select **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.



The output mode (Line Out, BTL, PBTL) of each channel is not stored in the configuration file and does not change when you switch preset configuration or load a configuration from a file.



### 7.3.4 Restoring to defaults

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

#### **Factory Default**

Reset all four configuration presets to the factory default settings.

## **Current Configuration Only**

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

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

## 7.4 KEYBOARD SHORTCUTS

The C-DSP 8x12 DL 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.



## 8 Additional Information

### 8.1 Specifications

# 8.1.1 General specifications

**Digital Signal Processor** 400 MHz, 32-bit floating-point SHARC Digital Signal Processor (ADSP21489)

Internal sample rate: 48 kHz.

Filtering capabilities Dirac Live correction (8 channels), matrix mixer, miniDSP processing on all 12

output channels

**Computer connectivity** Driverless USB 2.0 control interface for Windows and Mac

**Digital audio inputs** USB Audio, up to 6 channels at 48 kHz

TOSLINK Audio, stereo PCM at 44.1 to 192 kHz

A2B® multichannel audio (future option via firmware upgrade)

**Digital audio output** A2B® multichannel for linking amplifiers (future option via firmware upgrade)

Analog output modes (selectable from plugin)

BTL (bridge-tied load) mode for nominal 4  $\Omega$  loads (absolute minimum 2  $\Omega$ ) PBTL (parallel bridge-tied load) mode nominal 2  $\Omega$  loads (absolute minimum 1  $\Omega$ ) Line Out mode for driving external amplification with balanced line-level signal

**Analog outputs** See page 54.

**Configuration presets** Four onboard presets for Dirac Live and miniDSP processing stored onboard

Wired external remote External remote for control of active preset, master volume, subwoofer volume

and master mute. OLED status display.

Infrared remote "Learning remote" (NEC, Philips, Sony) controls master volume, mute, preset

selection.

**Remote trigger IN** Trigger level: 4 V DC

Maximum allowed input voltage: 16 V DC

**Power supply** 10 to 16 V DC compliant for car audio environment

**Dimensions (H x W x D)** 321 x 150 x 45 mm (main unit)

102 x 23 x 41 (wired remote)

(See next page for analog output specifications.)



## 8.1.2 Analog output specifications, BTL and PBTL modes

**Maximum output level** 12 x 40 W RMS (4  $\Omega$  load, BTL mode)

 $6 \times 80 \text{ W RMS}$  (2  $\Omega$  load, PBTL mode)

**THD+N** < 0.03 % (f = 1 kHz, 30 mW to 40 W, 4  $\Omega$  load, BTL mode)

< 0.08 % (f = 1 kHz, 30 mW to 80 W, 2  $\Omega$  load, PBTL mode)

**SNR** > 107 dB (f = 1 kHz, A-weighted, 40 W into 4  $\Omega$ , BTL and PBTL modes)

Frequency response 20 Hz – 20 kHz ± 0.5 dB (BTL and PBTL modes)

**Crosstalk** < -80 dB (1 kHz, BTL and PBTL modes))

### 8.1.3 Analog output specifications, Line Out mode

Maximum output level 5 V RMS balanced

**THD+N** < 0.03 % (f = 1 kHz, digital input 0 dB)

**SNR** > 102 dB (f = 1 kHz, A-weighted)

Frequency response  $20 \text{ Hz} - 20 \text{ kHz} \pm 0.2 \text{ dB}$ 



### 8.2 MCU FIRMWARE UPGRADE

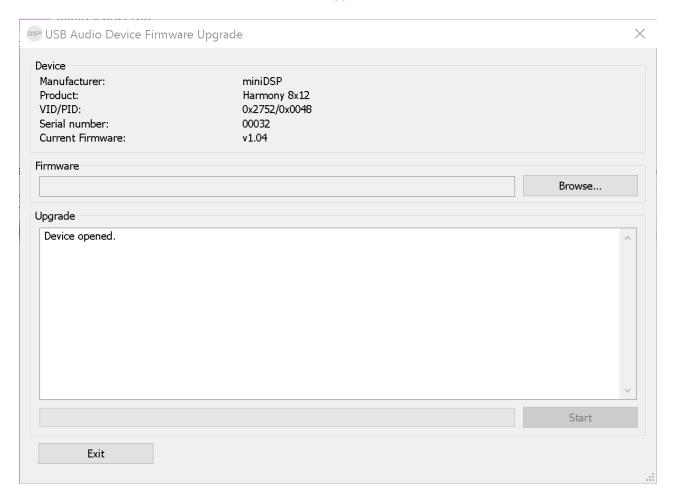
miniDSP may occasionally provide an upgrade to the processor's MCU firmware to enable new features. To upgrade the MCU firmware, first download the latest version of the C-DSP 8x12 DL software package from the **User Downloads** section of the miniDSP website, then extract it on your computer (on Windows, right-click and select "Extract All..."; on Mac, double-click).



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

#### 8.2.1 Windows

- 1. Connect the Harmony DSP 8x12 to your computer via USB (if not already connected) and power it on.
- 2. Navigate to the XMOS\_Firmware\Firmware\_Upgrade\_Tools\Windows\miniDSPUAC2Dfu folder of the software download.
- 3. Double-click on miniDSPUAC2Dfu.exe to run it. The application will start:

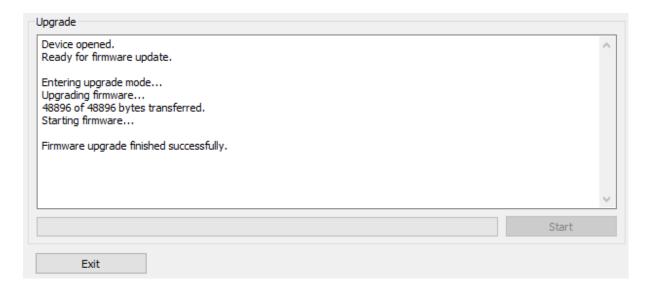




- 4. Click on the **Browse** button and select the firmware file from the **XMOS\_Firmware** folder of the software download. It will have a name like "Harmony\_8x12\_v1.4.bin." (The version number "v1.4" may change.)
- 5. Click on the Start button.
- 6. You will get a progress bar as the upgrade proceeds:



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

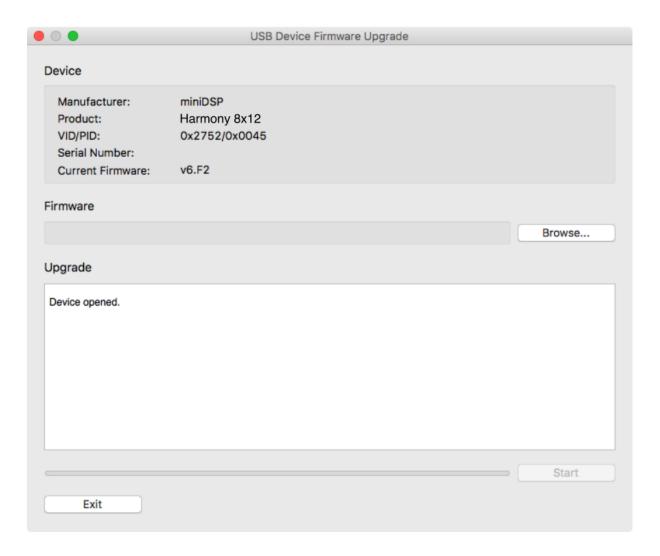


- 8. Click on Exit.
- 9. That's it! You're done. You can now use your processor with the new functionality.



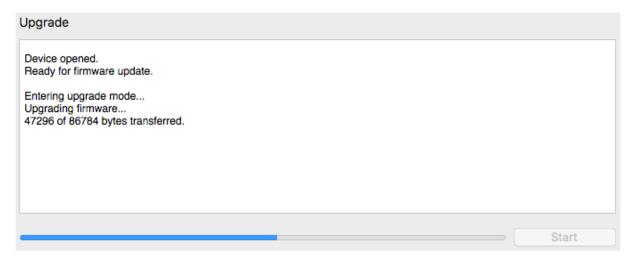
### 8.2.2 macOS

- 1. Connect the Harmony DSP 8x12 to your computer via USB (if not already connected) and power it on.
- 2. Navigate to the XMOS\_Firmware/Firmware\_Upgrade\_Tools/Mac folder of the software download.
- 3. Double-click on the .zip file to extract the firmware update software, then double-click on **miniDSP USB DFU.app** to run it. The app will start:

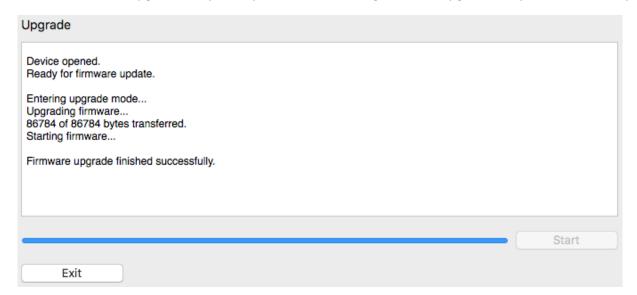




- 4. Click on the **Browse** button and select the firmware file from the **XMOS\_Firmware** folder of the software download. It will have a name like "Harmony\_8x12\_v1.4.bin." (The version number "v1.4" may change.)
- 5. Click on the **Start** button.
- 6. You will get a progress bar as the upgrade proceeds:



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



- 8. Click on Exit.
- 9. That's it! You're done. You can now use your processor with the new functionality.

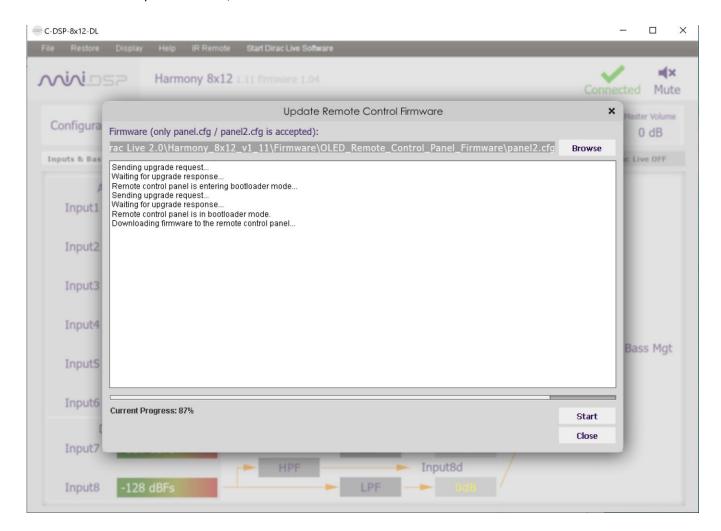


## 8.3 OLED REMOTE FIRMWARE UPGRADE

miniDSP may occasionally provide an upgrade to the firmware of the OLED wired remote to enable new features. To upgrade the MCU firmware, first download and install the latest version of the C-DSP 8x12 DL plugin.

Connect the Harmony DSP 8x12 to the computer with a USB cable, run the plugin and click the Connect button. Drop down the Restore menu and select "Update Remote Control Firmware." The upgrade window will open.

- Click on Browse and select the firmware file from the Firmware/
   OLED\_Remote\_Control\_Panel\_Firmware folder of the software download. It will have a name like "panel2.cfg."
- 2. Click on Start. The update will start.
- 3. When the update finishes, click Close.





# 8.4 TROUBLESHOOTING

The following table lists the most common causes of issues. If following this table does not provide a solution, see Obtaining support on the next page.

1	Cannot install software	a.	Confirm that you downloaded and installed the required frameworks first (see Software Installation).
2	C-DSP 8x12 DL plugin running in background but not showing	a.	The plugin may need a network connection the first time you run it. Close the plugin program, ensure that your computer has a network connection, and restart the plugin.
3	C-DSP 8x12 DL plugin cannot connect	a. b.	Check that the USB cable to the processor is firmly connected.  Reset the processor by power-cycling the unit.
4	No signal showing on input meters in C-DSP 8x12 DL plugin	a. b.	Check the cabling from your source.  Check that your source is playing audio and that it is not muted or have volume control turned down.  Check that the plugin is synchronized with the hardware unit.
5	Low audio on outputs	a. b. c. d. e. f.	Check the cabling from the processor to your amplifiers.  Check that your amplifiers are turned on and that any volume controls are turned up.  Check that the input and output meters are showing signal.  Check that master mute is not enabled.  Check the master volume level.  Check that your crossover frequencies are correct e.g. that you don't have high pass and low pass frequencies incorrectly set.  Check that the matrix mixer is sending the correct inputs to the correct outputs.
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.
7	Audio is coming through the wrong outputs	a. b.	Check the cabling from the processor to your amplifiers.  Check that you have correctly set up the matrix mixer to send the correct inputs to the correct outputs.
8	Cannot reload a configuration	a. b.	Confirm the file format of your file (.xml). Confirm the version of the file.



### 8.5 OBTAINING SUPPORT

- 1. Work through the Troubleshooting checklists starting on the previous page.
- 2. Check the forums on miniDSP.com to see if your issue has already been raised and a solution provided.
- 3. Contact miniDSP via the support portal at <a href="support.minidsp.com">support.minidsp.com</a> with:
  - a. The specific product you are having an issue with (in this case, Harmony DSP 8x12).
  - b. A clear explanation of the symptoms you are seeing.
  - c. A description of the troubleshooting steps performed and your results.