C-DSP 8x12 DL

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

User Manual

Revision history

Revision	Description	Date
2.0	Updated for Dirac Live 2/3, DiracLive documentation moved to separate miniDSP Dirac Live User Manual.	30 June 2020
2.1	Corrected support links	6 July 2020
2.2	Updated wired remote	1 August 2020
2.3	Correction for REW auto-EQ	2 December 2020
2.4	Correction to product overview	9 December 2020
2.5	Updates for OLED wired remote	16 December 2020
2.6	Adobe Air and Flash no longer needed	1 February 2021
2.7	Added digital input routing to Application Guide, Big Sur supported	5 August 2021

SUPPORTED OS

Windows 10

macOS Mojave

macOS Catalina

macOS Big Sur

If you are unable to run the above versions of the Windows or macOS operating system, you will need to use the original version of Dirac Live ("Dirac Live 1") with your C-DSP 8x12 DL processor. Please refer to the C-DSP 8x12 DL User Manual Legacy Support version.

Note: Use of Dirac Live with macOS requires some manual workarounds, which are documented in this manual.

CONTENTS

Important Information

Please read the following information before use. In case of any questions, please contact miniDSP via the support portal at support.minidsp.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

- **Microsoft• ® Windows® 10**, latest version with all updates
- Intel Pentium III or later, AMD Athlon XP or later
- 2 Gigabytes (GB) of RAM or higher
- Keyboard and mouse or compatible pointing device
- Two free USB 2.0 ports

Mac OS X

- **macOS 10.14 Mojave, macOS 10.15 Catalina**, or **macOS 11 Big Sur**, latest version with all updates
- Intel-based Mac with 1 GHz or higher processor clock speed
- 2 Gigabytes (GB) of RAM or higher
- Keyboard and mouse or compatible pointing device
- Two free USB 2.0 ports

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.

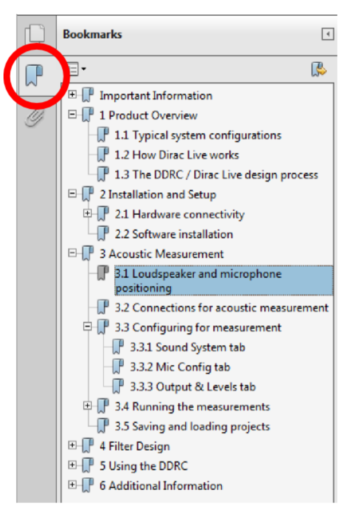
CE Mark Statement

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

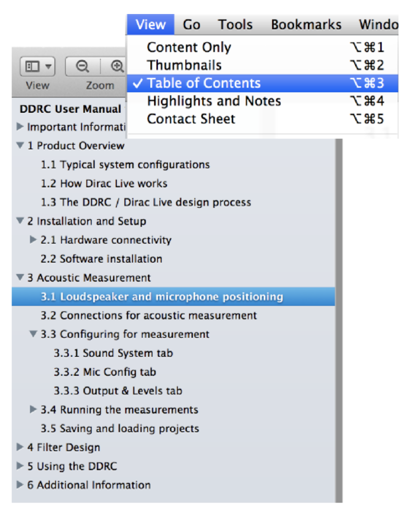
A note on this manual

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 C-DSP 8x12 DL advanced in-car audio processor. The C-DSP 8x12 DL features an onboard isolated power supply, two stereo digital inputs, 6 analog inputs, 12 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 C-DSP 8x12 DL 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 analog circuitry driven by a 32-bit AKM convertor ensures pristine audio quality in any vehicle environment.

Four complete processing configurations are stored on-board and can be selected from the wired remote or by infrared remote control. An SD card slot supports offline configuration and firmware upgrade.

1.1 Typical usage

The C-DSP 8x12 DL typically connects to a head unit with up to six outputs. Figure 1 illustrates a straightforward installation, with four channels supplied by the head unit and four speakers being driven by a power amplifier. The included miniDSP UMIK-1 is used to perform Dirac Live calibration.

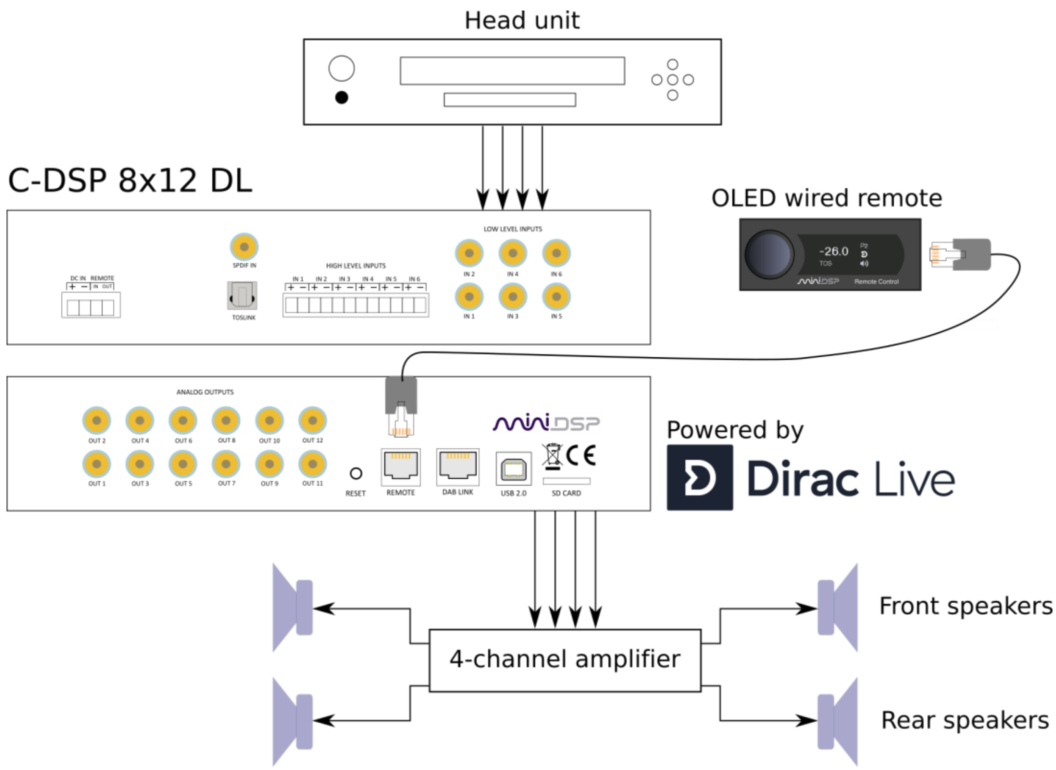


Figure 1. Basic system setup with four-channel acoustic correction

The C-DSP 8x12 DL also supports more advanced configurations. With up to 12 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:

- From stereo up to full 5.1 surround input from head unit supported.
- 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.
- Stereo digital input source switchable to any combination of outputs.
- 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 C-DSP 8x12 DL. This example implements active speakers on all four speaker channels and integrates one or two subwoofers.

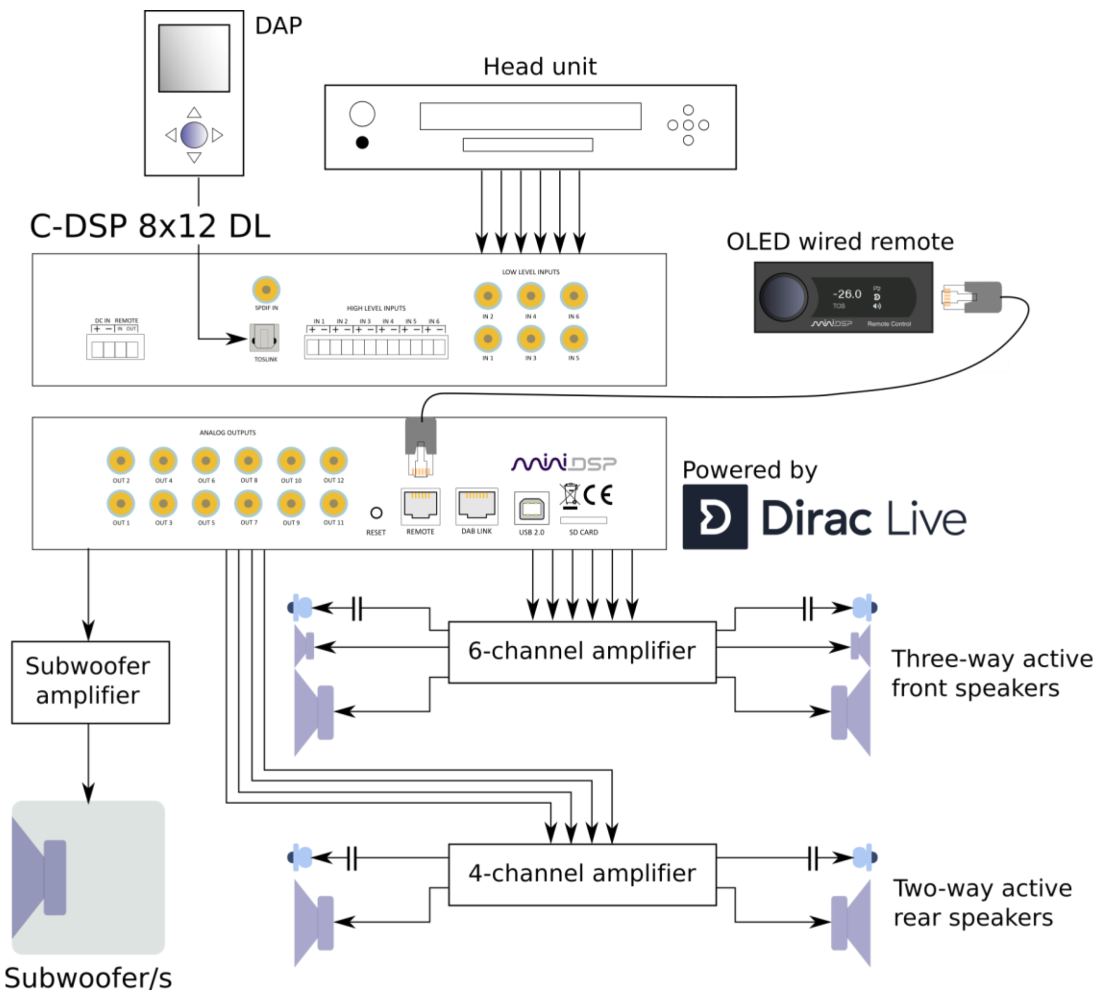
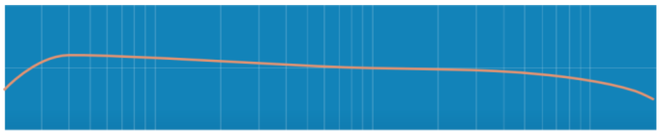


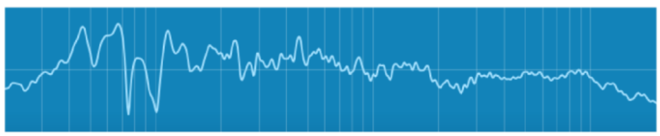
Figure 2. Advanced system configuration with multiple sources and active speakers

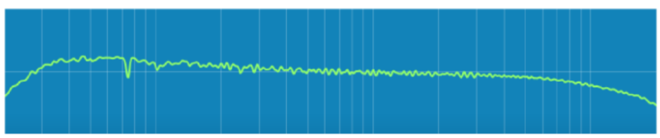
1.2 Dirac Live

The C-DSP 8x12 DL executes Dirac Live® digital room correction, from Dirac Research. 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.

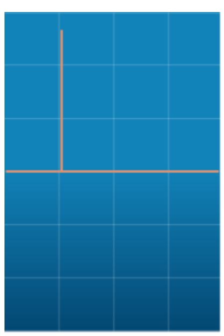
To accomplish its remarkable improvement in listening quality, the **DiracLive** application steps you through the procedure for taking measurements around your listening area. Dirac Live® employs a sophisticated analysis algorithm to make the optimal correction across the *whole* listening area, not 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.

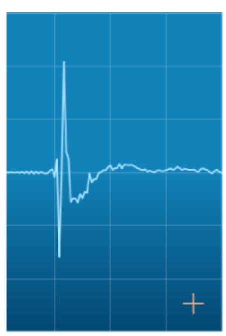

Target magnitude response


Uncorrected magnitude response


Corrected magnitude response

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.


Ideal
impulse response


Uncorrected
impulse response


Corrected
impulse response

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

2 SOFTWARE INSTALLATION

The C-DSP 8x12 DL is configured with software running on a PC or Mac. There are two sets of software to download and install, from live.dirac.com and from miniDSP.com.

2.1 WINDOWS 10

This section describes software download and installation for Windows 10.

The software described in this section runs on the latest version of Windows 10 **only**. Other versions of Windows are not supported by the current version of Dirac Live. If you are running an unsupported version of Windows, refer to the C-DSP 8x12 DL User Manual Legacy Support version.

2.1.1 Download and install the DiracLive application

Download the **DiracLive** application for Windows 10 from https://live.dirac.com/download/.

Double-click on the downloaded installer to run it. It will have a name like **diraclive-v3.0.14-setup.exe**. We recommend that you accept the default installation settings. Do not run the application yet.

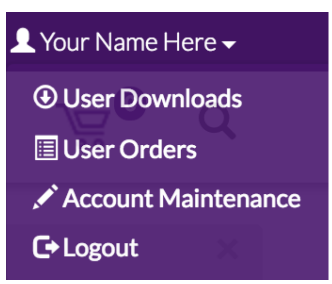
2.1.2 Download the miniDSP software

If you purchased your processor directly from miniDSP, your software will be available from the User Downloads 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 **C-DSP 8x12 DL Dirac 3.0**.

After downloading, unzip the file (right-click and select "Extract All..."). The unzipped download has a name like **C_DSP_8x12_DL_v1_11_mbfw_v2_11_oledfw_v2_3**. (The version numbers embedded in the folder name may be different.)

2.1.3 Install the miniDSP software

2.1.3.1 Possible Windows installation issues

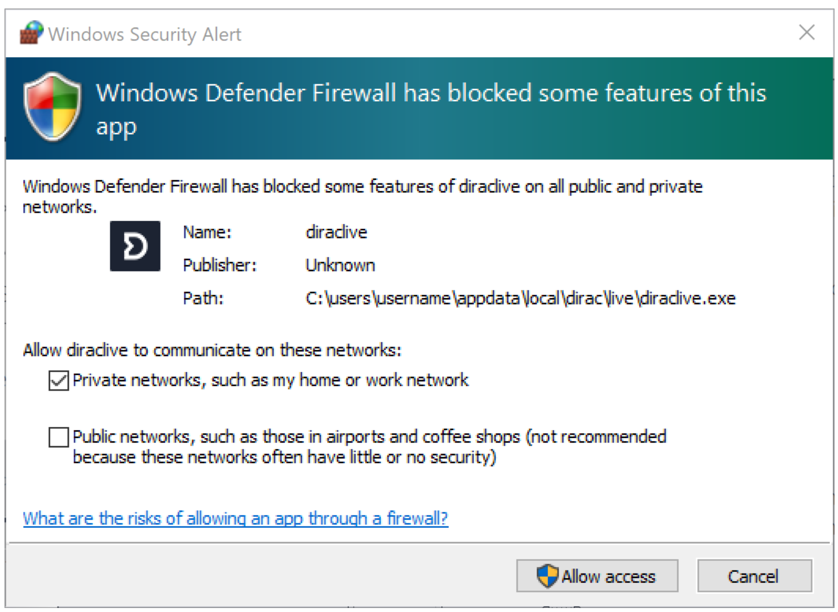
The miniDSP software requires that a number of other frameworks be installed for it to work. These packages should be installed automatically, but you can manually install them if you receive an error message that required software is missing.

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

2.1.3.2 Install the plugin

1. Navigate to the **Plugins** folder of the software download and then to the **Windows** folder.
2. Double-click on the plugin installer to run it. It will be named **C-DSP-8x12-DL.exe**. 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, macOS 10.15 Catalina or macOS 11 Big Sur **only**. Other versions of OS X / macOS are not supported by the current version of Dirac Live. If you are running an unsupported version of OS X / macOS, refer to the C-DSP 8x12 DL User Manual Legacy Support version.

2.2.1 Download and install the DiracLive application

Download the **DiracLive** application for macOS from https://live.dirac.com/download/.

1. Double-click on the downloaded file to unzip it. Then double-click on the unzipped installer file to run it. It will have a name like **DiracLive v3.0.14 Setup Darwin.app**. We recommend that you accept the default installation settings. Do not run the application yet.

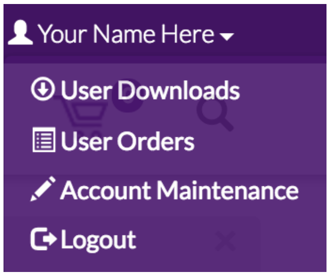
2.2.2 Download the miniDSP software

If you purchased your processor directly from miniDSP, your software will be available from the User Downloads 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 **C-DSP 8x12DL Dirac 3.0**.

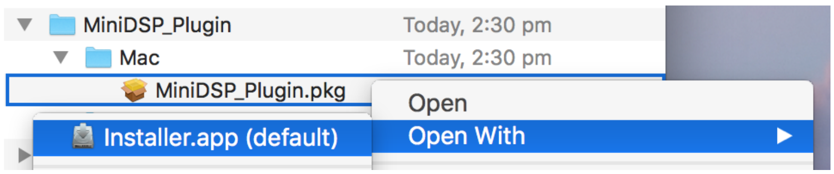
After downloading, unzip the file (double-click on it). The unzipped download has a name like **C_DSP_8x12_DL_v1_11_mbfw_v2_11_oledfw_v2_3**. (The version numbers embedded in the folder name may be different.)

2.2.3 Install the miniDSP software

2.2.3.1 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.)

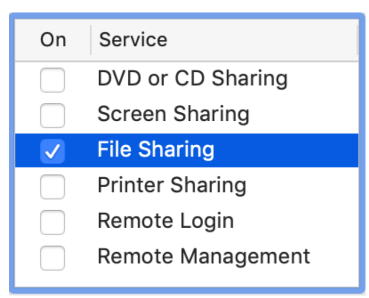


3. The following window will appear. Click on “Open.”

2.2.3.2 Install the plugin

1. Navigate to the **Plugins** folder of the software download and then to the **Mac** folder.
2. The installer is named **C-DSP-8x12-DL.pkg**. To run it, double-click on it, or right-click and open as described above. We recommend that you accept the default installation settings.
3. To run the 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.

2.2.3.3 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.

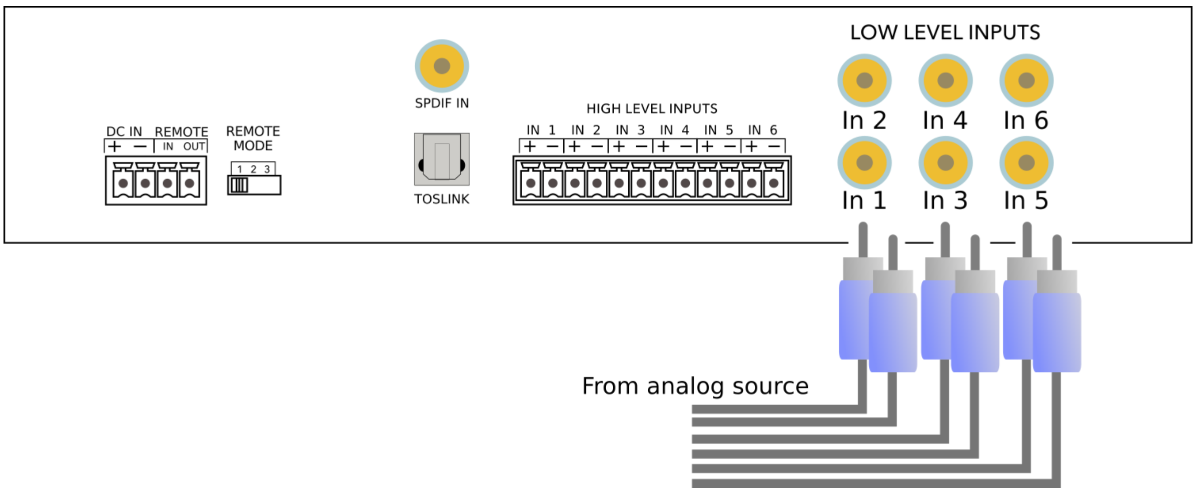
3 HARDWARE CONNECTIVITY

Connections to the C-DSP 8x12 DL are made on the front and rear panels.

3.1 ANALOG INPUTS

3.1.1 Low-level inputs

Low-level analog connections are made directly to the RCA jacks on the front panel. Be sure to take careful note of the channel numbering shown in this diagram and on the front panel. These inputs accept a maximum input voltage of either 2 or 4 VRMS, depending on the input sensitivity switch setting (see page 55).



Note that these are fully differential inputs. A regular RCA-RCA cable be used to connect from equipment with single-ended outputs, as shown in Figure 3.

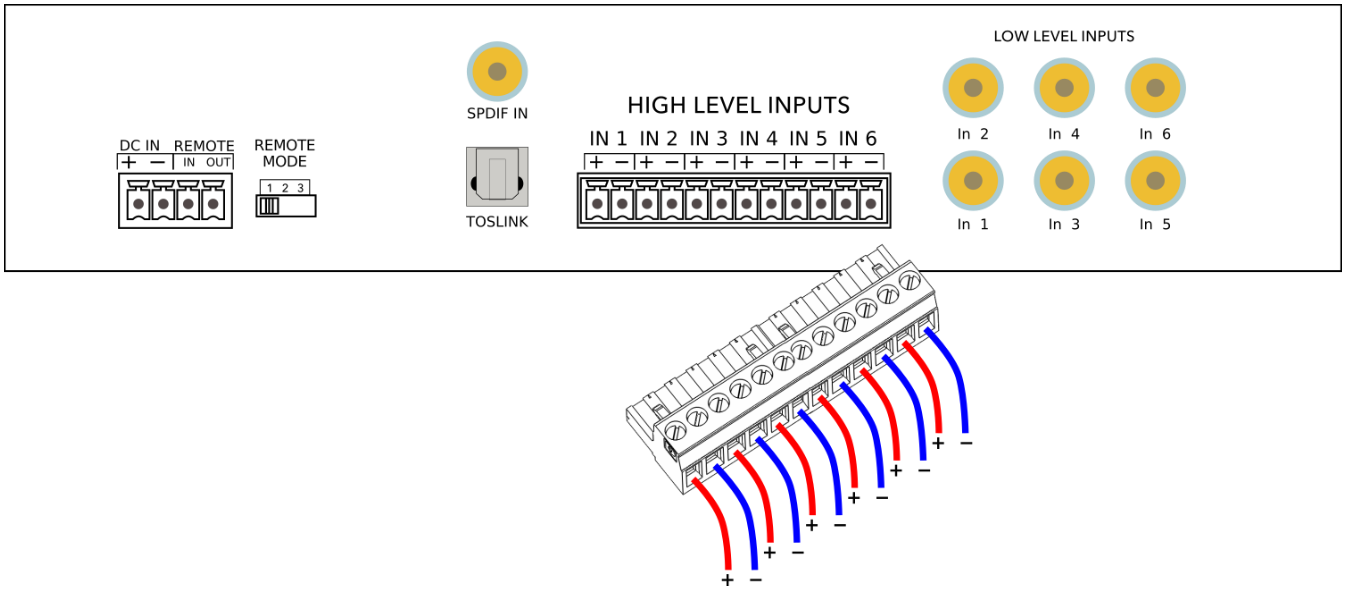
Figure 3. Single-ended RCA connection

If connecting to equipment with balanced output, connect the negative or "cold" leg to the RCA shield and the positive or "hot" leg to the RCA tip, as shown in Figure 4.

Figure 4. Connecting a balanced source to the C-DSP 8x12 DL.

3.1.2 High-level inputs

High-level (speaker-level) connections can be made by connecting bare wire ends to the push-in terminal block. Remove the terminal block and connect individual positive and negative wires to each screw terminal. After all connections are secure, firmly re-insert the terminal block.



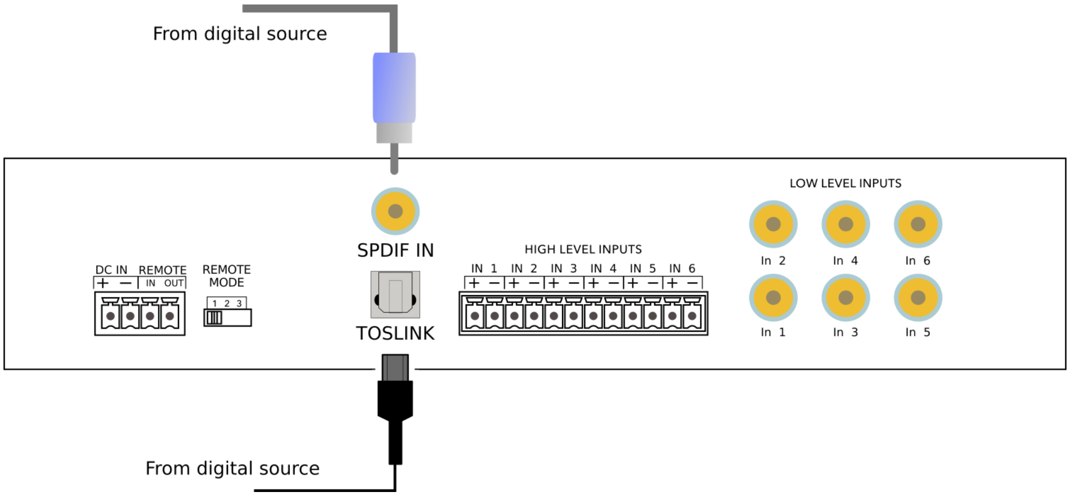
The high-level inputs have an input impedance of 68 Ω and are designed for connection to the speaker outputs of a head unit. The inputs are fully differential, therefore:

- Amplifiers with bridged outputs can be used.
- Do **not** connect the “–” outputs together. This can potentially damage the amplifiers in your head unit.

The maximum (differential) input voltage is either 8 or 12 V RMS, depending on the input sensitivity switch setting (see page 55). The 8 V RMS setting corresponds to a maximum amplifier power of 16 Watts into 4 ohms. The 12 V RMS setting corresponds to a maximum amplifier power of 36 Watts into 4 ohms.

3.2 DIGITAL INPUT

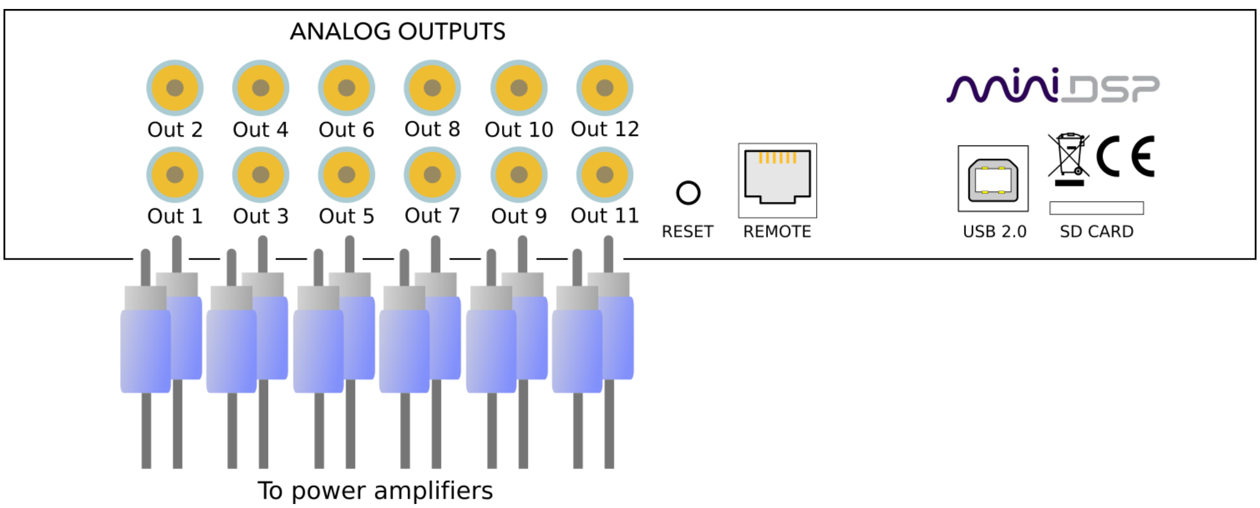
Two digital sources can be connected via the S/PDIF coax (RCA) connector and the TOSLINK (optical) connector. Switching between sources can be done from within the plugin or with an infrared remote control.



Note: the digital inputs accept only a stereo PCM digital signal. They do not accept encoded or multichannel digital audio.

3.3 ANALOG OUTPUTS

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

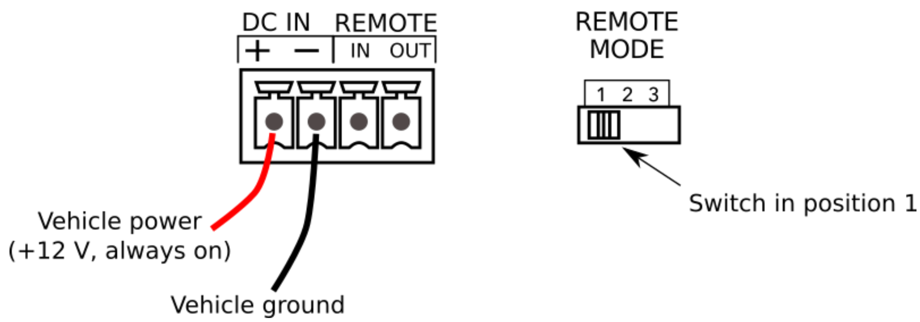


3.4 DC POWER

The C-DSP 8x12 DL incorporates an isolated DC-DC power convertor and is designed for direct connection to the vehicle's power supply (nominally 12 V DC). Power is connected via a four-way terminal block. There are two modes of operation, described below.

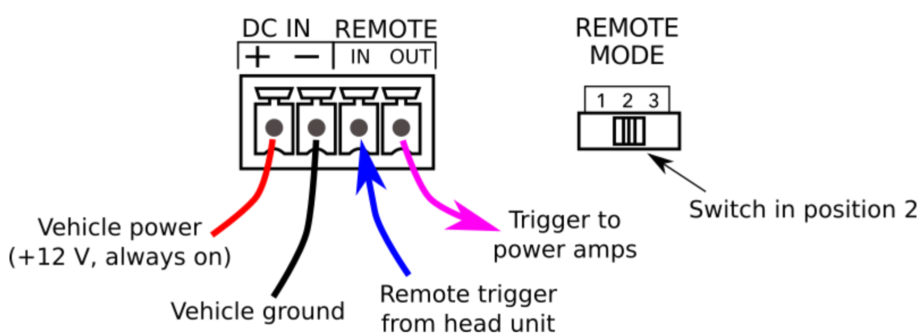
Powered on power (position 1)

To power on the C-DSP 8x12 DL whenever 12 V DC is applied to the **DC IN** terminals, set the **REMOTE MODE** switch to position 1. Typically, power is provided from the main vehicle supply (always on).



Remote trigger (position 2)

To reduce battery drain, the remote trigger option should be used. To enable remote trigger, set the **REMOTE MODE** switch to position 2. In this case, the C-DSP 8x12 DL is powered on when the voltage on the **REMOTE IN** terminal exceeds 4 V DC. Typically, **REMOTE IN** is connected to the remote trigger output from the head unit.

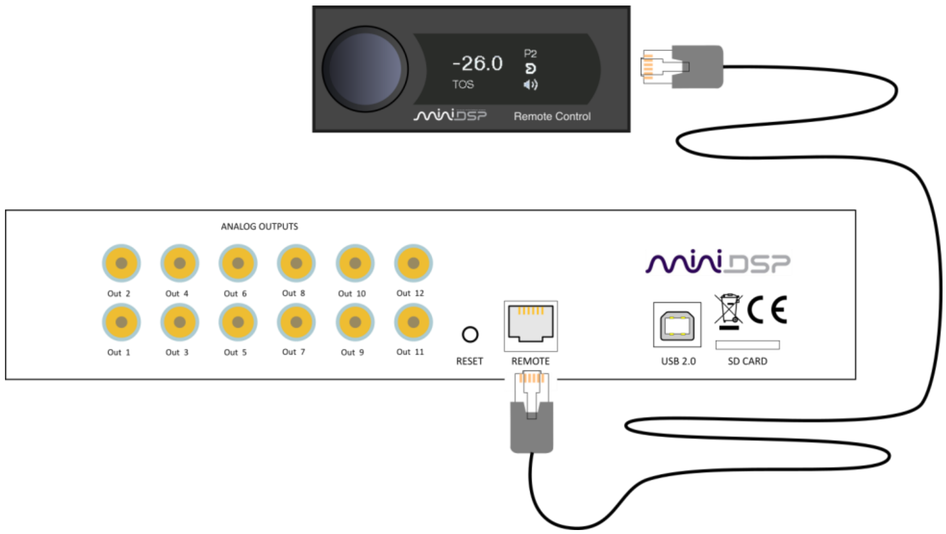


In this mode, the **REMOTE OUT** terminal should be used to turn on the power amplifier(s). There is a time delay between **REMOTE IN** going positive and **REMOTE OUT** going positive. This can be configured in the plugin – see page 56.

Position 3 of the **REMOTE MODE** switch is reserved for future use. Do not set the switch to position 3.

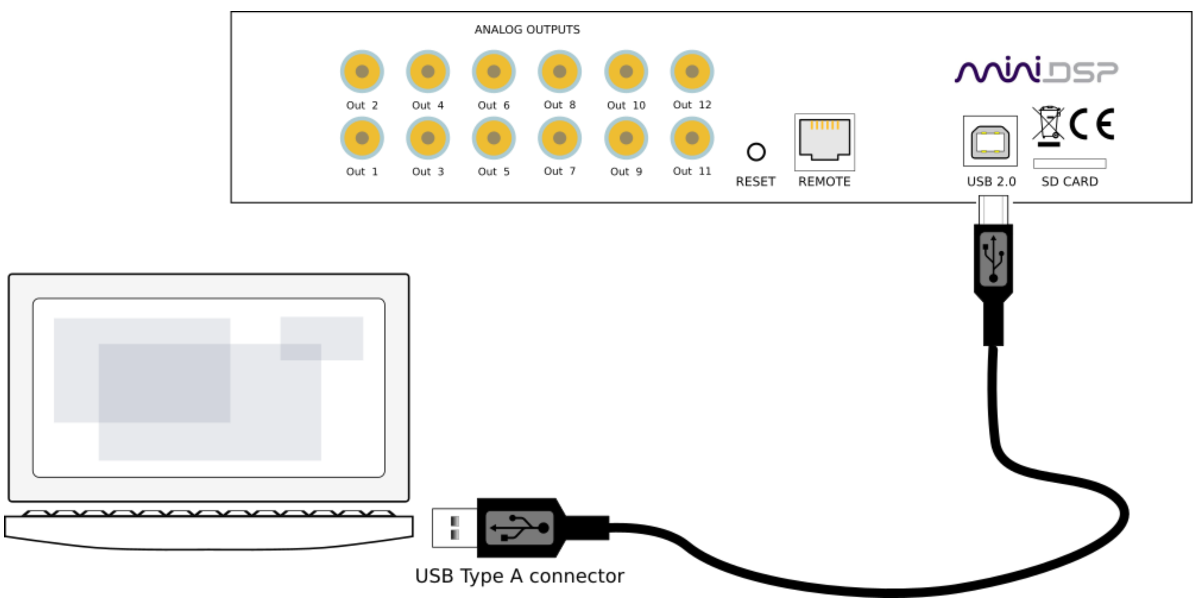
3.5 WIRED REMOTE

The wired remote can be positioned anywhere in the vehicle and is connected via the supplied RJ11 phone cable. The plugs on each end simply plug into the base of the remote and into the rear panel of the C-DSP 8x12 DL. The wired remote also contains the receiver for infrared remote control.



3.6 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.



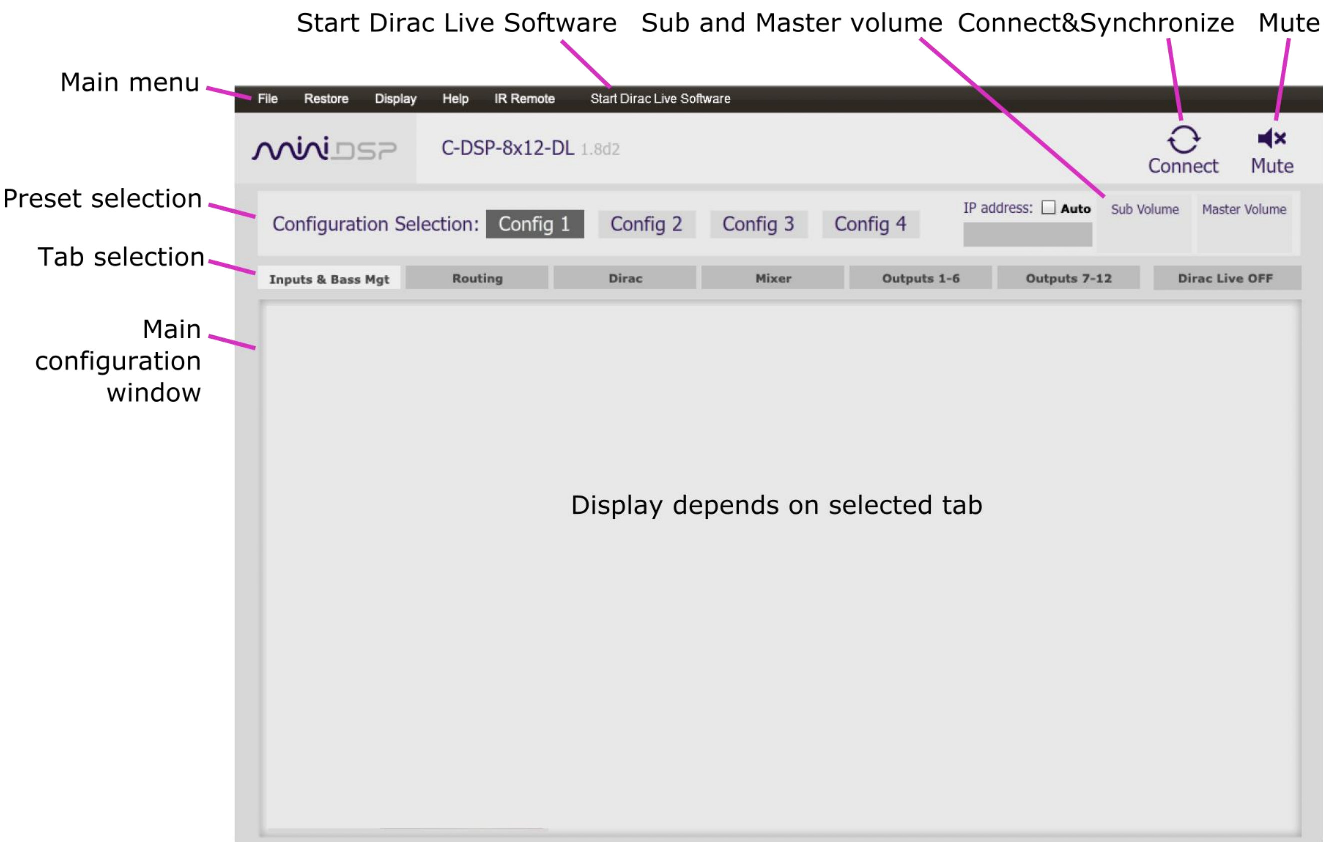
Configurations can also be loaded from a micro SD card – see page 52.

4 PLUGIN OVERVIEW

The overall power of the C-DSP 8x12 DL comes not only from its comprehensive I/O but it's 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:



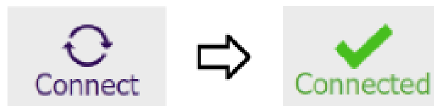
During *initial* configuration of the processor, it is strongly recommended that any connected amplification be muted or powered off.

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.

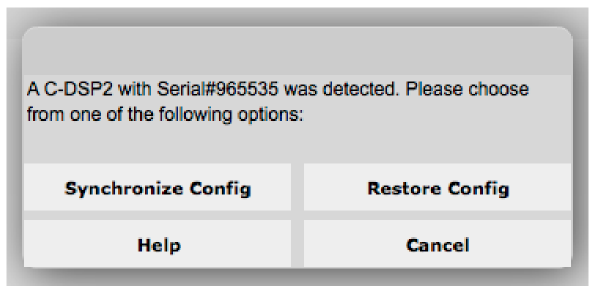
4.2 SYNCHRONIZING WITH THE PROCESSOR

Communication with the C-DSP takes place over USB. 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 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.

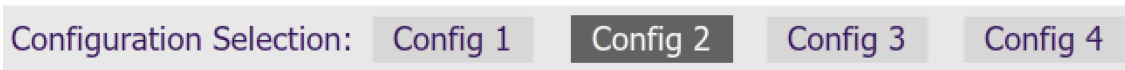
4.3 GLOBAL CONTROLS

4.3.1 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:

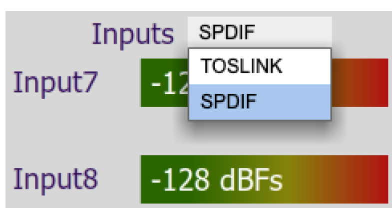


4.3.2 Tab selection

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

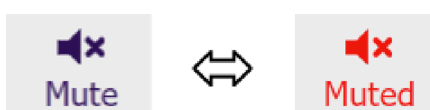
4.3.3 Digital input selection

When the plugin is connected to the C-DSP 8x12 DL, the currently select digital input appears on the **Inputs & Bass Mgt** tab, just above the level meters for channels 7 and 8. Click on it to pop up the selection menu and select the desired input.



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 C-DSP 8x12 DL 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 39.

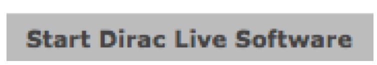
The current values for master and subwoofer volume are displayed in the plugin while connected:

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 Start Dirac Live Software

The **Start Dirac Live Software** button starts the separate **DiracLive** application. When it is pressed, the C-DSP 8x12 DL plugin will disconnect from the C-DSP 8x12 DL processor and then start up **DiracLive**.



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 C-DSP 8x12 DL processor.

4.3.8 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 C-DSP 8x12 combines Dirac Live with miniDSP's audio processing, refer to Figure 5. 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:

- Six analog input channels and one stereo digital pair 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 C-DSP 8x12 DL 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. If you wish to use the advanced features of the C-DSP 8x12 DL and are not familiar with acoustic measurement software, see page 57.

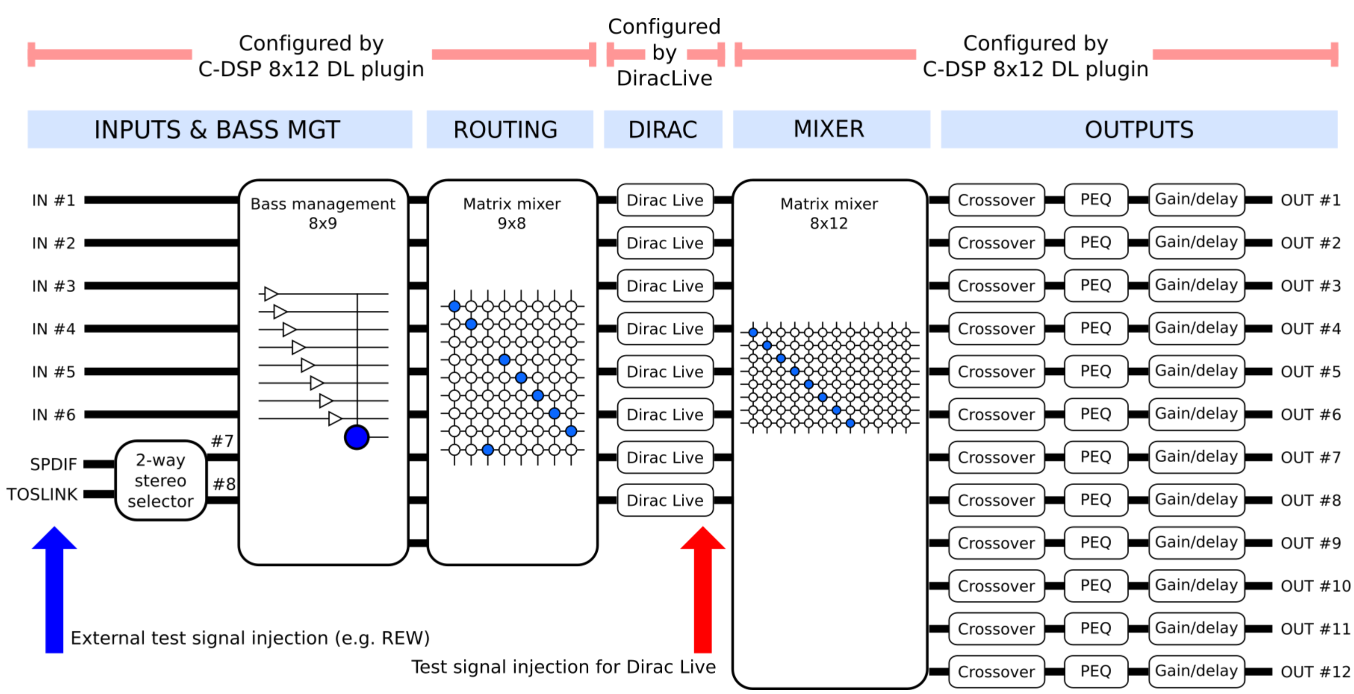
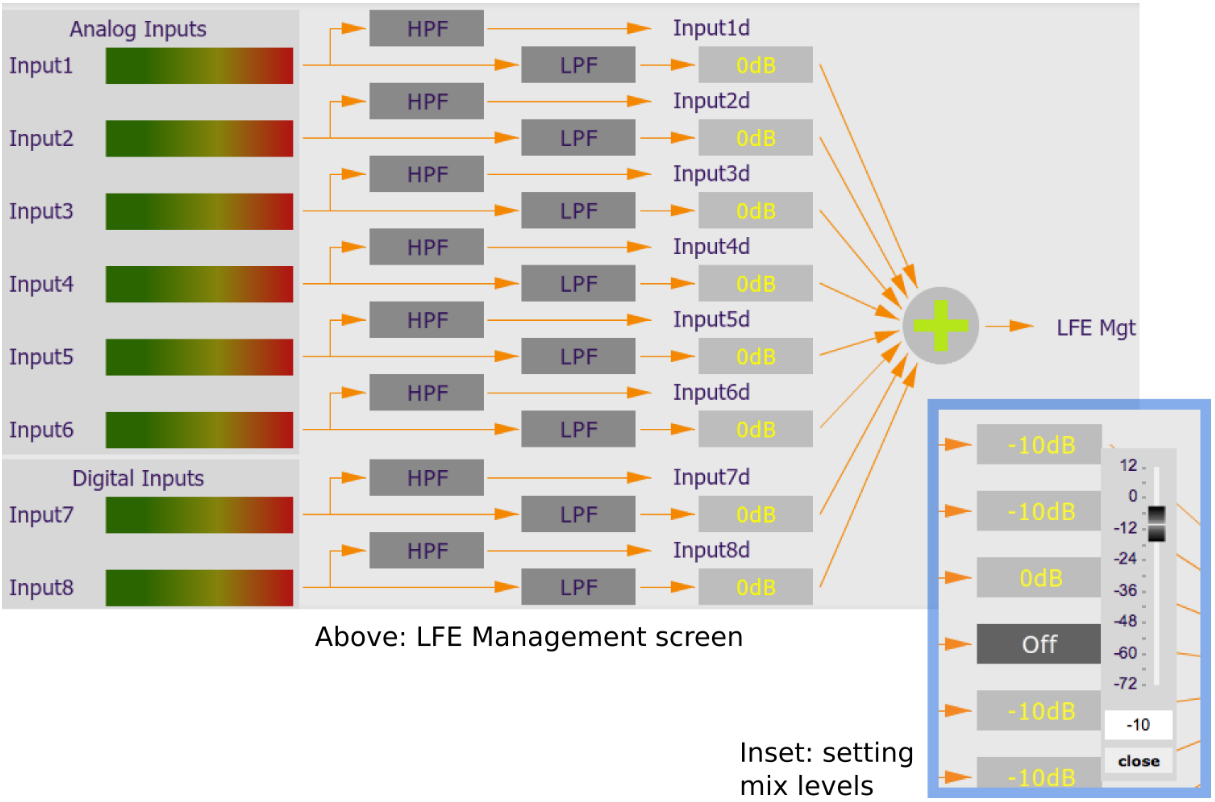


Figure 5. Signal flow through the C-DSP 8x12 DL

4.5 Inputs & Bass Mgt tab

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



Above: LFE Management screen

Inset: setting mix levels

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

This tab also provide selection between the digital TOSLINK and SPDIF inputs.

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:

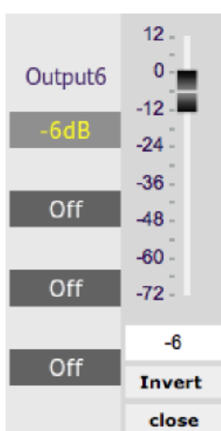
	Dirac 1	Dirac 2	Dirac 3	Dirac 4	Dirac 5	Dirac 6	Dirac 7	Dirac 8
Input1d	0dB	Off	Off	Off	Off	Off	Off	Off
Input2d	Off	0dB	Off	Off	Off	Off	Off	Off
Input3d	Off	Off	0dB	Off	Off	Off	Off	Off
Input4d	Off	Off	Off	0dB	Off	Off	Off	Off
Input5d	Off	Off	Off	Off	0dB	Off	Off	Off
Input6d	Off	Off	Off	Off	Off	0dB	Off	Off
Input7d	0dB	Off	Off	Off	Off	Off	0dB	Off
Input8d	Off	0dB	Off	Off	Off	Off	Off	0dB
Bass Mgt	Off	Off	Off	Off	Off	Off	Off	Off

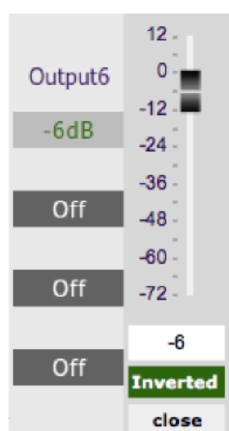
At each cross-point of the matrix, the input channel (labels along the left) is mixed into the corresponding output channel (labels along the top) if the lettering is highlighted in yellow. The cross-point is turned on and off by clicking on it. Any number of input channels can be mixed to each output channel.

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

At each cross-point, the gain of the mixed signal can be set to a value between -72 and +12 dB. To set the gain, right-click on the cross-point to pop up the gain control. Set the gain by moving the slider, or alternatively, type the value in directly into the entry box below, *press the Enter key*, then click the **close** button.

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 C-DSP 8x12 DL. (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:

	Input	Gain	Delay	Output
Dirac 1	-103 dBFs	-0.3 dB	1.23 ms	-191 dBFs
Dirac 2	-109 dBFs	-1.4 dB	1.13 ms	-199 dBFs
Dirac 3	-92 dBFs	0 dB	0.48 ms	-181 dBFs
Dirac 4	-109 dBFs	-0.4 dB	0 ms	-194 dBFs
Dirac 5	-109 dBFs	-2 dB	3.73 ms	-199 dBFs
Dirac 6	-109 dBFs	-1.6 dB	3.63 ms	-200 dBFs
Dirac 7	-116 dBFs	0 dB	0 ms	-116 dBFs
Dirac 8	-116 dBFs	0 dB	0 ms	-116 dBFs

Dirac Live ON

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.

Dirac Live OFF ⇔ Dirac Live ON

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.

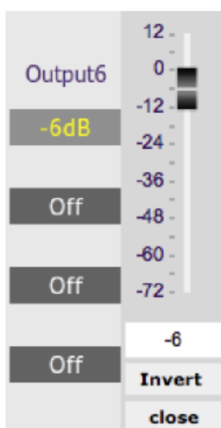
This is the default Mixer tab:

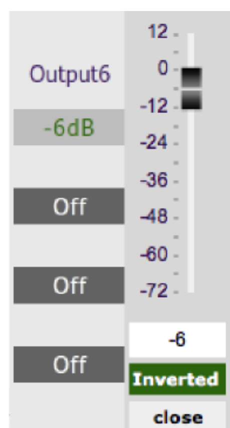
	Output1	Output2	Output3	Output4	Output5	Output6	Output7	Output8	Output9	Output10	Output11	Output12
Dirac1	0dB	Off	Off	Off	Off	Off	Off	Off	Off	Off	Off	Off
Dirac2	Off	0dB	Off	Off	Off	Off	Off	Off	Off	Off	Off	Off
Dirac3	Off	Off	0dB	Off	Off	Off	Off	Off	Off	Off	Off	Off
Dirac4	Off	Off	Off	0dB	Off	Off	Off	Off	Off	Off	Off	Off
Dirac5	Off	Off	Off	Off	0dB	Off	Off	Off	Off	Off	Off	Off
Dirac6	Off	Off	Off	Off	Off	0dB	Off	Off	Off	Off	Off	Off
Dirac7	Off	Off	Off	Off	Off	Off	0dB	Off	Off	Off	Off	Off
Dirac8	Off	Off	Off	Off	Off	Off	Off	0dB	Off	Off	Off	Off

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.

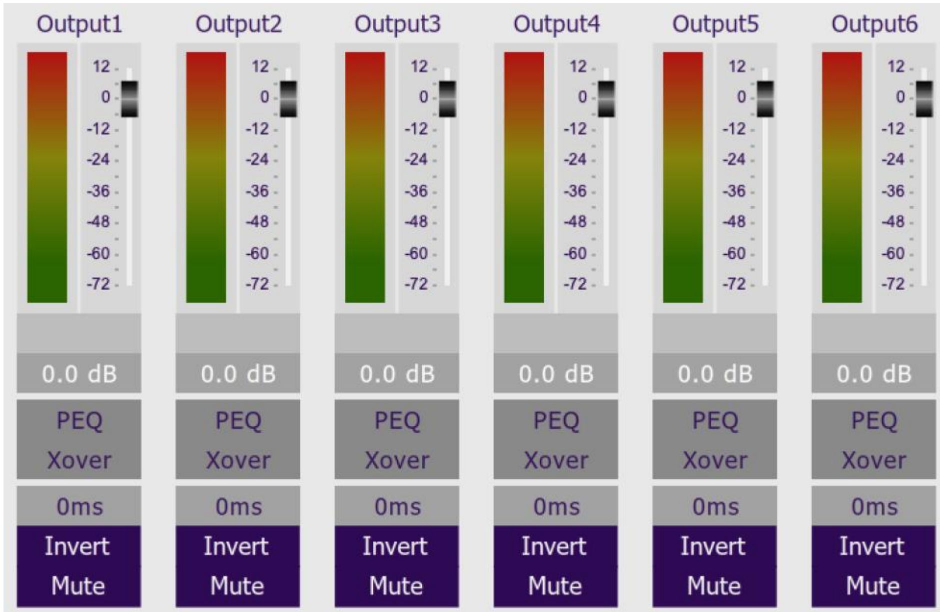
As with the Routing tab, each cross-point has adjustable level and an Invert option.



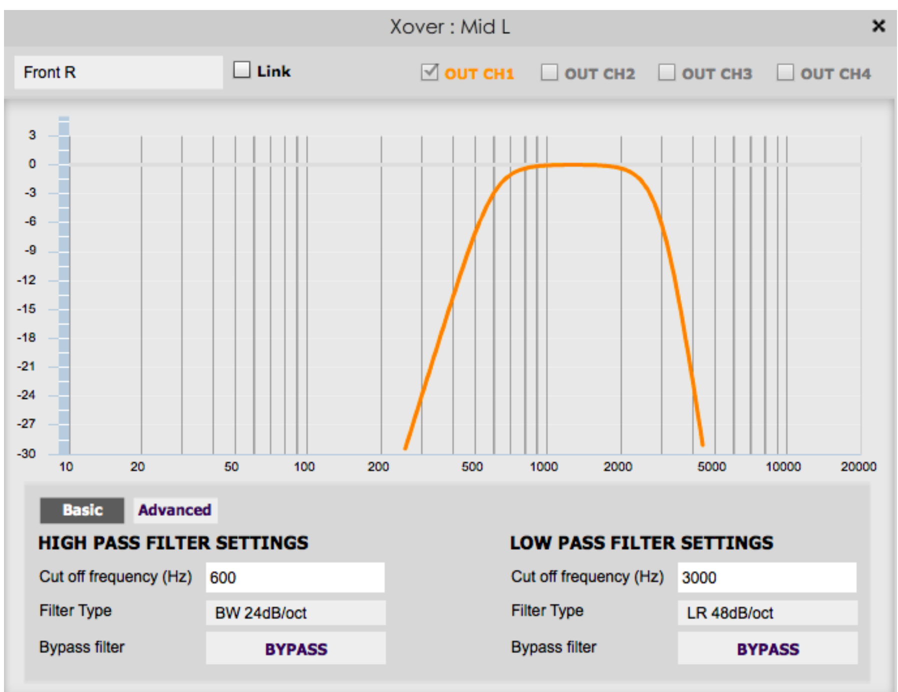


4.9 OUTPUT TABS

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, gain adjustment and level meter, and signal processing (see the Plugin Reference starting on page 42 for full details). For example, the crossover screen opened by clicking on the **Xover** button looks like this:



4.10 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.

4.10.1 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 6 illustrates the signal flow and processing in dark blue, overlaid on the signal flow diagram from Figure 5. 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.

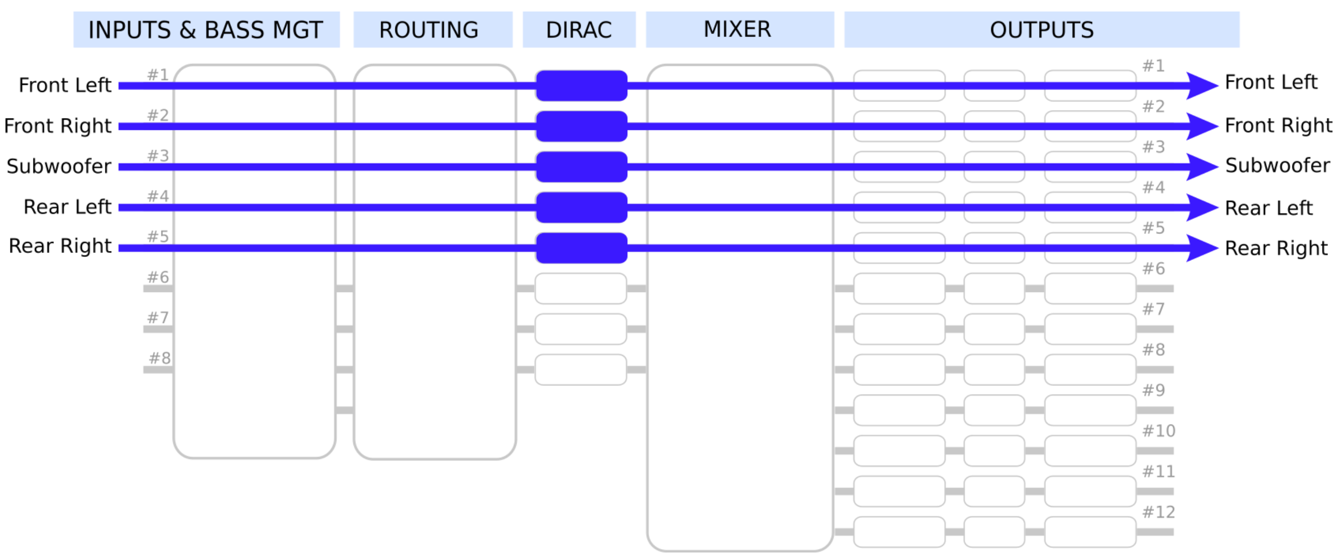


Figure 6. Straight through processing in the C-DSP 8x12 DL

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.

4.10.1 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 7 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.

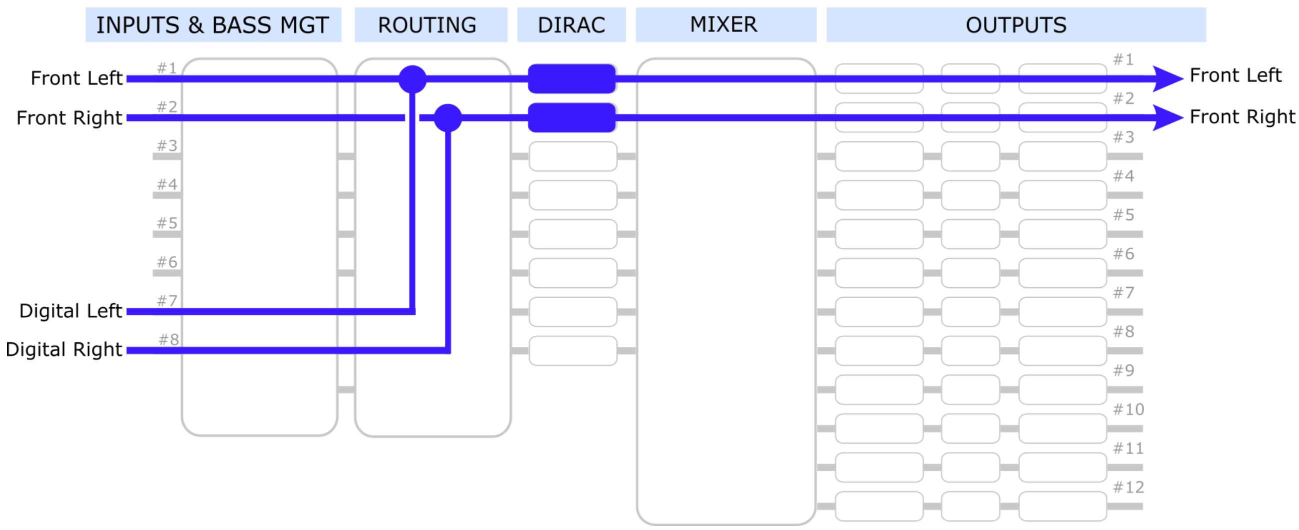


Figure 7. Routing the stereo digital input

Use the Routing tab to implement this routing:

4.10.2 Bass management

You can synthesize a subwoofer signal from the speaker inputs. Figure 8 illustrates the signal flow and processing for a typical system with four input channels. The Bass Management processing:

- Applies high pass filters (“HPF”) to the signal sent to Dirac Live and then to 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.

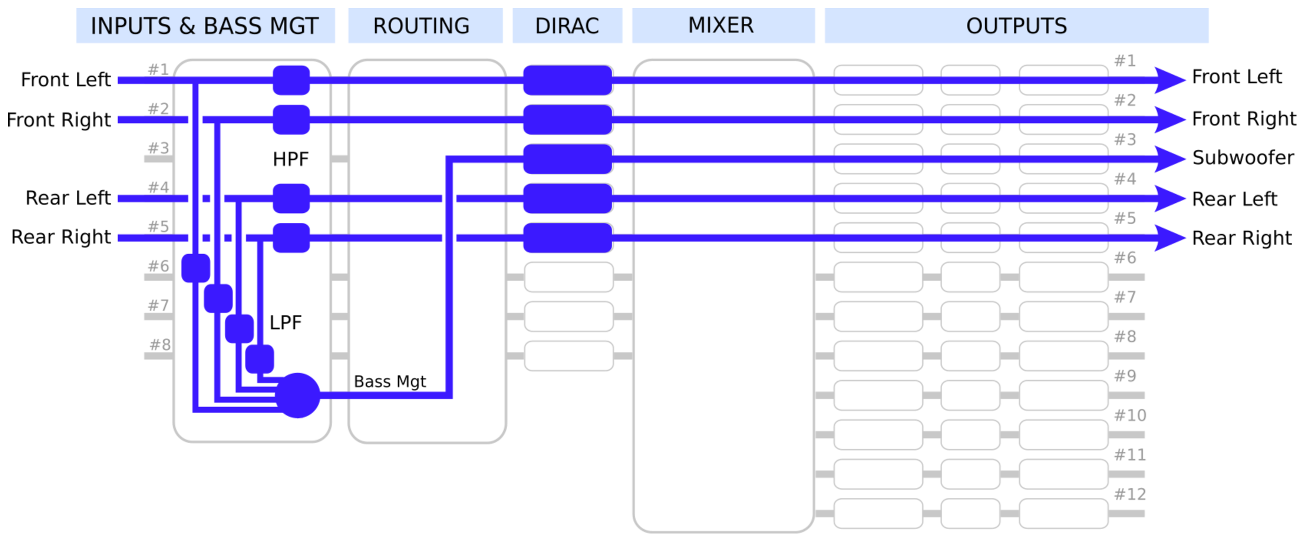
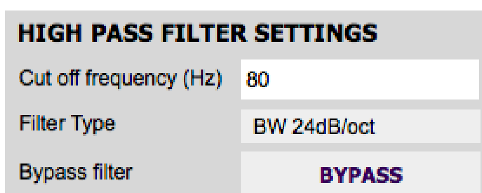


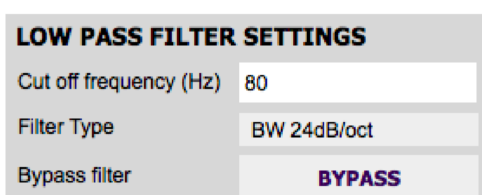
Figure 8. Bass management in the C-DSP 8x12 DL

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

In Figure 8, we're assuming that you don't have a subwoofer output from your head unit. If you do, you will want to connect it as well. In that case, the processing flow will be as shown in Figure 9. In this example, we have left out the rear channels to keep the diagram manageable. 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), then routed via the Bass Mgt signal.

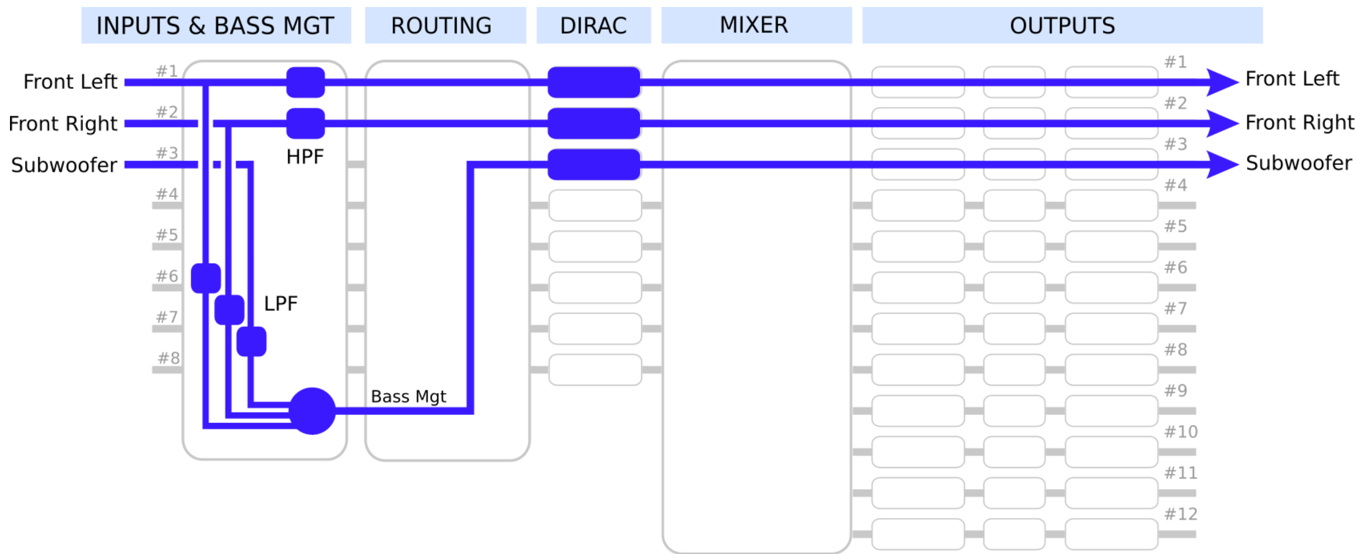


Figure 9. Bass management when you have a subwoofer input feed

In either case, 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:

Inputs & Bass Mgt | Routing | Dirac | Mixer

	Dirac 1	Dirac 2	Dirac 3	Dirac 4	Dirac 5
Front L	0dB	Off	Off	Off	Off
Front R	Off	0dB	Off	Off	Off
⋮					
Bass Mgt	Off	Off	0dB	Off	Off

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.

4.10.3 Dual subwoofers

Multiple subs can be connected to separate outputs of the C-DSP 8x12 DL. 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 10.

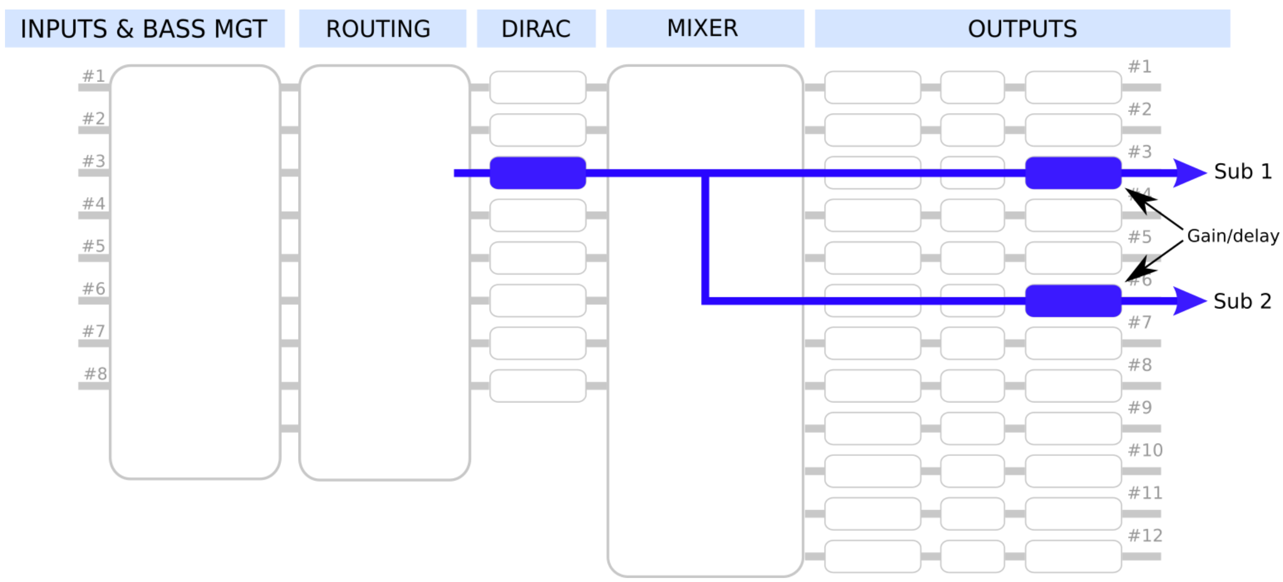


Figure 10. Processing flow example for multiple subwoofers

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:

Inputs & Bass Mgt | Routing | Dirac | Mixer

	Front L	Front R	Sub 1	Rear L	Rear R	Sub 2
Dirac1	0dB	Off	Off	Off	Off	Off
Dirac2	Off	0dB	Off	Off	Off	Off
Dirac3	Off	Off	0dB	Off	Off	0dB
Dirac4	Off	Off	Off	0dB	Off	Off
Dirac5	Off	Off	Off	Off	0dB	Off

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

4.10.4 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 11.

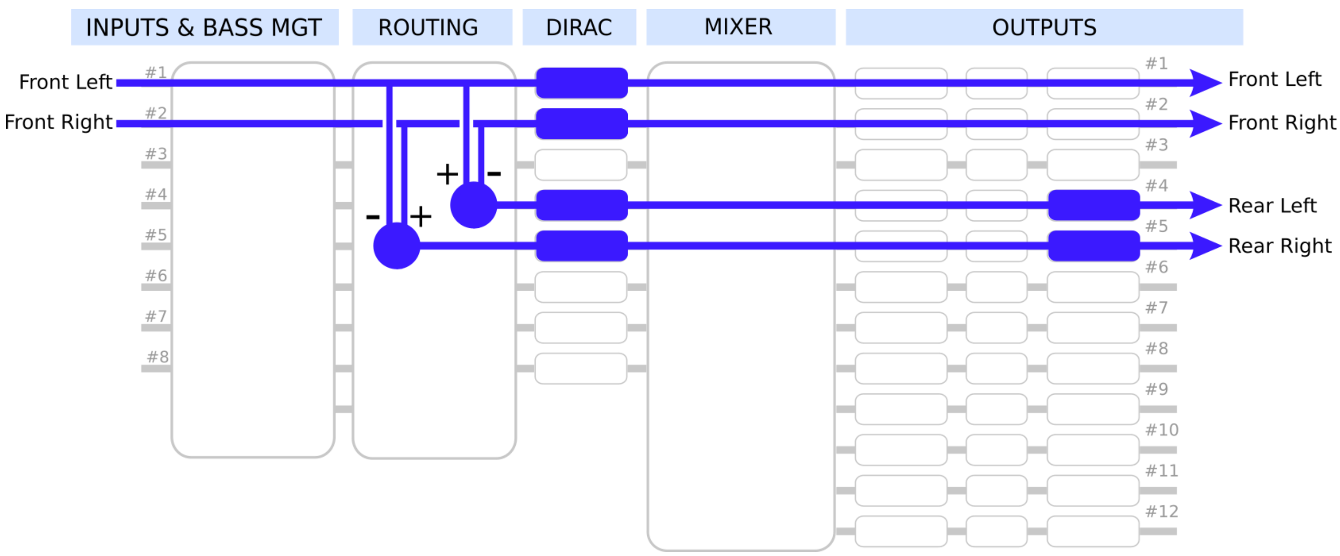


Figure 11. Processing flow for rear channel synthesis

The basic strategy is as follows:

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

	Dirac 1	Dirac 2	Dirac 3	Dirac 4
Left In	0dB	Off	-6dB	-6dB
Right In	Off	0dB	-6dB	-6dB

2. Add a delay of up to 20 ms to the rear channel outputs (experiment to see what sounds best to you).
3. 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.

4.10.5 Active speaker crossover

The C-DSP 8x12 DL can implement active speaker crossovers of any complexity up to the number of available output channels. An example processing flow is illustrated in Figure 12.

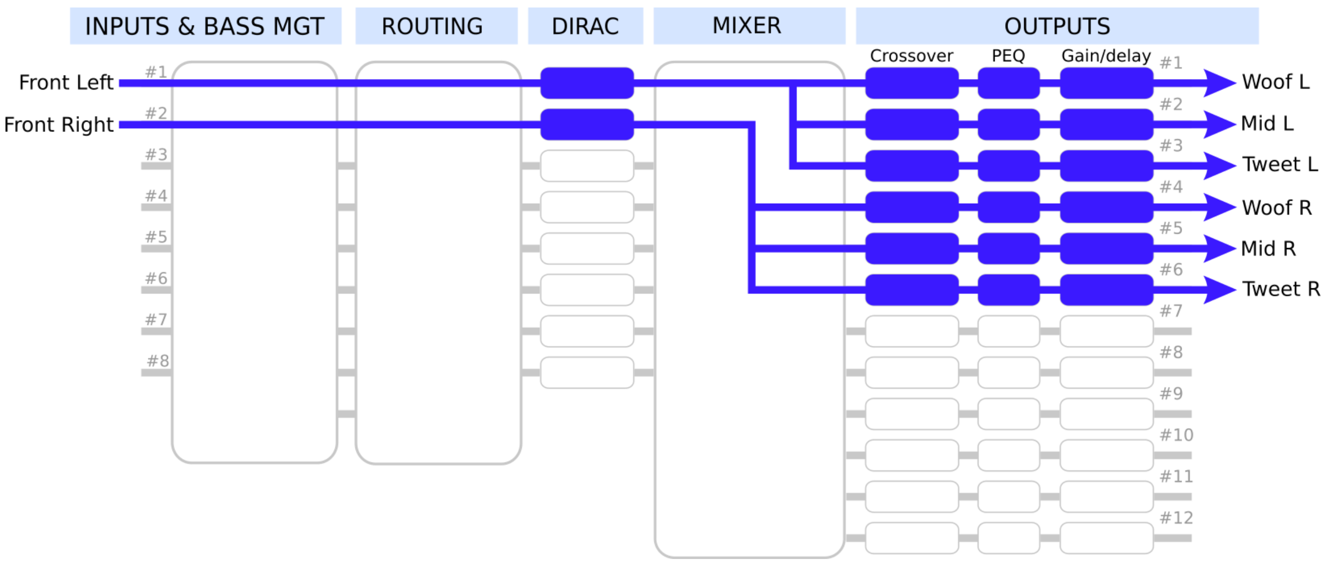


Figure 12. Processing flow for active speakers (three-way example)

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:

Inputs & Bass Mgt	Routing	Dirac	Mixer

	Woof L	Mid L	Tweet L	Woof R	Mid R	Tweet R
Dirac1	0dB	0dB	0dB	Off	Off	Off
Dirac2	Off	Off	Off	0dB	0dB	0dB

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 at this link. For example, the midrange crossover might look like this:

HIGH PASS FILTER SETTINGS		LOW PASS FILTER SETTINGS	
Cut off frequency (Hz)	600	Cut off frequency (Hz)	3000
Filter Type	LR 24dB/oct	Filter Type	LR 48dB/oct
Bypass filter	BYPASS	Bypass filter	BYPASS

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 12 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:

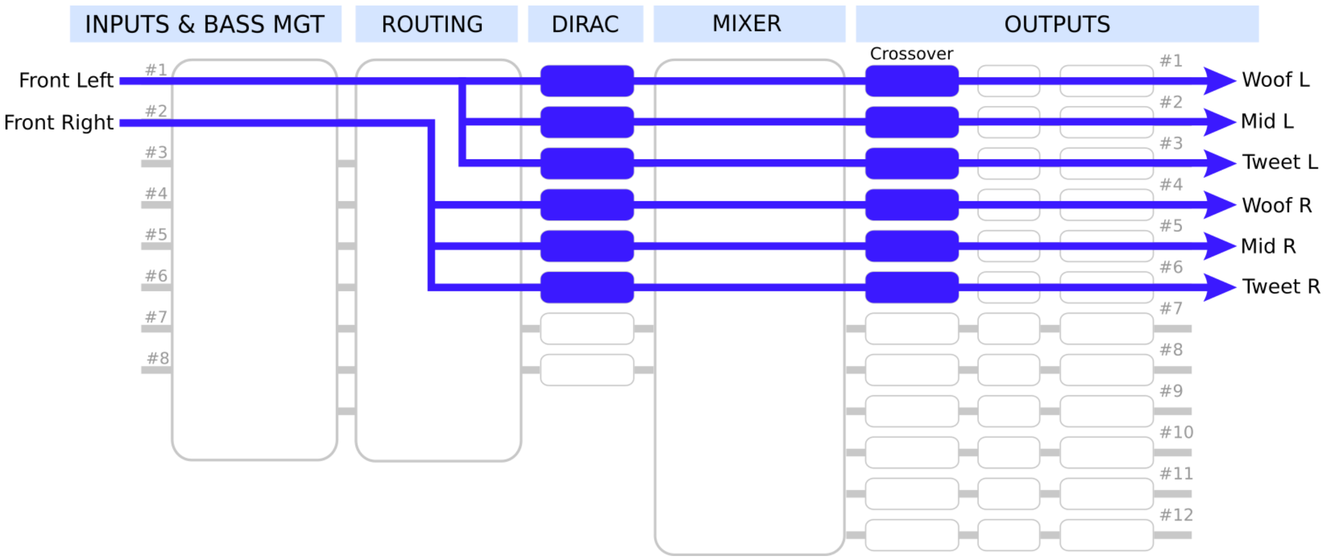


Figure 13. Alternative processing flow for active speakers (not recommended)

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:

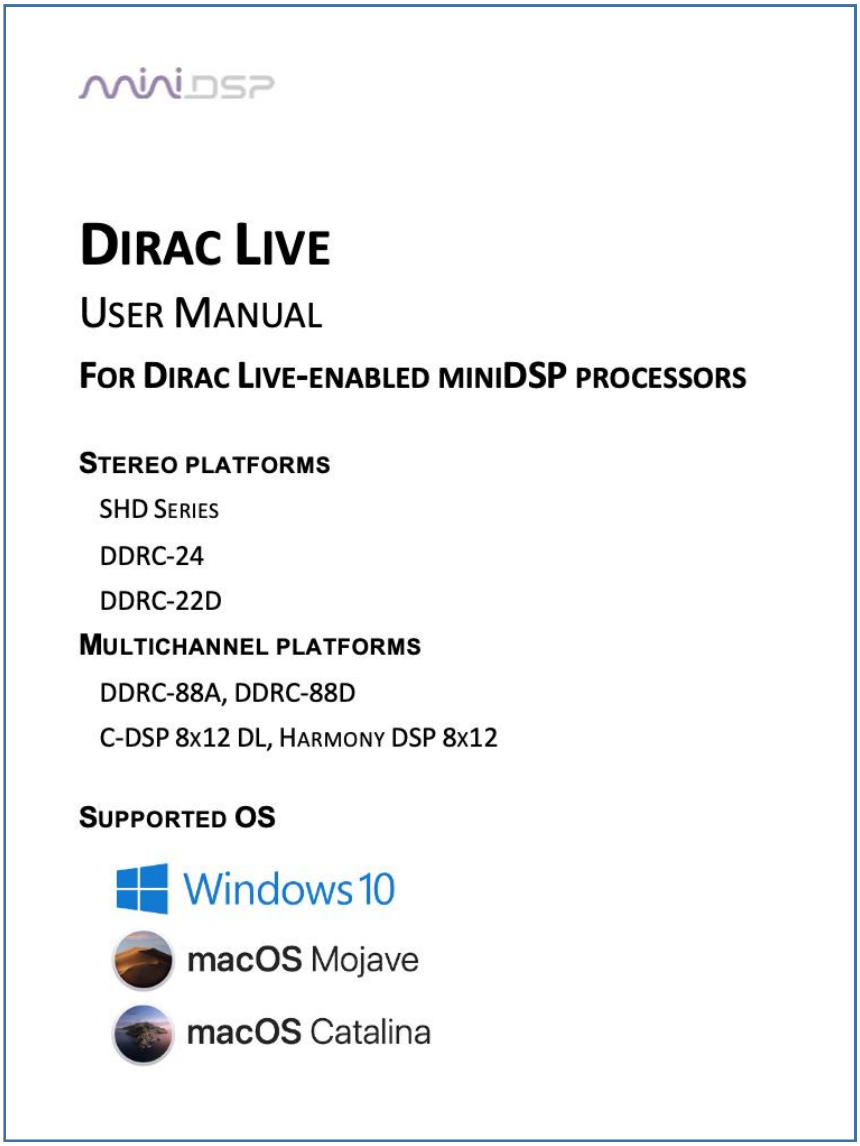
- Dirac Live will not be able to correct for the phase shift through the crossover(s). In other words, impulse response correction will not be working as expected.
- Dirac Live will not be able to correct the *combined* response of the drivers in the crossover region.

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.

5 Dirac Live Calibration

Dirac Live calibration with the C-DSP 8x12 DL is described in the separate miniDSP Dirac Live User Manual. It can be downloaded from the C-DSP 8x12 DL product page on our website.

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 C-DSP 8x12 DL processor.



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
- Master mute
- Volume mode selection (master/subwoofer)
- Preset selection
- Digital input source selection
- Dirac Live on/off

6.1 STATUS INDICATORS

The OLED display shows the currently selected preset, master volume, selected digital input source, 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 change the digital input source

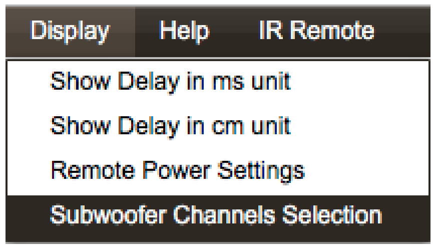
Quickly press the control knob three times. The display will change to show either “TOSLINK” or “SPDIF” in large letters. Rotate the knob until the desired selection is displayed, then press the control knob or wait for a second.

To enable or disable Dirac Live processing

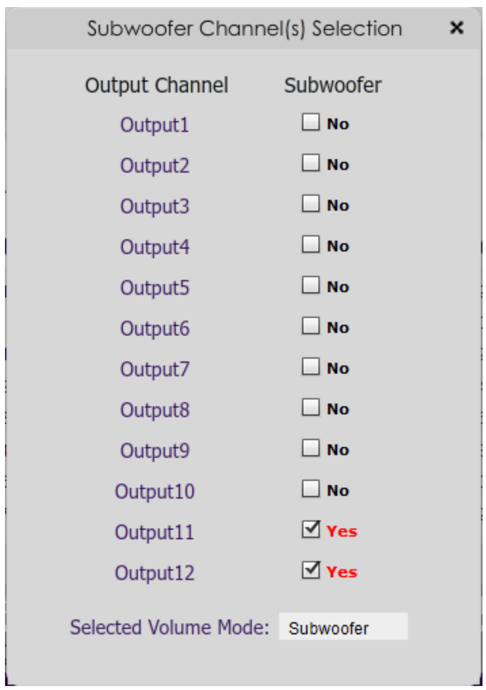
Quickly press the control knob four times. The display will change to show either “Dirac On” or “Dirac Off” in large letters. Rotate the knob until the desired selection (on or off) is displayed, then press the control knob or wait for a second.

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**.



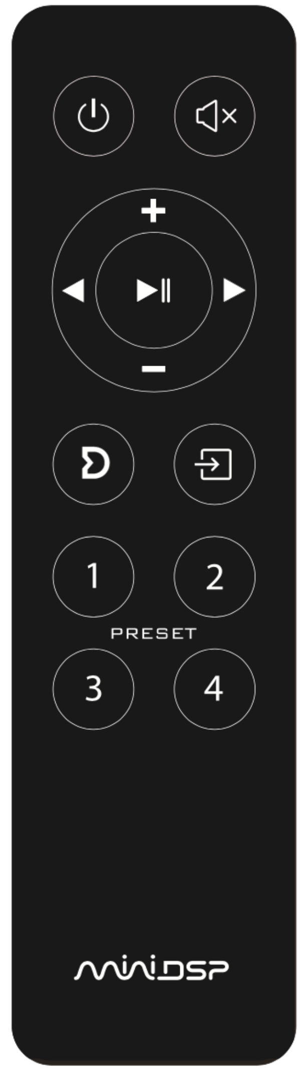
The selection dialog appears:



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 C-DSP 8x12 DL.

Mute
Mute and unmute audio output.

\+ –

Volume
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 C-DSP 8x12 DL.

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

Selects between TOSLINK and SPDIF digital inputs.

1

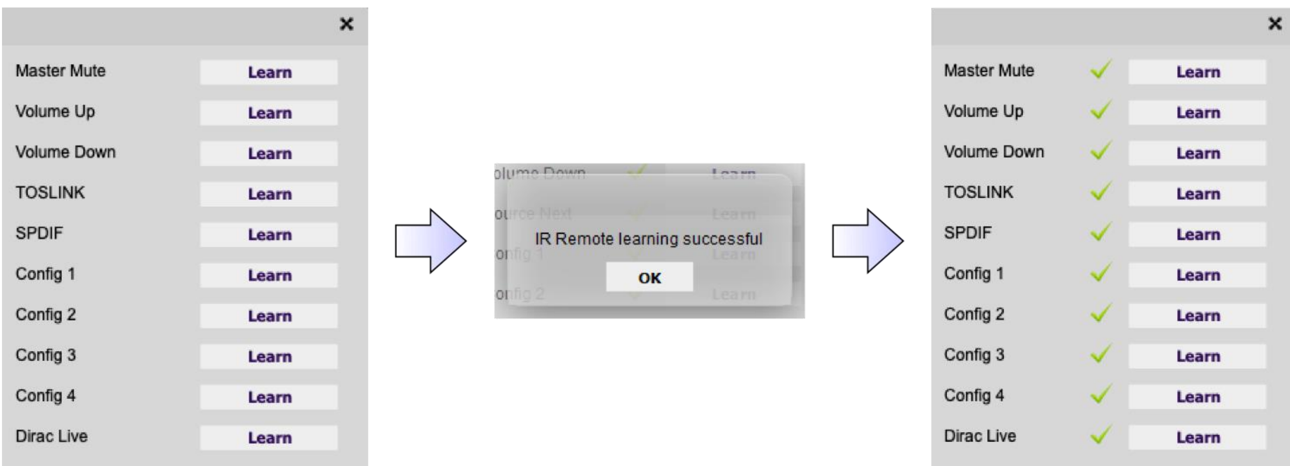
Preset (1 through 4)
Switch to the selected preset. It takes a few seconds for the preset selection to complete, while the processor loads the new filters from its flash memory into the DSP.

6.5 LEARNING THIRD-PARTY REMOTE CODES

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

- 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:



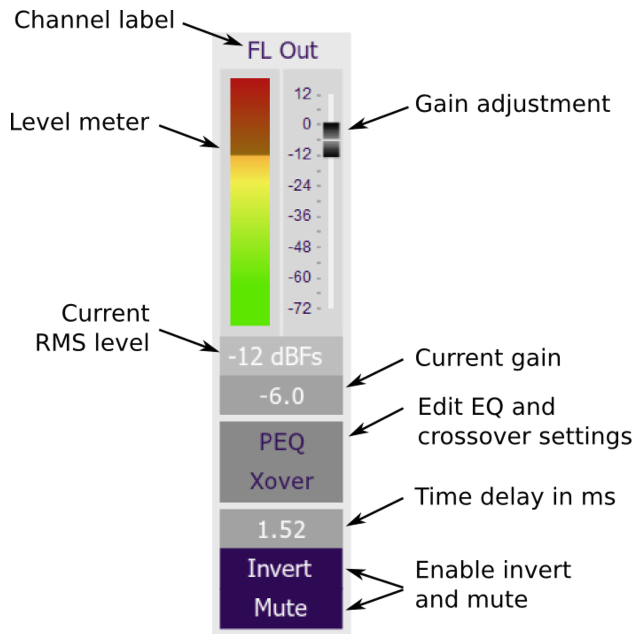
To "unlearn" a command, press the **Learn** button and wait for the plugin to time out. Note that you cannot "learn" the miniDSP remote – it will always work, even if you learn another remote's codes.

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.)

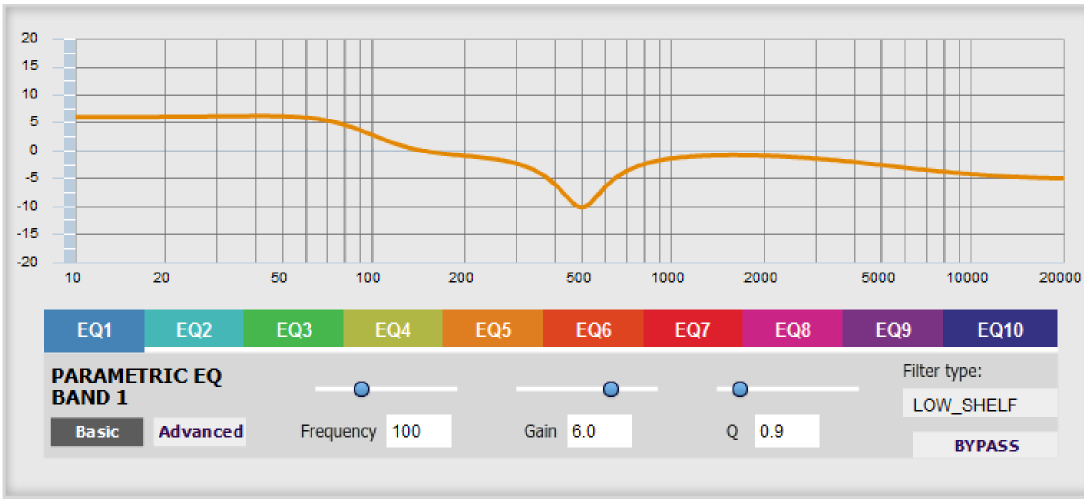
Gain adjustment

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.

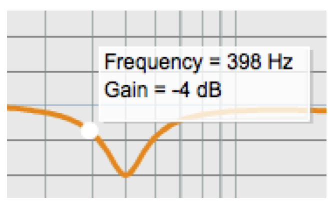
7.1.3 Parametric EQ (PEQ)

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 48.

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.

Enabled Bypassed

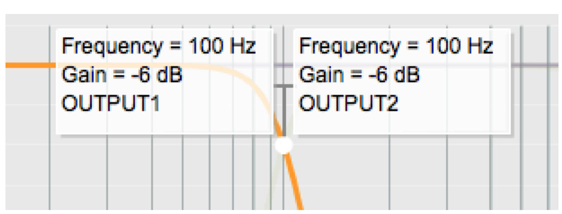
7.1.4 Crossover (Xover)

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 48.

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

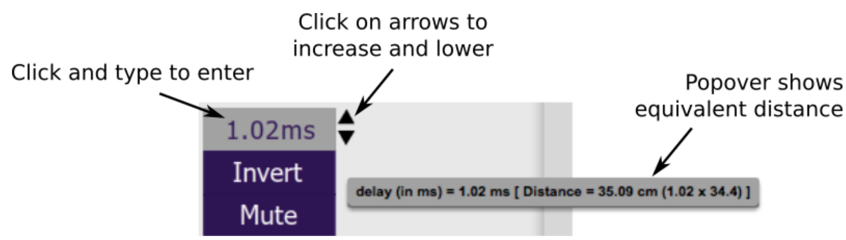
The **Bypass** button enables or disables a crossover filter. The filter is enabled if the button says "BYPASS" and disabled if the button says "BYPASSED" (see screenshot below).

Enabled Bypassed

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.

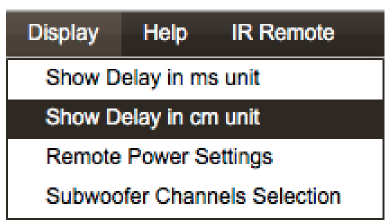
7.1.5 Time delay

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.

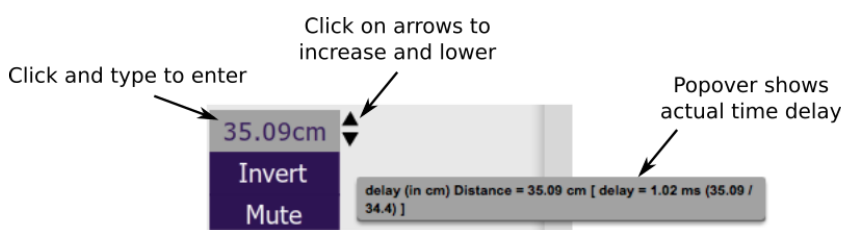


The time delay corresponds to a distance. This distance is shown in cm when the cursor is hovered over the time delay box. The maximum time delay of 20 ms corresponds to a distance of approximately 6.9 meters (about 22.5 feet).

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**.

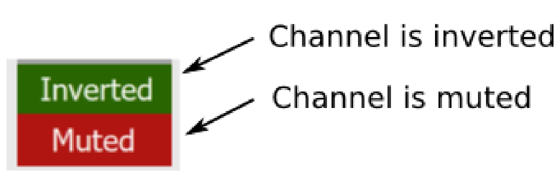


In this case, hovering the cursor over the entry box will display the actual time delay:



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

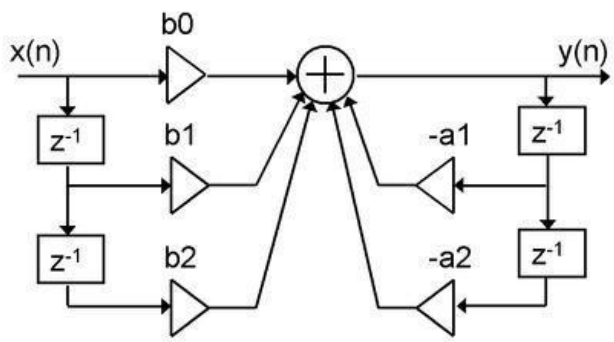
Custom biquad programming is available in the **PEQ** and **Xover** (crossover) blocks. Its purpose is to allow you to directly provide the low-level parameters aka *biquad coefficients* that control the digital filters of the processor, thus providing an almost infinite degree of flexibility.

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:

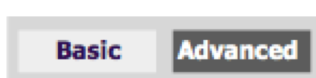
$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}$$



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

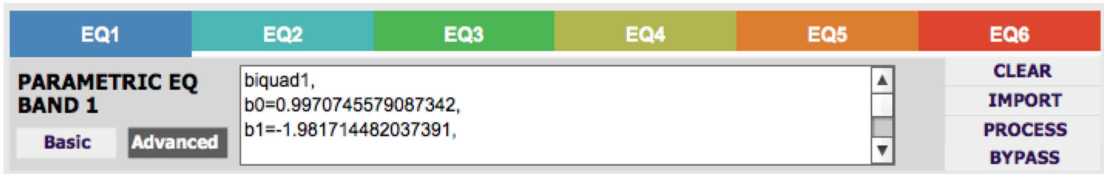
Each crossover block and PEQ filter has a selector that switches it to advanced mode:



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:

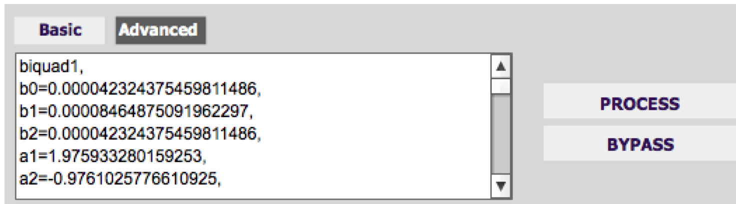
```
biquad1,
b0=0.998191200483864,
b1=-1.9950521500467384,
b2=0.996920046761057,
a1=1.9950521500467384,
a2=-0.9951112472449212,
biquad2,
b0=0.999640139948623,
b1=-1.9981670485581222,
...
biquad3,
...
biquad4,
...
biquad10,
b0=1.0010192374642126,
b1=-1.9950555192569264,
b2=0.9940580112181501,
a1=1.995060938714333,
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 Auto EQ with Room EQ Wizard. 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 C-DSP 8x12 DL 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.

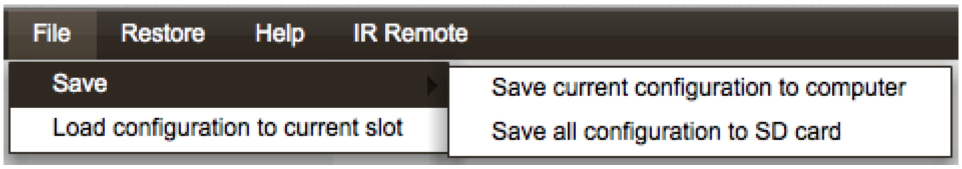
The plugin is in offline mode

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.

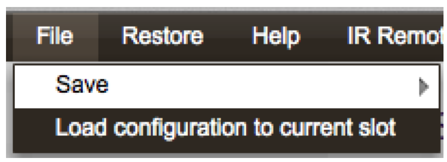
7.3.3 Saving and loading configurations

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.



To load a configuration from a saved file, first select the configuration preset that you wish to load into ("Config 1", "Config 2" etc). Drop down the **File** menu and select **Load configuration to current slot**.



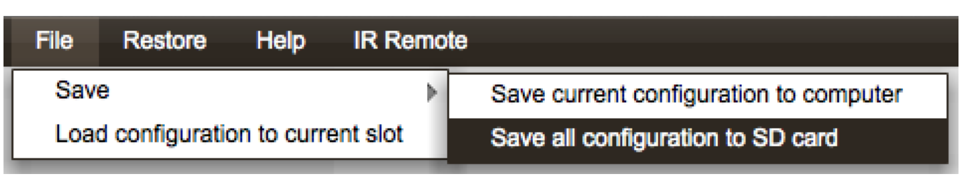
If the plugin is in online mode, the new configuration data will be downloaded to the processor immediately. If the plugin is in offline mode, the configuration will be loaded into the user interface only and will be downloaded to the processor the next time it is synchronized.

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.

7.3.4 Loading configurations from microSD card

The **C-DSP 8x12 DL** can load a set of configurations from a microSD card. This can be used to load new configuration sets without requiring a computer connection.

1. Connect a microSD card to the computer. If the computer has an SD card slot, you can use a microSD card adapter. Or, use an external card reader connected via USB. If the card hasn't previously been formatted, format in FAT format.
2. Start the plugin. Don't connect or synchronize to the C-DSP 8x12 DL. From the **File** menu, select **Save** and then **Save all configuration to SD card**.



3. From the file save dialog box that opens, type in a file name, navigate to the SD card and click on Save. This saves all four configurations to a special file on the SD card.
4. Eject the (micro) SD card from the computer.
5. Power off the **C-DSP 8x12 DL** if it is on. Insert the microSD card into the slot on the C-DSP 8x12 DL rear panel. (It pushes in and latches.) Power on the **C-DSP 8x12 DL**.
6. The C-DSP will load all four configurations. Each button on the wired remote will light as that configuration is loaded.
7. Eject the microSD card. (Push it in gently to unlatch it, then pull it out.) The C-DSP 8x12 DL is ready for use with a new configuration set.

7.3.5 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

Computer connectivity	Driverless USB 2.0 control interface for Windows and Mac
Analog audio inputs	• 6 x high-level differential on terminal Block, max 8/12 VRMS (switch-selectable) • 6 x low-level differential on RCA connectors, max 2/4 VRMS (switch-selectable)
Input impedance	Low-level inputs: 10 kΩ (4V setting) or 5 kΩ (2V setting) High-level inputs: 68 Ω
Digital audio Input	• S/PDIF on RCA connector / Isolated with digital audio transformer • TOSLINK optical A high quality onboard Asynchronous Sample Rate Converter ensures compatibility with most input sample rates, from 44.1 up to 192 kHz.
Analog audio outputs	12 x unbalanced on RCA connectors, 6.0V RMS full scale output
Output impedance	560 Ω
Audio resolution	24-bit input and output, 48 kHz internal sample rate
Audio processing	400 MHz 32-bit Floating Point SHARC Digital Signal Processor, ADSP21489
Filtering capabilities	Dirac Live mixed-phase filtering, calibration with the DiracLive application. User-programmable IIR filters: high pass and low pass crossover filters up to 48 dB/octave per output channel; ten biquad filters (parametric) EQ per output channel – peaking, low-shelf, and high-shelf types. Advanced biquad capability.
Storage/presets	All settings controllable in real time from software user interface. Up to 4 presets stored in local flash memory.
Wired external remote	External remote for control of active preset, master volume, subwoofer volume and master mute. LED indication of active preset.
microSD Card	Allows setup without a laptop: a configuration can be built offline and loaded automatically to the unit in the car via a microSD Card.
Infrared remote	“Learning remote” (NEC, Philips, Sony) controls master volume, mute, preset selection.
Remote trigger	REMOTE IN: 4 V DC trigger level REMOTE OUT: 12 V DC @ 100mA max current out
Power supply	Isolated DC-DC conversion, 10 to 14 V DC compliant for car audio environment
Dimensions (H x W x D)	47 x 205 x 122 mm

8.2 INPUT SENSITIVITY SETTING

The sensitivity of all inputs can be changed with a set of DIP switches internal to the unit. All inputs can have the sensitivity set independently. To change input sensitivity settings, remove the top cover. Referring to Figure 14 below, move **both** switches for each input to the desired setting.

The switches are very small, so you will need a small device like the end of a paperclip to move them. Don't use force, just push the switch in the desired direction and it will "click" into the new position.

Table 1. Input sensitivity switch settings

Input type	Low level inputs		High level inputs	
Switch setting	Off (default)	On	Off	On (default)
Full scale input voltage (RMS)	4V	2V	12V	8V
Input impedance	10 kΩ	5 kΩ	68 Ω	68 Ω

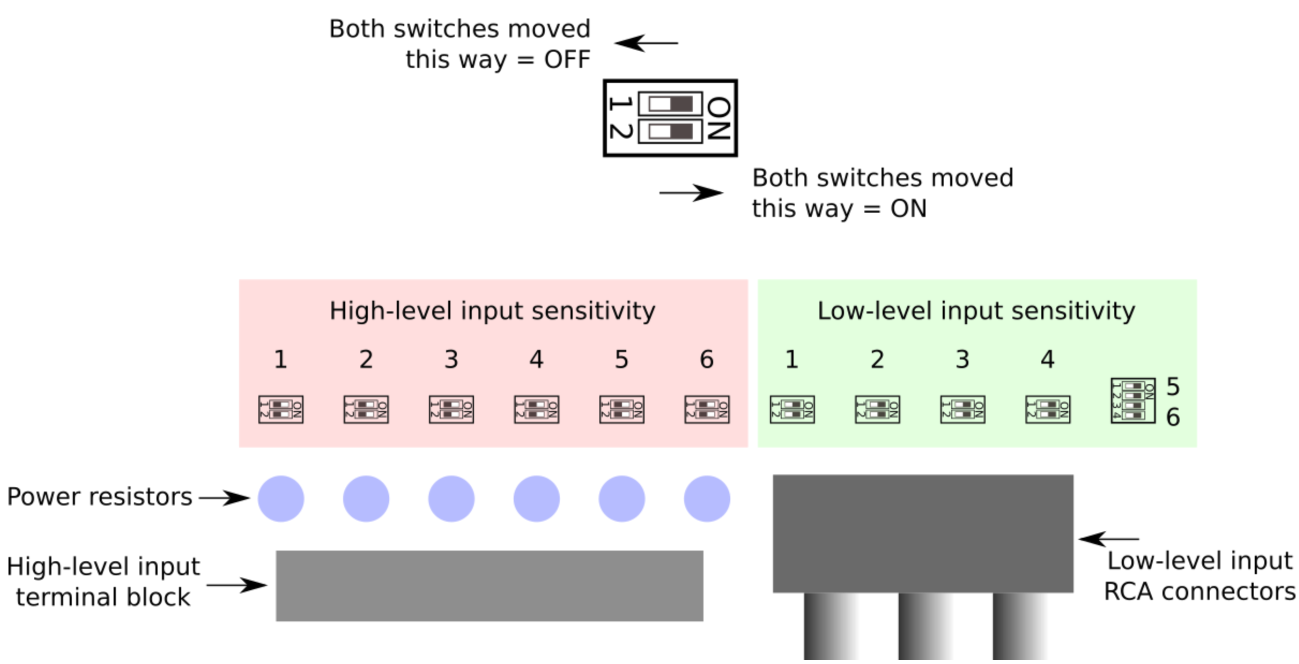
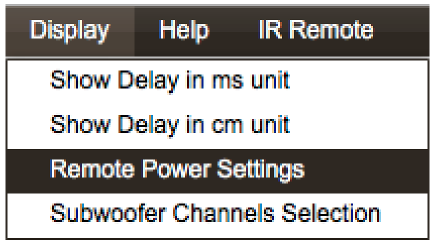


Figure 14. Input sensitivity switch locations

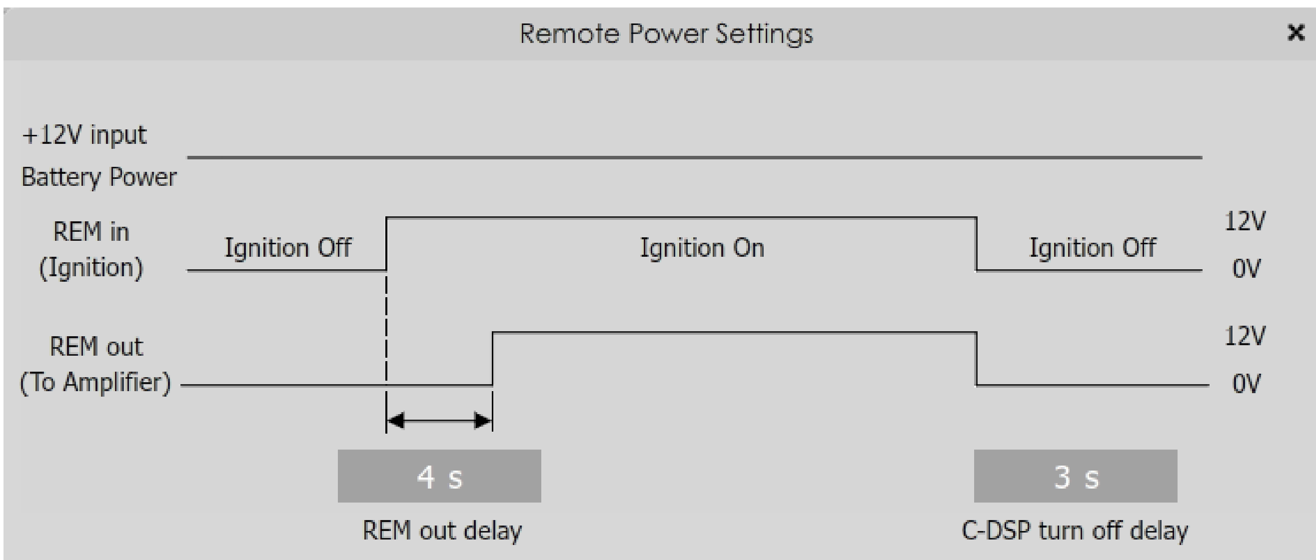
8.3 REMOTE TRIGGER TIMING

When the REMOTE MODE switch (see page 17) is set to position 2, the REMOTE OUT terminal can be used to enable and disable power amplification.

The relevant delays can be configured from within the plugin. To do so, open the C-DSP 8x12 DL plugin, drop down the **Display** menu and select **Remote Power Settings**:



The window shows a timing diagram with two editable text boxes that can be used to change the timing:



REM out delay

This value controls the delay from REMOTE IN going high to REMOTE OUT going high.

C-DSP turn off delay

This value controls the delay from REMOTE IN going low until the C-DSP 8x12 DL turns off.

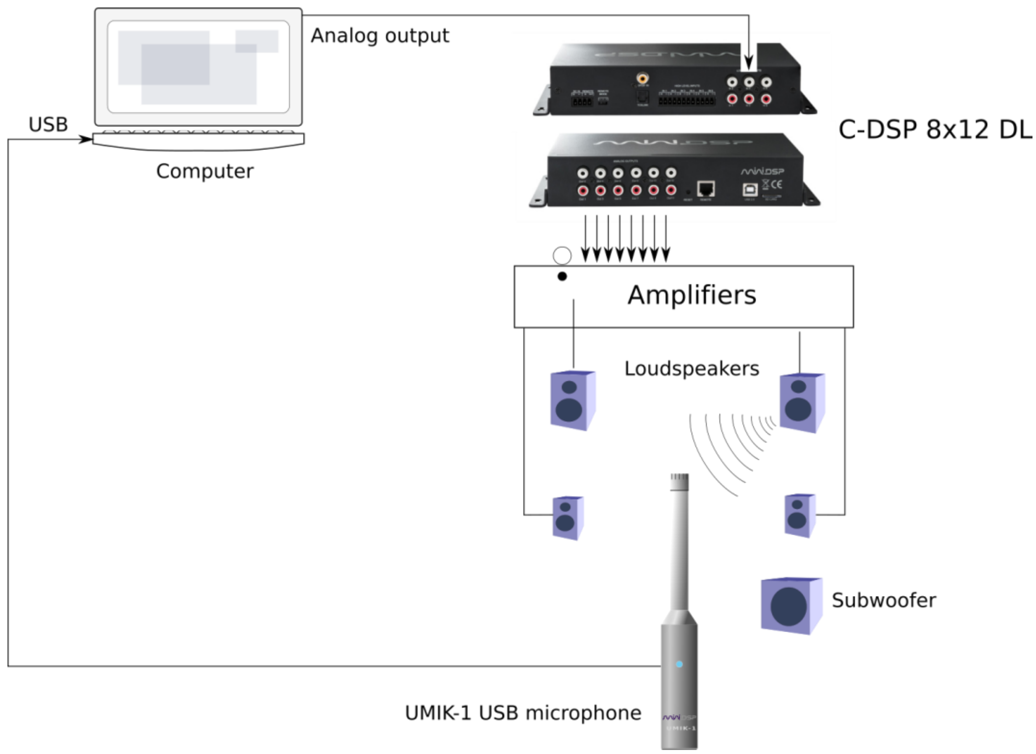
Enter the desired value for each time delay parameter and press the Enter key. When finished, click the close "X" icon at the top right, and the settings will be downloaded to the C-DSP 8x12 DL.

Note: the default settings are suitable for 90% of installations. There's no need to change from the default if it's working fine for you.

8.4 ACOUSTIC MEASUREMENT SETUP FOR REW

To obtain most accurate setup for active speakers and subwoofer crossovers, acoustic measurement will be required. You can use any program you like, but the freeware REW (Room EQ Wizard) is a good choice.

This diagram illustrates the connection. An analog output from the computer is connected to the relevant input of the C-DSP 8x12 DL. The output can a headphone or line output, or via a USB soundcard. The UMIK-1 is configured as the inputs source in REW, so that the output corresponding to the connected input can be measured.



REW generates a test signal that plays through the system and each speaker in turn. This program uses a swept-sine wave technique, which is a signal that starts at low frequencies and gets higher in frequency over the course of a few seconds. After playing the signal and recording the resulting audio, REW analyzes the captured signal and displays a frequency response plot as shown above. It is also capable of many other types of analysis that you can use to optimize your audio or home theater system.

To set up your computer to use Room EQ Wizard and a UMIK-1, use one of the following application notes:

- Using the UMIK-1 and REW with HDMI output - Windows
- Using the UMIK-1 and REW with HDMI output - Mac

To learn more about how to analyse and understand the results of your measurements, see:

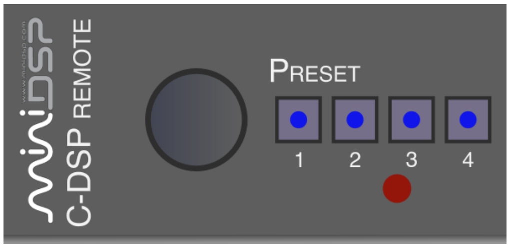
- Acoustic measurement with the UMIK-1 and REW

8.5 TROUBLESHOOTING

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

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 a plugin. 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. Check that the USB cable to the processor is firmly connected. b. Reset the processor by power-cycling the unit.
4	No signal showing on input meters in C-DSP 8x12 DL plugin	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.
5	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 and output meters are showing signal. d. Check that master mute is not enabled. e. Check the master volume level. f. Check that your crossover frequencies are correct e.g. that you don't have high pass and low pass frequencies incorrectly set. g. Check that the matrix mixer is sending the correct inputs to the correct outputs.
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. 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.
8	Cannot reload a configuration	a. Confirm the file format of your file (.xml). b. Confirm the version of the file.

8.6 4-BUTTON WIRED REMOTE (LEGACY)



Note: The 4-button wired remote shipped with C-DSP 8x12 DL units up until December 2020. From January 2021, all units ship with the new OLED wired remote.

The 4-button wired remote and/or an infrared remote can be used to control:

- Master volume
- Subwoofer volume
- Master mute
- Preset selection
- Volume mode selection

Status indicators

The currently selected preset is indicated by a blue LED in one of the selection buttons.

Operation of the 4-button wired remote

To change the volume

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

To mute and unmute

Press on the control knob.

To change the preset

Press one of the four buttons. The LED of the selected preset will flash several times. When it stays lit, the selected preset is now in operation.

To change the volume control mode

Press buttons 1 and 2. LEDs 1 and 2 will light either solid or blinking, depending on the current mode:

- LEDs 1 and 2 are solid (not blinking): **Master Volume** mode
- LEDs 1 and 2 are blinking: **Subwoofer Volume** mode

Press button 4 to change the mode. Press buttons 1 and 2 to return to normal operation.

To quickly adjust subwoofer volume while in Master Volume mode:

1. Press buttons 1 and 2, then button 4 to switch to Subwoofer Volume mode.
2. Adjust the rotary control until the subwoofer level sounds right.
3. Press button 4 to switch back to Master Volume mode.
4. Press buttons 1 and 2 to return to normal operation (for preset selection).

8.7 FIRMWARE UPGRADE – MAIN BOARD AND WIRED REMOTE

See "Upgrade instructions for C-DSP2.txt" in the **firmware** folder of the software download.

8.8 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 support.minidsp.com with:
    a. The specific product you are having an issue with (in this case, C-DSP 8x12 DL).
    b. A clear explanation of the symptoms you are seeing.
    c. A description of the troubleshooting steps performed and your results.