C-DSP 8x12 DL

8-IN 12-OUT ADVANCED IN-CAR AUDIO PROCESSOR

WITH DIRAC LIVE® ACOUSTIC CORRECTION

User Manual – Legacy Support Version

Supported OS

Windows Vista® SP1/Win7/Win8
OS X 10.9 – 10.11, macOS 10.12 – 10.13

NOTICE

This is the Legacy Support version of this manual.

It describes use of the original Dirac Live ("Dirac Live 1") with the C-DSP 8x12 DL processor. miniDSP continues to support this version for the benefit of customers unable to run the latest versions of Windows and macOS. It is for:

- Windows Vista® SP1/Win7/Win8
- OS X 10.9 – 10.11, macOS 10.12 – 10.13

This manual will not be updated after June 2020. Please refer to the current C-DSP 8x12 DL User Manual for any updates to product functionality.

Revision history

Revision	Description	Date
V0.1	Preliminary version for Dirac Live	3 April 2019
V0.4	Preliminary public release	20 May 2019
V1.0	Public release	23 August 2019
V1.1	Marked as Legacy Support ("Dirac Live 1")	8 June 2020
V1.1a	Updated Legacy support download path	13 July 2020

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

- Intel Pentium III or later, AMD Athlon XP or later
- 2 Gigabytes (GB) of RAM or higher
- Keyboard and mouse or compatible pointing device
- Microsoft• ® Windows® Vista® SP1/Win7/Win8/Win10
- Two free USB 2.0 ports

Mac OS X

- 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
- OS X 10.9 (Mavericks) to macOS 10.13 (High Sierra)
- 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

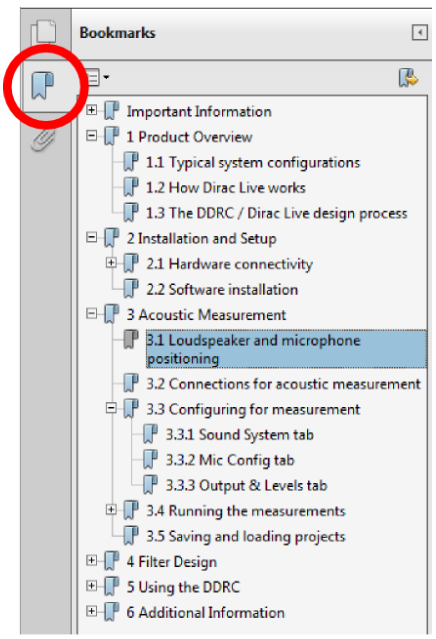
The C-DSP 8x12 DL has passed the test performed according to European Standard EN 55022 Class B.

A Note on this Manual

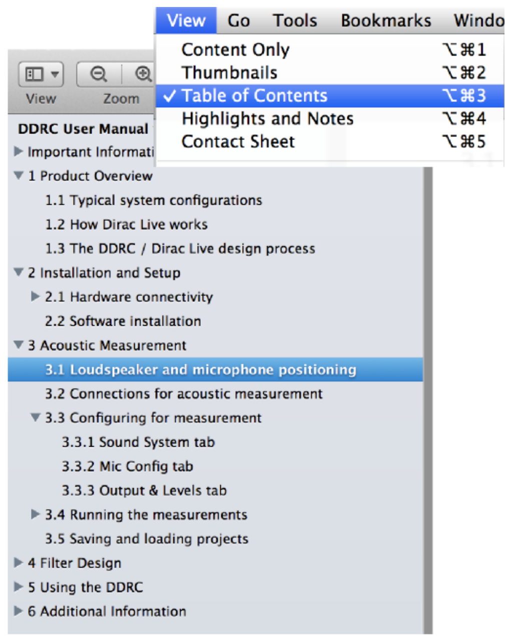
This User Manual is designed for reading in both print and on the computer. If printing the manual, please print double-sided. The embedded page size is 8 ½" x 11". Printing on A4 paper will result in a slightly reduced size.

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. Displaying this table of contents will make navigation much easier:

- In Adobe Reader on Windows, click on the "bookmarks" icon at the left. The table of contents will appear on the left and can be unfolded at each level by clicking on the "+" icons.
- In Preview on the Mac, click on the **View** menu and select **Table of Contents**. The table of contents will appear on the left and can be unfolded at each level by clicking on the triangle icons.



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.[1]

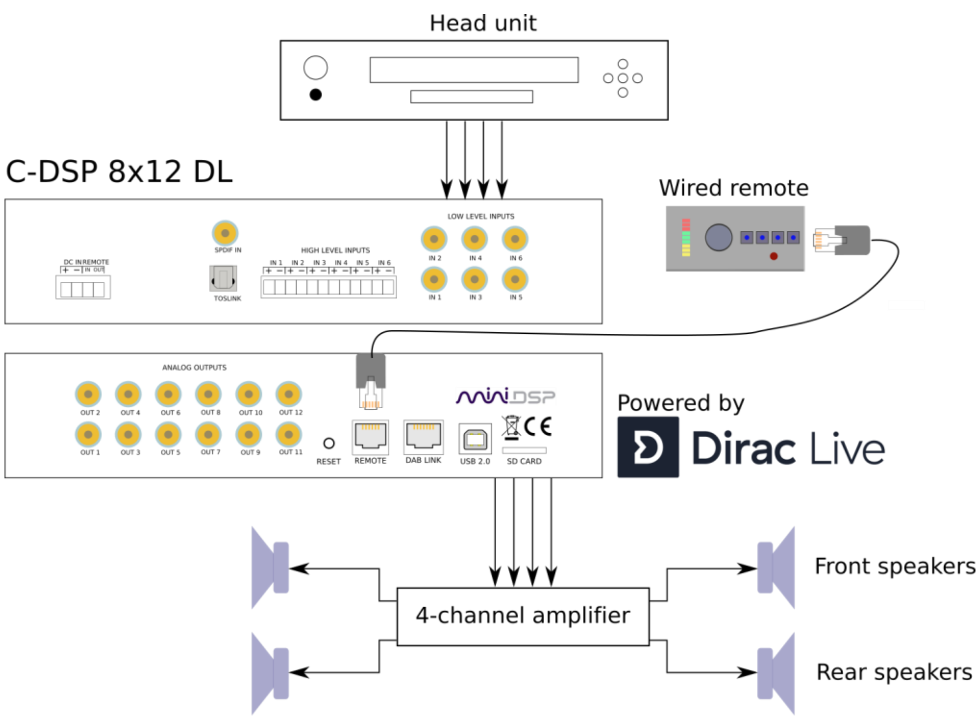


Figure 1. Basic system setup with four-channel acoustic correction

[1] Note that the miniDSP version of Dirac Live Calibration Tool **requires** a miniDSP UMIK-1 Other microphones cannot be used.

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. While designed specifically for car use, the advanced features and high performance of the C-DSP 8x12 DL also make its use feasible in other situations where complex multichannel processing is needed, such as homes, recording studios, churches and halls, and exhibitions.

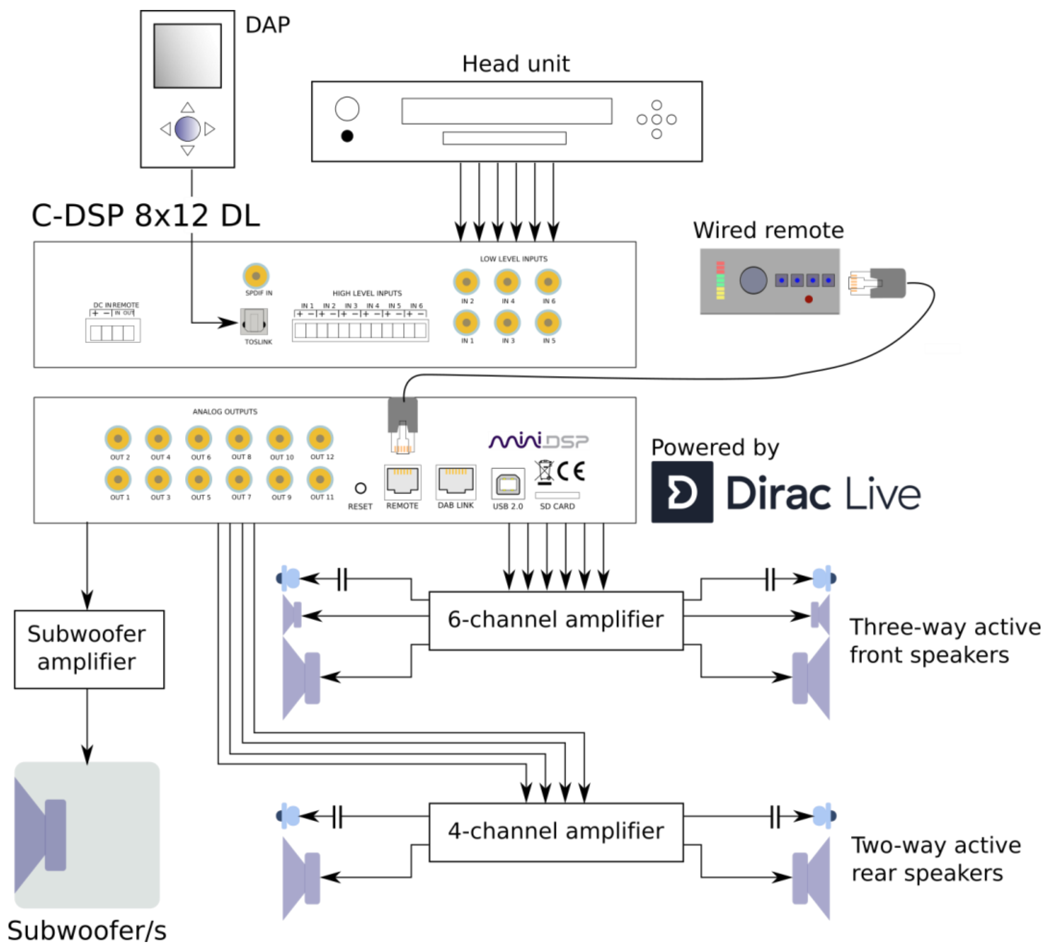
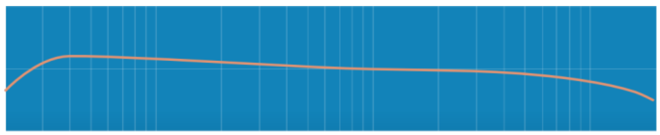


Figure 2. Advanced system configuration with multiple sources and active speakers

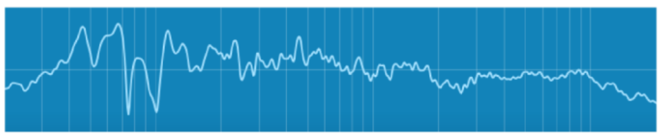
1.2 DIRAC LIVE

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.

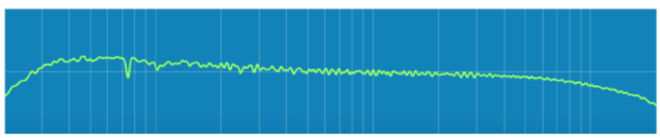
To accomplish its remarkable improvement in listening quality, the **Dirac Live Calibration Tool** (DLCT) steps you through the procedure for taking measurements around your listening area. Dirac Live® employs a sophisticated analysis algorithm to make the optimal correction across the *whole* listening 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

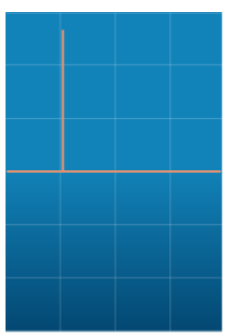


Uncorrected magnitude response

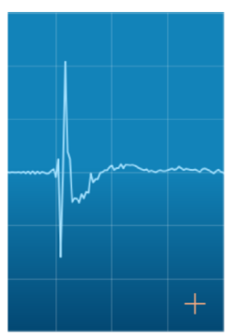


Corrected magnitude response

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

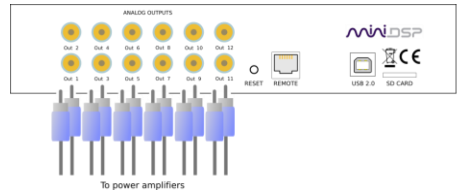
Corrected
impulse response

The C-DSP 8x12 DL provides, in addition to Dirac Live, a full suite of miniDSP's audio processing functions on each channel. Combined with two flexible matrix mixers for audio routing, this enables advanced applications such as in-vehicle active crossovers to be corrected by Dirac Live, all in a compact 12V-powered unit.

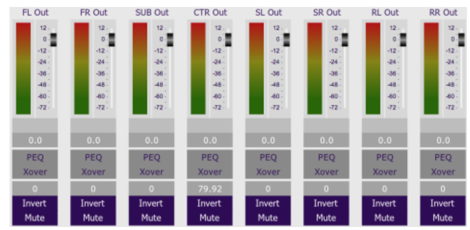
1.3 Overview of Configuration Steps

The steps for configuring the C-DSP 8x12 DL with Dirac Live® are summarized as follows:

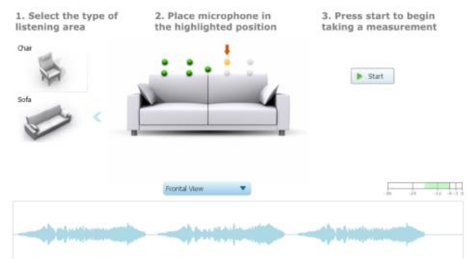
1. Connect the *C-DSP 8X12 DL* audio processor into your system and install software. See Section 2, Hardware Connectivity and Section 3, Software Installation.



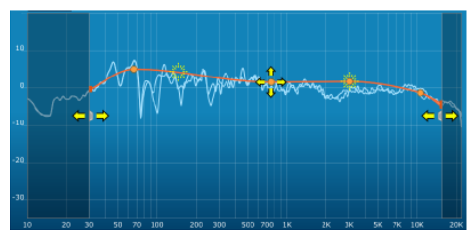
2. Configure output channel processing with the C-DSP 8x12 DL plugin. This sets up individual control of each output channel in order to implement (for example) subwoofer crossover, active speaker crossovers, or rear/center synthesis. See Section 4, Plugin Overview.



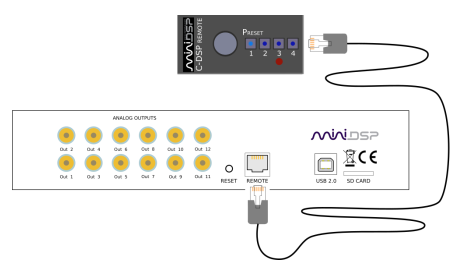
3. Run a series of acoustic measurements using the **Dirac Live Calibration Tool** program, to capture the acoustic behavior of your speakers and acoustic environment. See Section 5, Acoustic Measurement for Dirac Live.



4. Generate digital correction filters that will be executed by the C-DSP 8x12 DL. Up to four filter sets can be downloaded into the processor for easy real-time recall and auditioning. See Section 6, Dirac Live Filter Design.



5. Once configuration is complete, the computer is not needed. See Section 7, Remote Control.



2 SOFTWARE INSTALLATION

The C-DSP 8x12 DL is configured by software running on a PC or Mac.

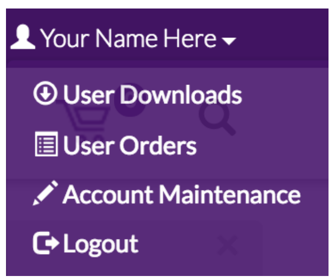
2.1 DOWNLOAD THE SOFTWARE

If you purchased your product directly from miniDSP, your software will be available from the User Downloads section of the miniDSP website when your order ships.

If you purchased your product from a miniDSP dealer, you will receive a coupon together with the product. Redeem this coupon and select the Plugin Group "miniDSP C-DSP 8x12" at the link below:

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

To access the download, you will need to be logged into the miniDSP.com website with the account you created when purchasing. The User Downloads link is visible from the dropdown menu at the top right of the website page:



Navigate to the **C-DSP plug-ins** section and download the zip file under the heading **CDSP-8x12DL Dirac 1.0.**

Unzip the downloaded file: on Windows, right-click and select "Extract All..."; on Mac, double-click.). The unzipped download has a name like **C_DSP_8x12_DL_v1_7_mbfw_v2_7** and will contain the following folders:

Dirac Live

This folder contains the installers for **Dirac Live Calibration Tool** (DLCT), which is used to perform the Dirac Live calibration, including taking measurements, generating correction filters, and loading them into the processor. There are separate Windows and Mac versions.

Plugins

This folder contains the installers for the **C-DSP 8x12 DL** plugin, used to set up non-Dirac signal processing, configure remote control codes and perform various other maintenance operations on the processor. There are separate Windows and Mac versions.

firmware

This folder contains the firmware for the processor. See Firmware Upgrade on page 78.

2.2 Software installation — Windows

Possible Windows installation issues

The miniDSP software requires that a number of other frameworks be installed for it to work. For Windows 7 and later, these packages should be installed automatically. For earlier versions of Windows, please download and install the following frameworks before attempting to install any miniDSP software. You can also manually install these if you receive an error message that required software is missing.

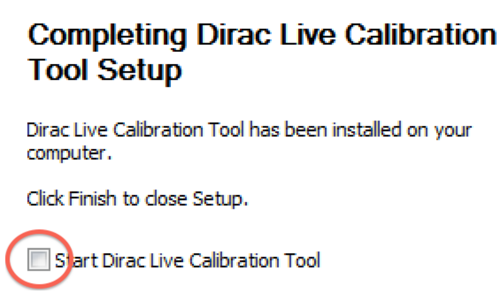
- Microsoft .NET framework (version 3.5 or later)
- Latest version of Adobe Air
- Microsoft Visual C++ 2010 Redistributable Package: for x86 (32-bit operating system) or x64 (64-bit operating system).

C-DSP 8x12 DL plugin installation

1. Navigate to the **Plugins** folder of the software download and then to the **Windows** folder.
2. Double-click on the **C_DSP_8x12_DL.exe** installer program to run it. We recommend that you accept the default installation settings.

Dirac Live Calibration Tool (DLCT) installation

1. Navigate to the **Dirac Live** folder of the software download and then to the **Windows** folder.
2. Double-click on the installer to run it. The installer will have a name similar to **Dirac Live Calibration Tool v1.2.41.8863 Setup.exe** (the version number starting with v1.2... may be different). We recommend that you accept the default installation settings. However, on the last screen, uncheck the box to start Dirac Live automatically (you will need to install the driver as described on the next page before using DLCT).



Note 1: The Adobe Air framework may need to connect to the Internet the first time you run the plugin.

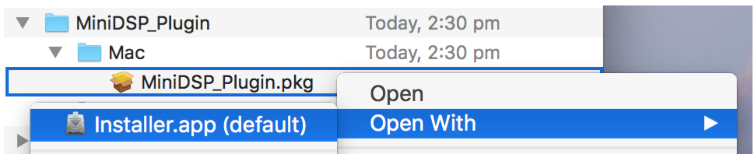
Note 2: The first time you run the plugin, you may see a warning from Windows Firewall asking whether the software should be allowed network access. If you do, ensure that “Private networks...” is checked and “Public networks...” is not checked. Then click on “Allow access.”

2.3 SOFTWARE INSTALLATION — MACOS / OS X

Possible Mac installation issues

If double-clicking on an installer brings up a message that the installer cannot run, use this alternate method (note that the name of the plugin will be **C-DSP-8x12-DL.pkg**, not **MiniDSP_Plugin.pkg** as shown in the example screenshots):

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)."



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

C-DSP 8x12 DL plugin installation

1. Navigate to the **Plugins** folder of the software download and then to the **Mac** folder.
2. The installer program 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 **C-DSP-8x12-DL.app** in the Applications -> miniDSP folder and double-click on it. To make it easier to run in future, right-click on its dock icon and select Options -> Keep in Dock.

Note: The Adobe Air framework may need to connect to the Internet the first time you run the plugin.

Dirac Live Calibration Tool (DLCT) installation

1. Navigate to the **Dirac Live** folder of the software download and then to the **Mac** folder.

2. The installer program will have a name similar to **Dirac Live Calibration Tool v1.2.41.8863.pkg** (the version number starting with v1.2... may be different). 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 DLCT, locate **Dirac Live Calibration Tool.app** 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.

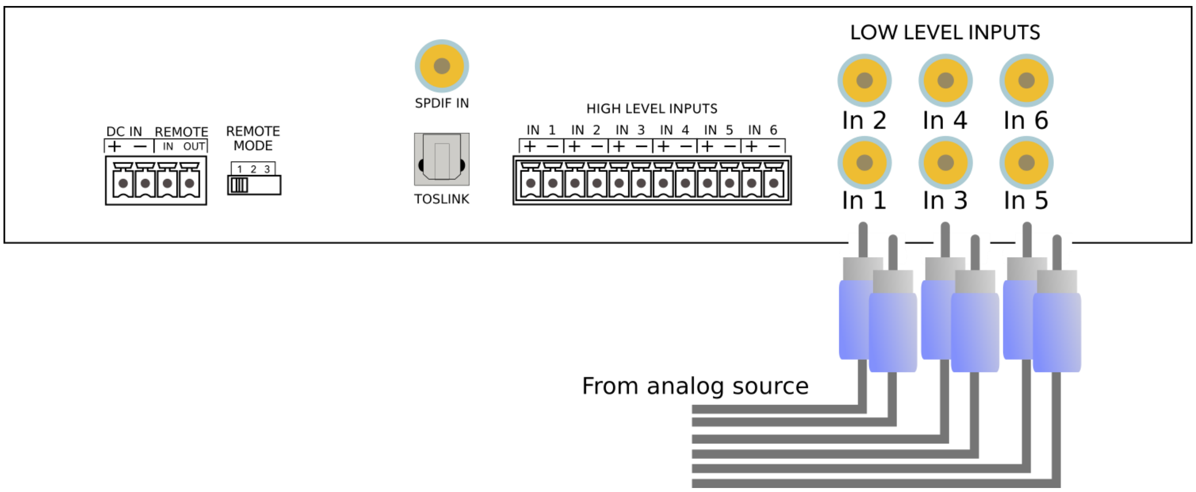
3 HARDWARE CONNECTIVITY

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 73).



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.

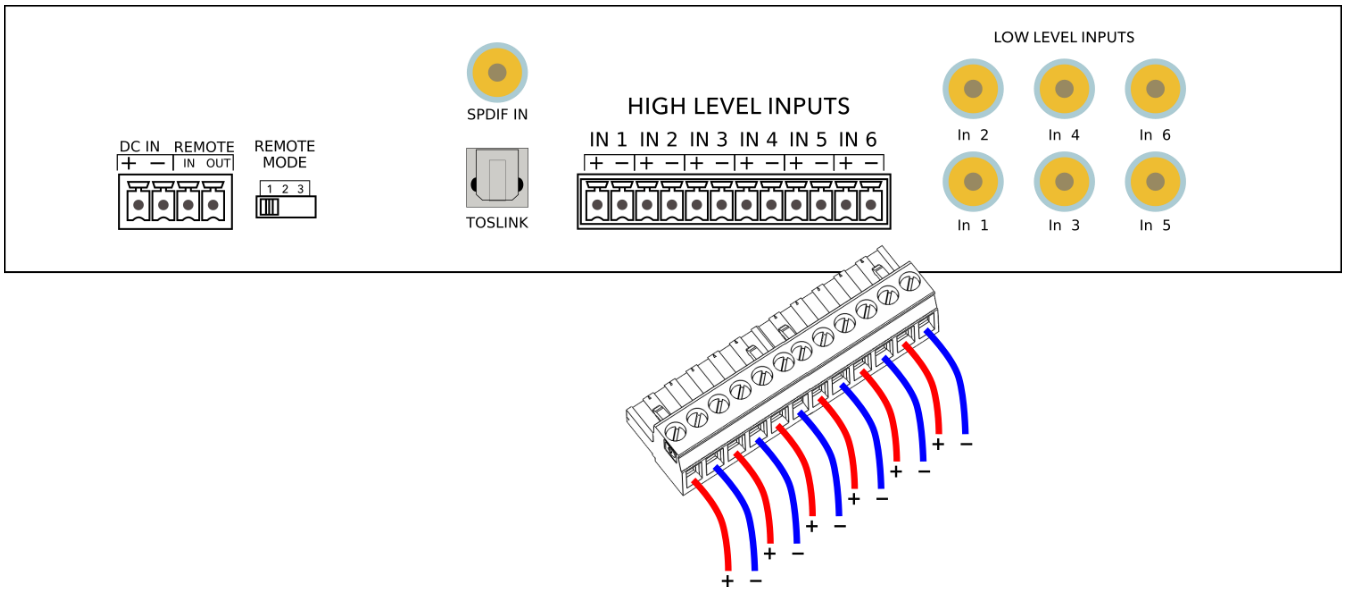
Figure 3. Single-ended RCA connection

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.



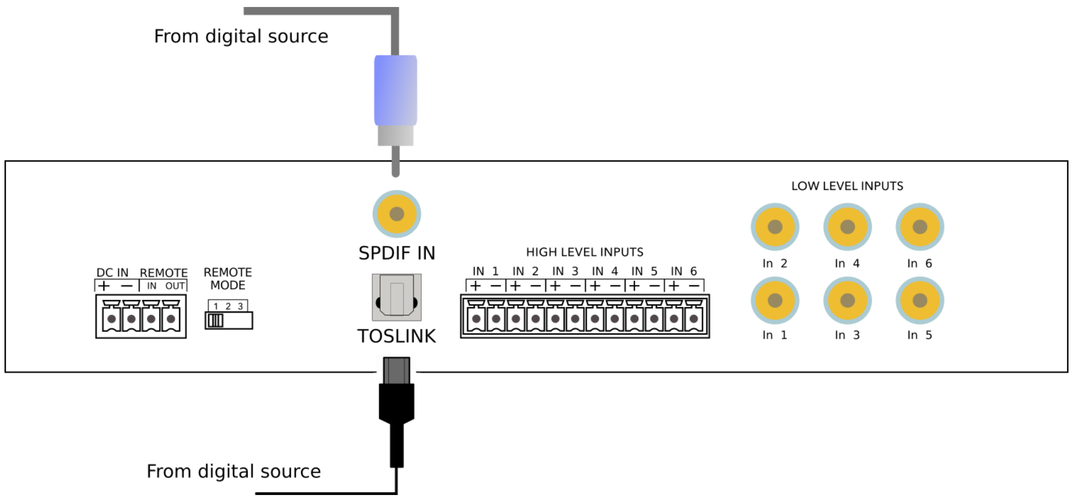
The high-level inputs have an input impedance of 68 Ω and are designed for connection to the speaker outputs of a head unit. The inputs are fully differential, therefore:

- 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 73). 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

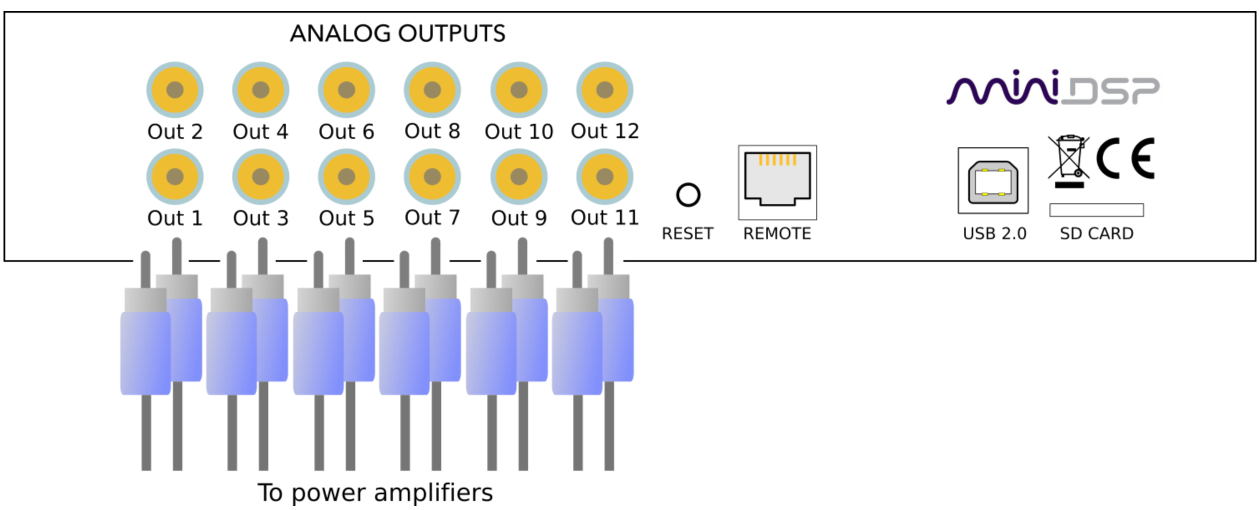
Two digital sources can be connected via the S/PDIF coax (RCA) connector and the TOSLINK (optical) connector. Switching between sources can be done from within the plugin or with an infrared remote control.



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.

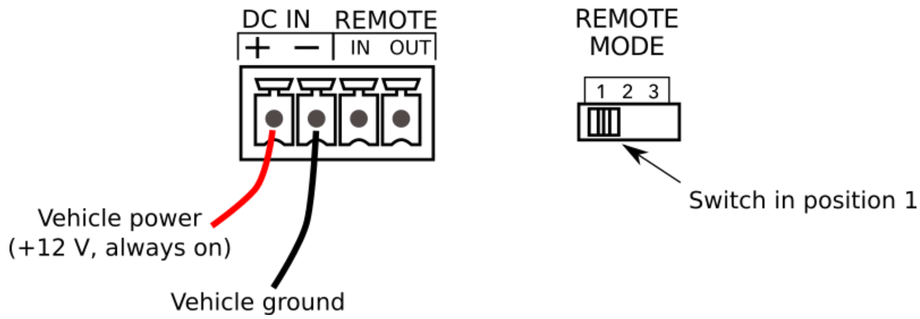


3.4 DC POWER

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.

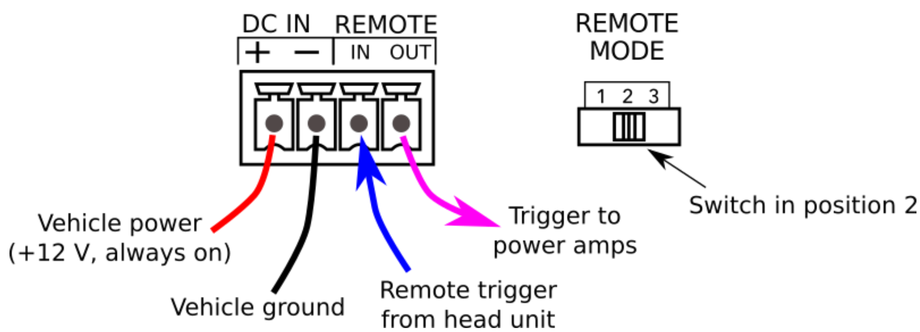
3.4.1 Powered on power (position 1)

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).



3.4.2 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.

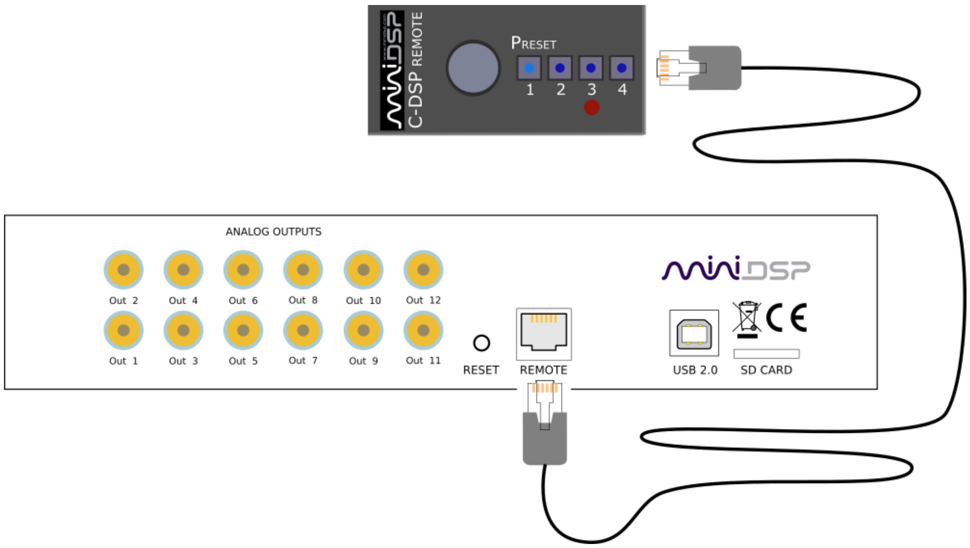


In this mode, the **REMOTE OUT** terminal should be used to turn on the power amplifier(s). There is a time delay between **REMOTE IN** going positive and **REMOTE OUT** going positive. This can be configured in the plugin – see page 74.

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

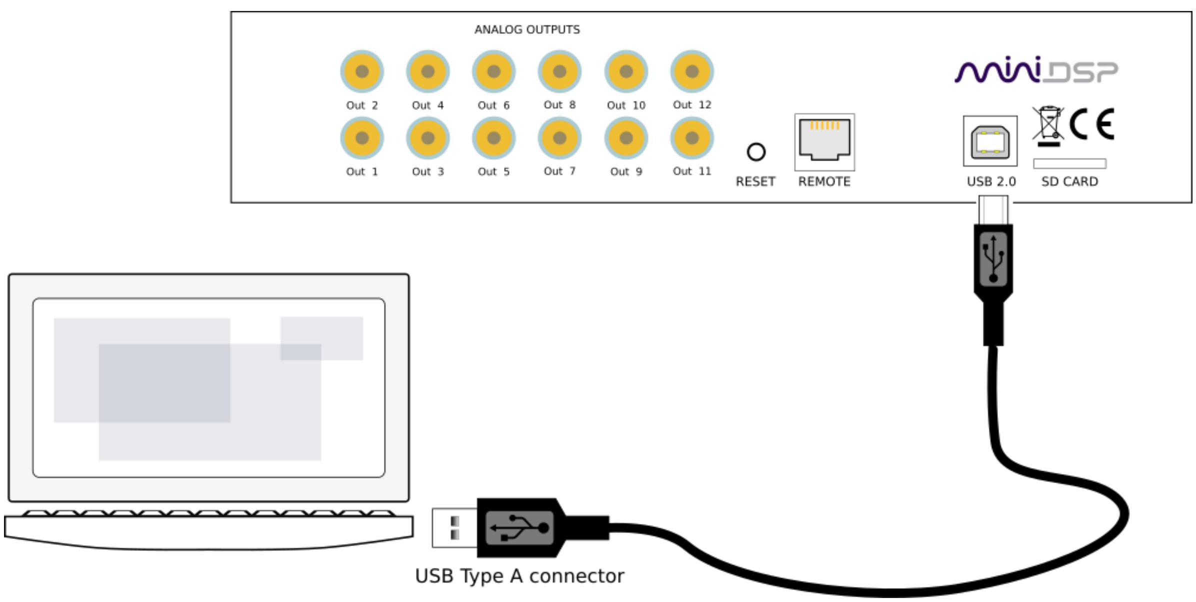
3.5 WIRED REMOTE

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.



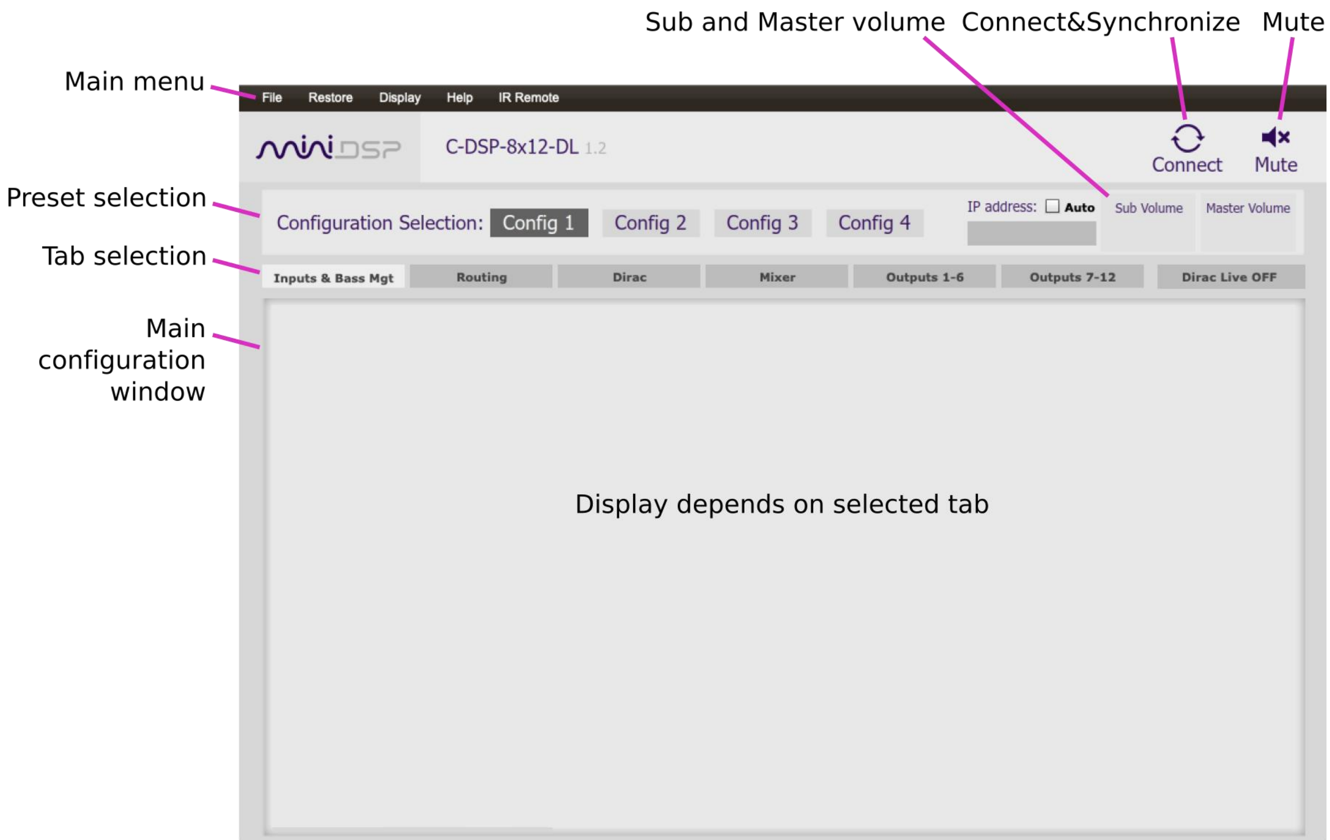
Configurations can also be loaded from a micro SD card – see page 70.

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:



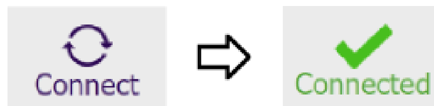
During *initial* configuration of the processor, it is strongly recommended that any connected amplification be muted or powered off.

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.

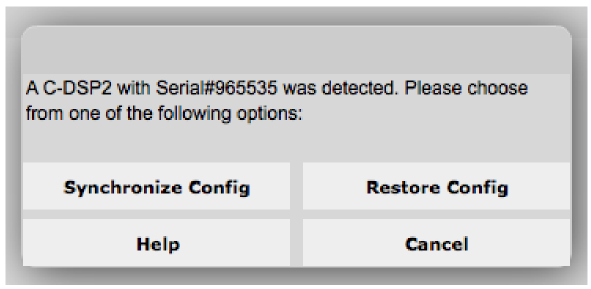
4.2 SYNCHRONIZING WITH THE PROCESSOR

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:

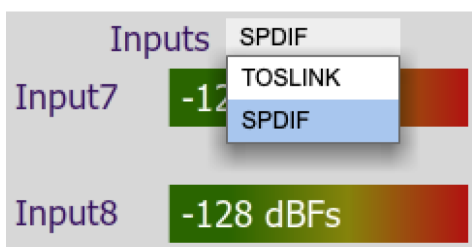
To switch to a different preset, just click on the desired button:

4.3.2 Tab selection

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

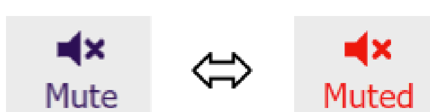
4.3.3 Digital input selection

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 57.

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 DLCT:

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 – DLCT 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 75.

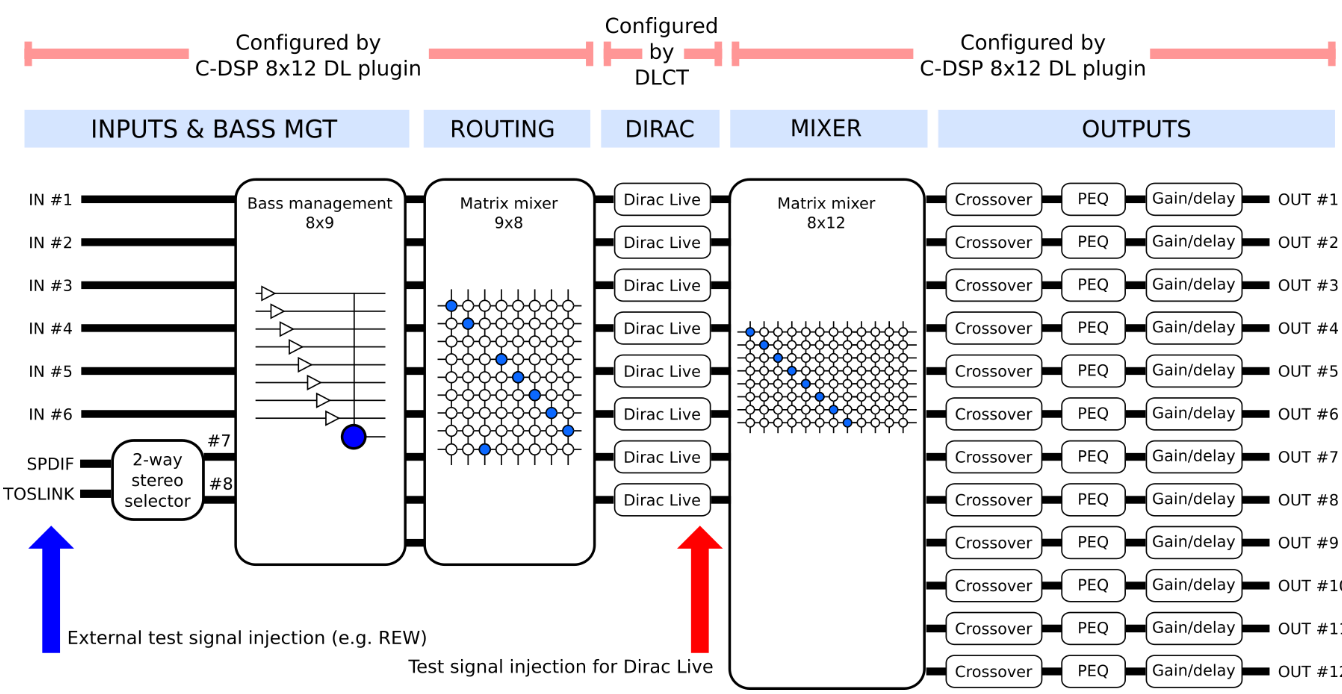
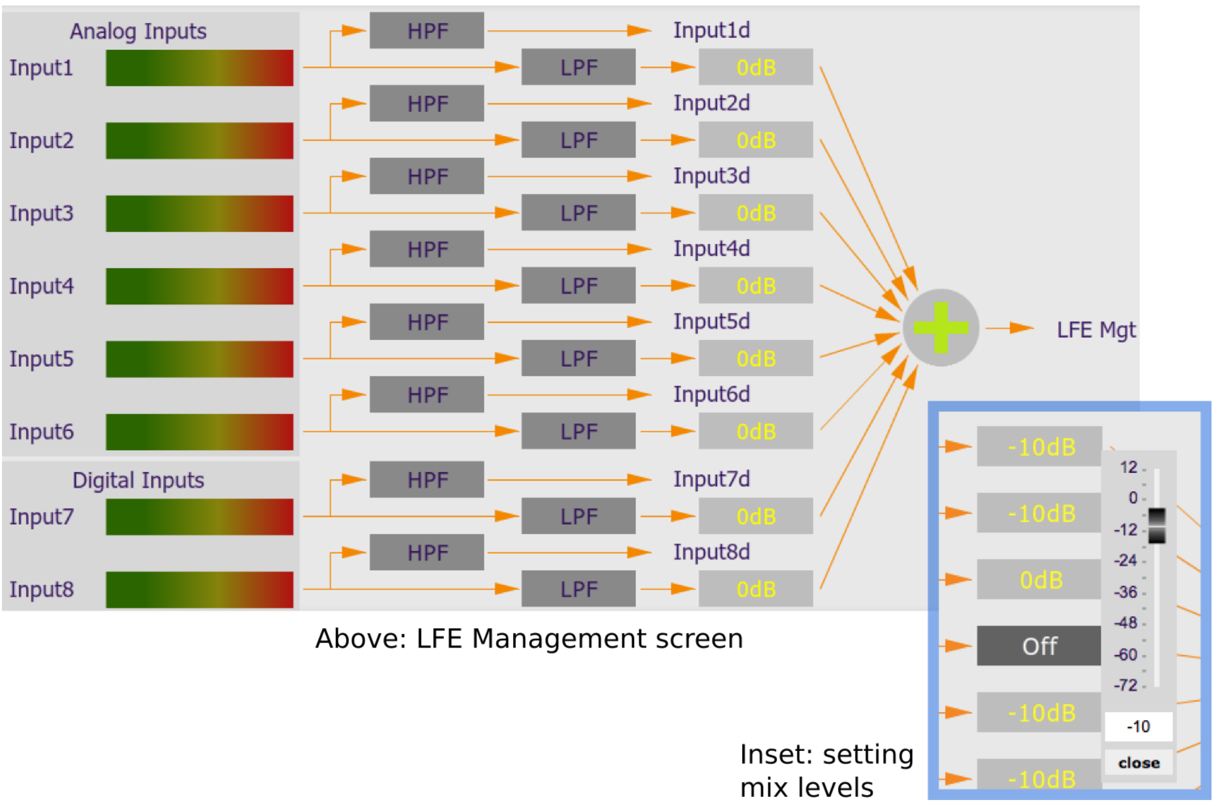


Figure 5. Signal flow through the C-DSP 8x12 DL

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 – see page 23.

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:

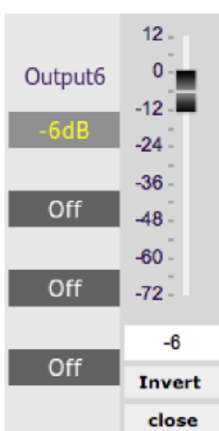
	Dirac 1	Dirac 2	Dirac 3	Dirac 4	Dirac 5	Dirac 6	Dirac 7	Dirac 8
Input1d	0dB	Off	Off	Off	Off	Off	Off	Off
Input2d	Off	0dB	Off	Off	Off	Off	Off	Off
Input3d	Off	Off	0dB	Off	Off	Off	Off	Off
Input4d	Off	Off	Off	0dB	Off	Off	Off	Off
Input5d	Off	Off	Off	Off	0dB	Off	Off	Off
Input6d	Off	Off	Off	Off	Off	0dB	Off	Off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

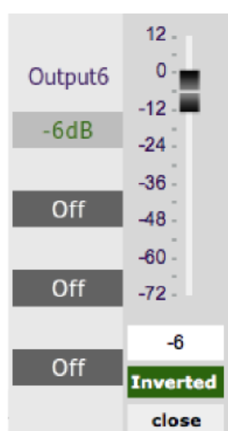
At each cross-point of the matrix, the input channel (labels along the left) is mixed into the corresponding output channel (labels along the top) if the lettering is highlighted in yellow. The cross-point is turned on and off by clicking on it. Any number of input channels can be mixed to each output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

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.

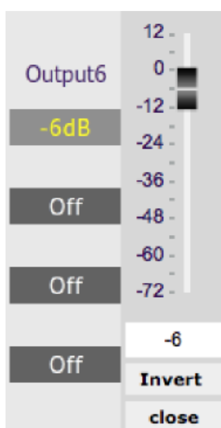
This is the default Mixer tab:

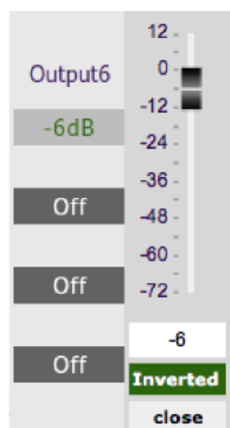
	Output1	Output2	Output3	Output4	Output5	Output6	Output7	Output8	Output9	Output10	Output11	Output12
Dirac1	0dB	Off	Off	Off	Off	Off	Off	Off	Off	Off	Off	Off
Dirac2	Off	0dB	Off	Off	Off	Off	Off	Off	Off	Off	Off	Off
Dirac3	Off	Off	0dB	Off	Off	Off	Off	Off	Off	Off	Off	Off
Dirac4	Off	Off	Off	0dB	Off	Off	Off	Off	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.

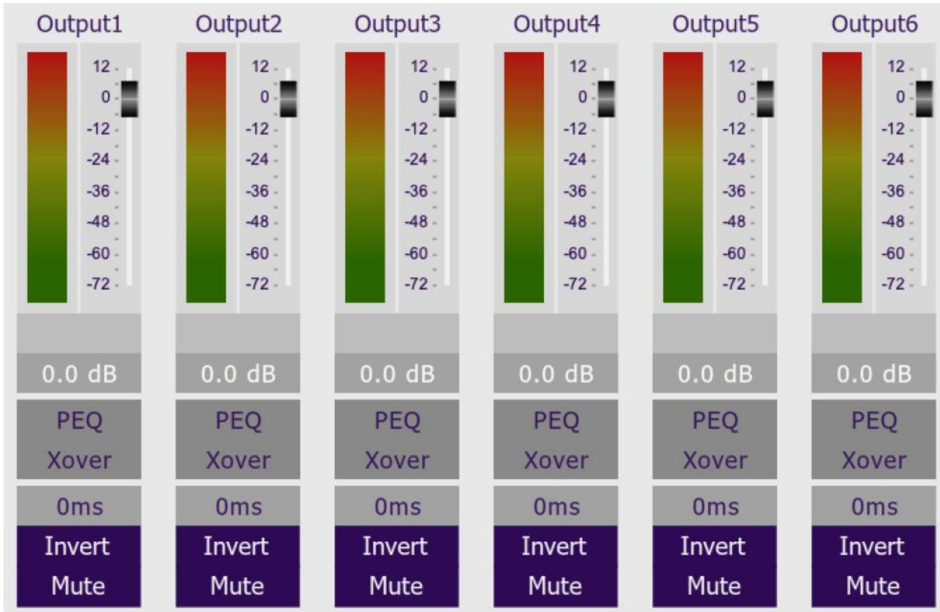
As with the Routing tab, each cross-point has adjustable level and an Invert option.



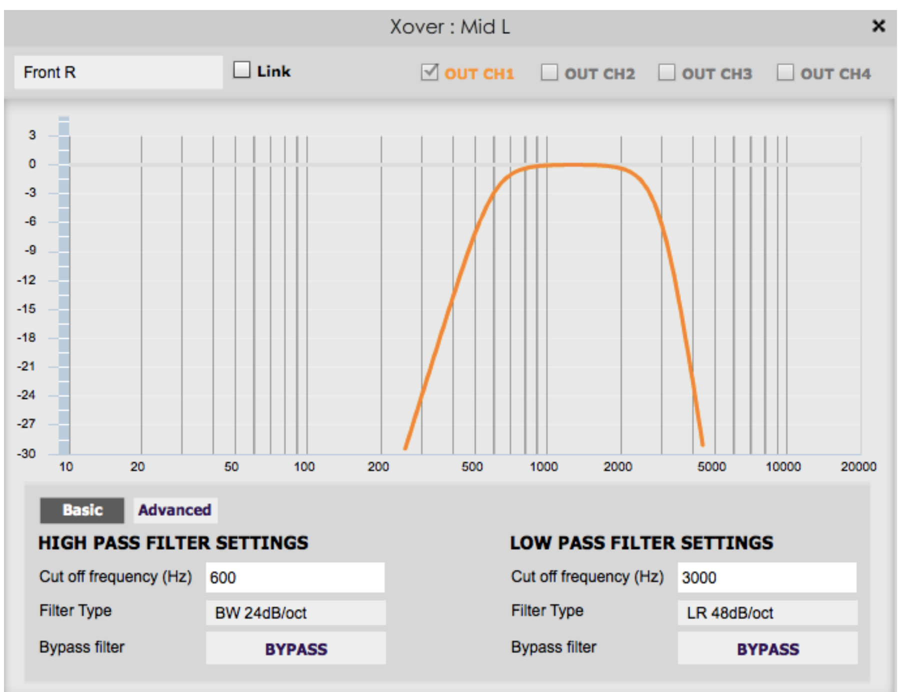


4.9 OUTPUT TABS

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 60 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. The only processing that takes place while playing audio is in Dirac Live. All input channels proceed straight to the Dirac Live block, and all output channels are taken from the Dirac Live 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.

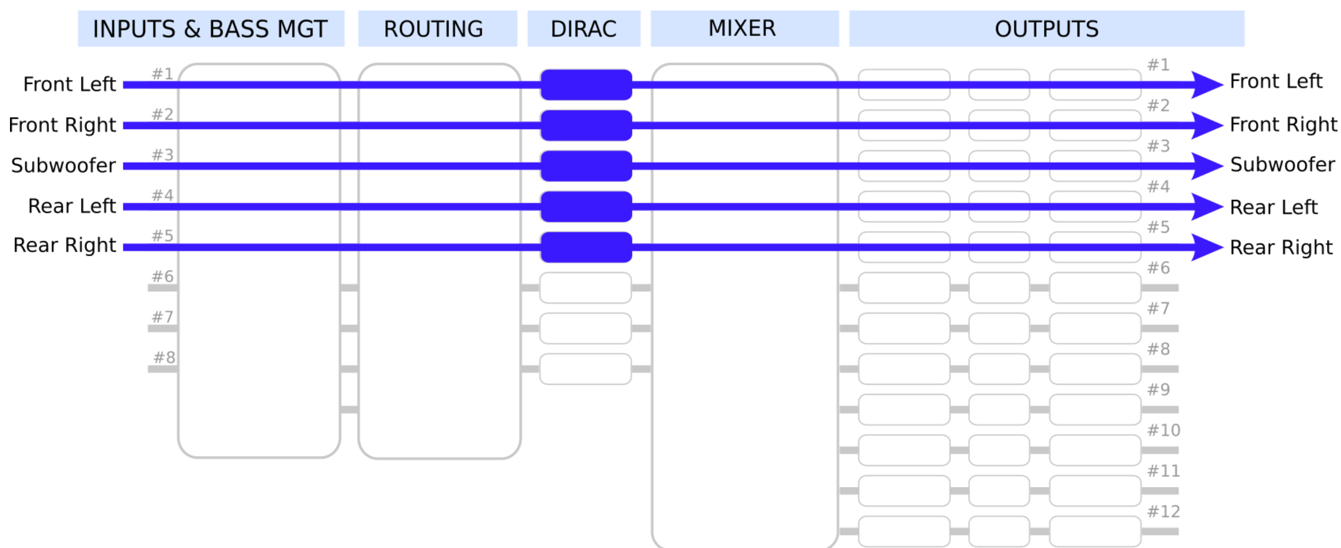


Figure 6. Straight through processing in the C-DSP 8x12 DL

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.

4.10.2 Bass management

You can synthesize a subwoofer signal from the speaker inputs. Figure 7 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.

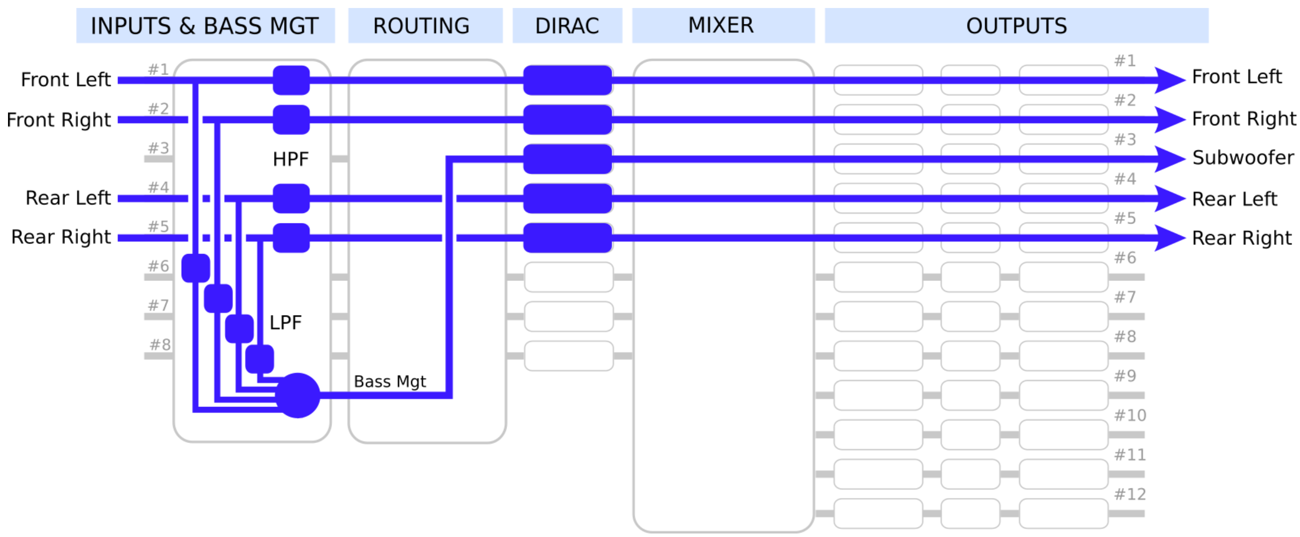


Figure 7. Bass management in the C-DSP 8x12 DL

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 7, 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 8. 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.

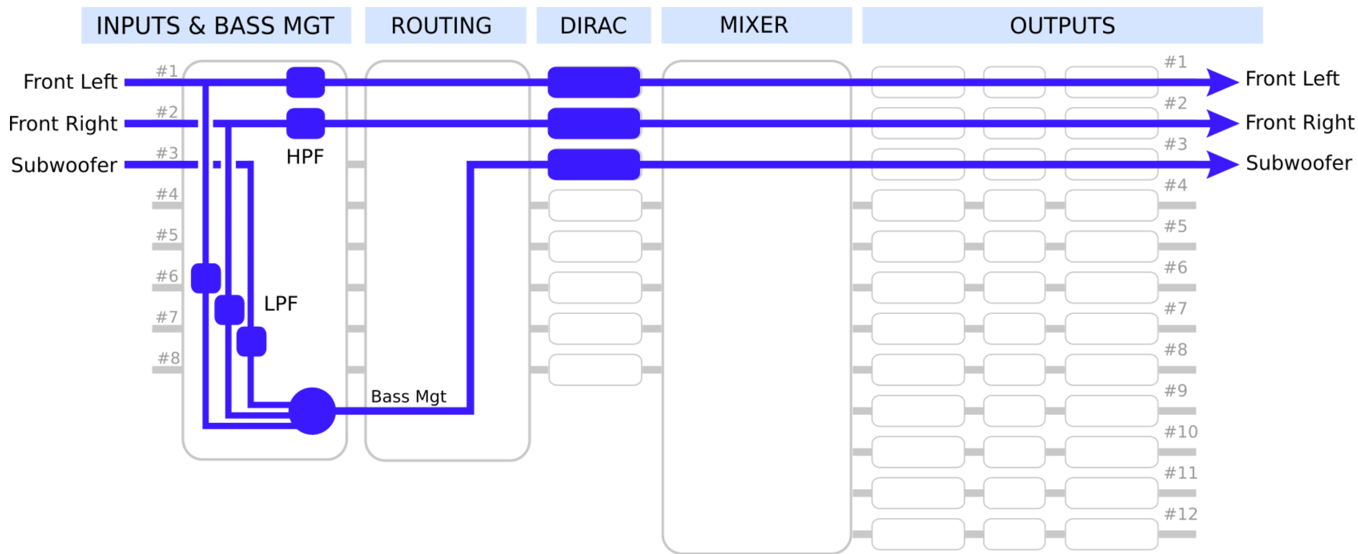


Figure 8. Bass management when you have a subwoofer input feed

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:

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 9.

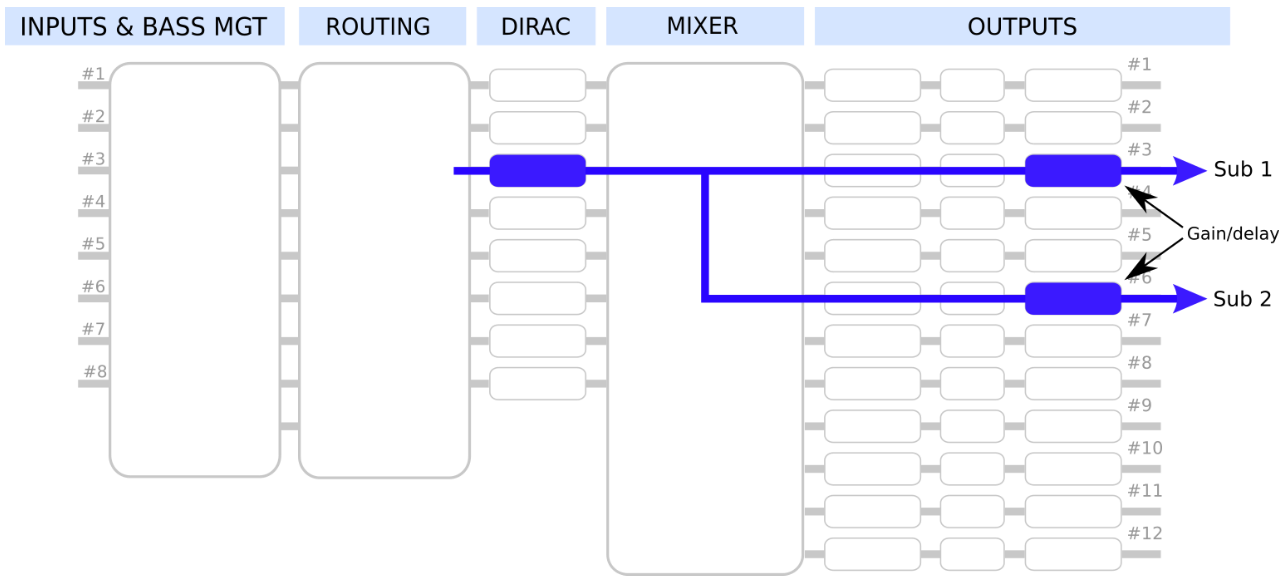


Figure 9. Processing flow example for multiple subwoofers

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:

	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 10.

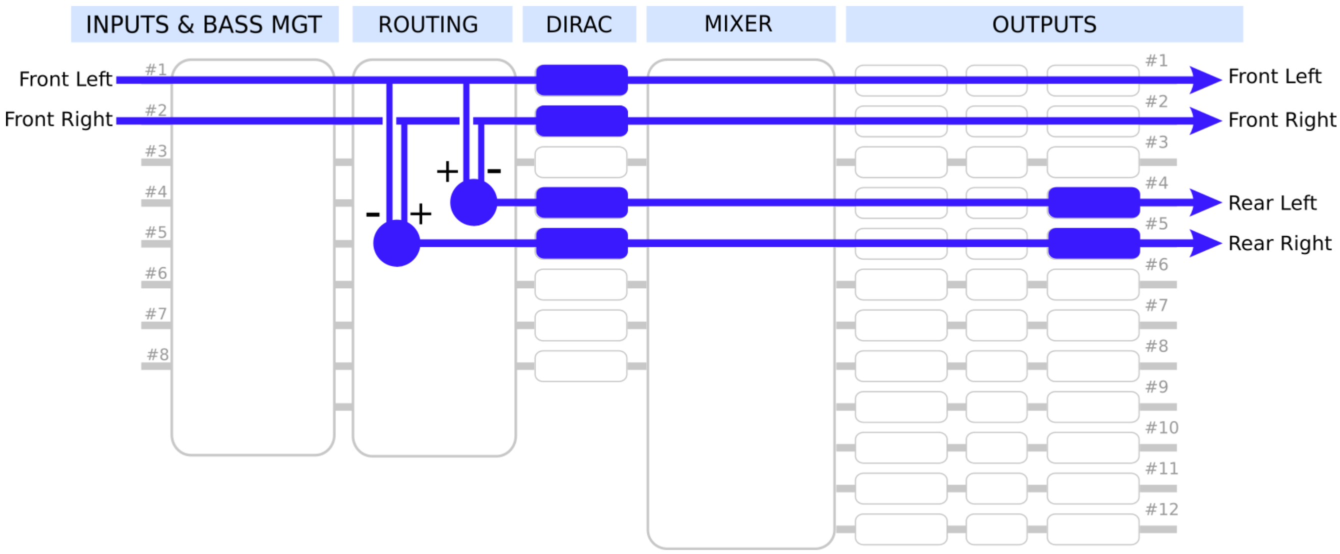


Figure 10. Processing flow for rear channel synthesis

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 11.

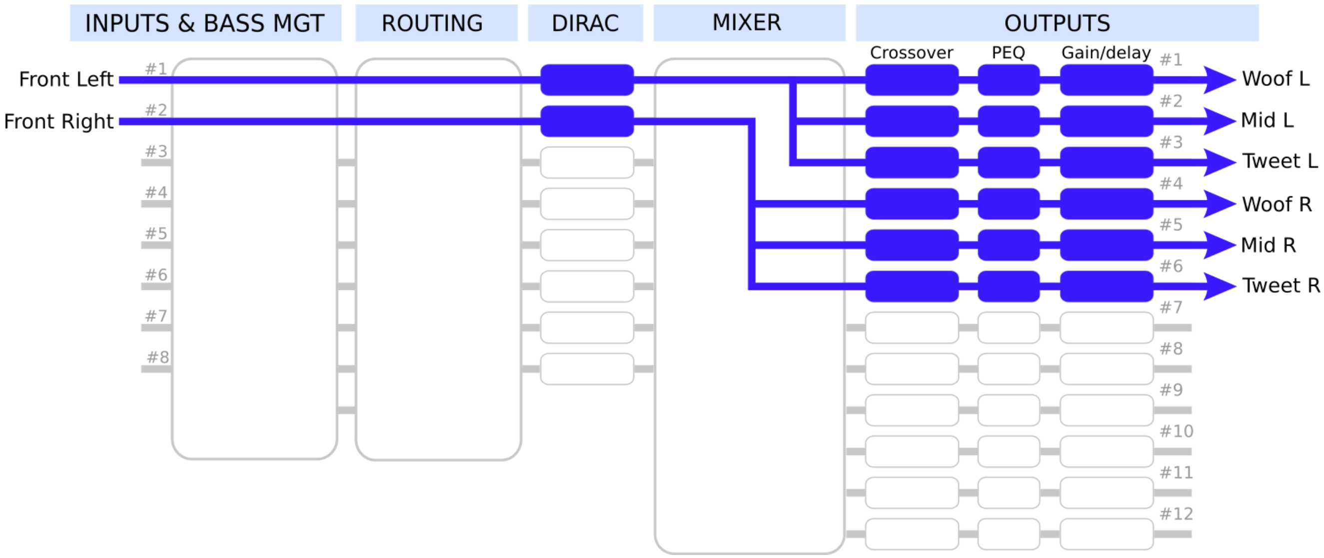
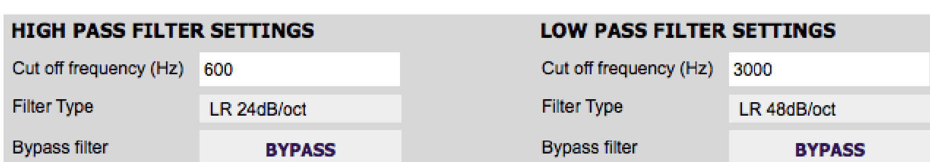


Figure 11. Processing flow for active speakers (three-way example)

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



Your system may well 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 11 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:

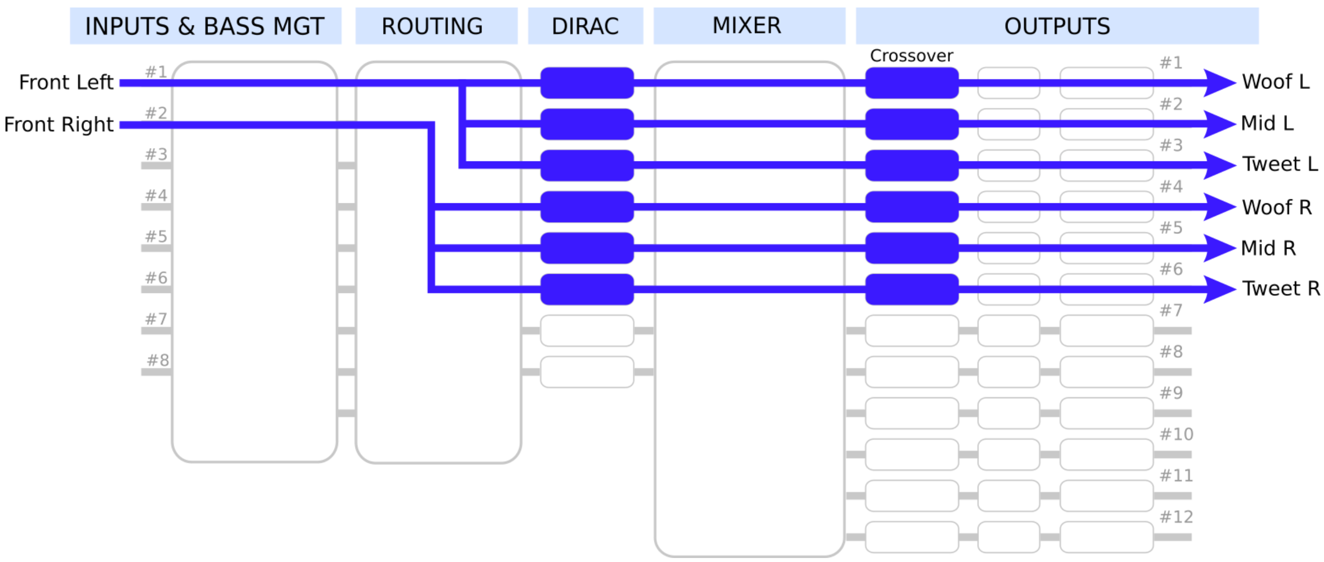


Figure 12. Alternative processing flow for active speakers (not recommended)

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 ACOUSTIC MEASUREMENT FOR DIRAC LIVE

The **Dirac Live Calibration Tool For miniDSP** uses a set of measurements made in your vehicle to gather all the acoustical information about your speakers acoustical environment that it needs to calculate the correction filters.

5.1 OVERVIEW

A Dirac Live calibration requires a total of nine measurements, with the microphone located in different positions. Typically, the microphone is pointed vertically (that is, at the roof). The first measurement must be taken at the center of the listening area, as this location sets the levels and delays of each speaker. Eight more measurements are then taken at locations spread around the listening area and at different heights.

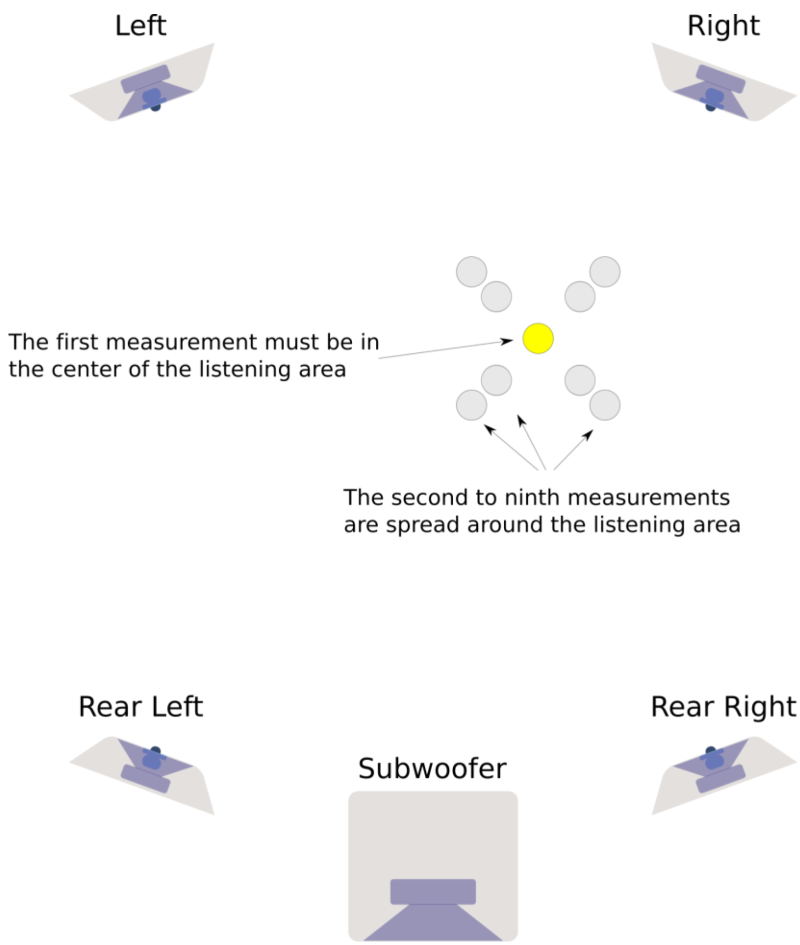


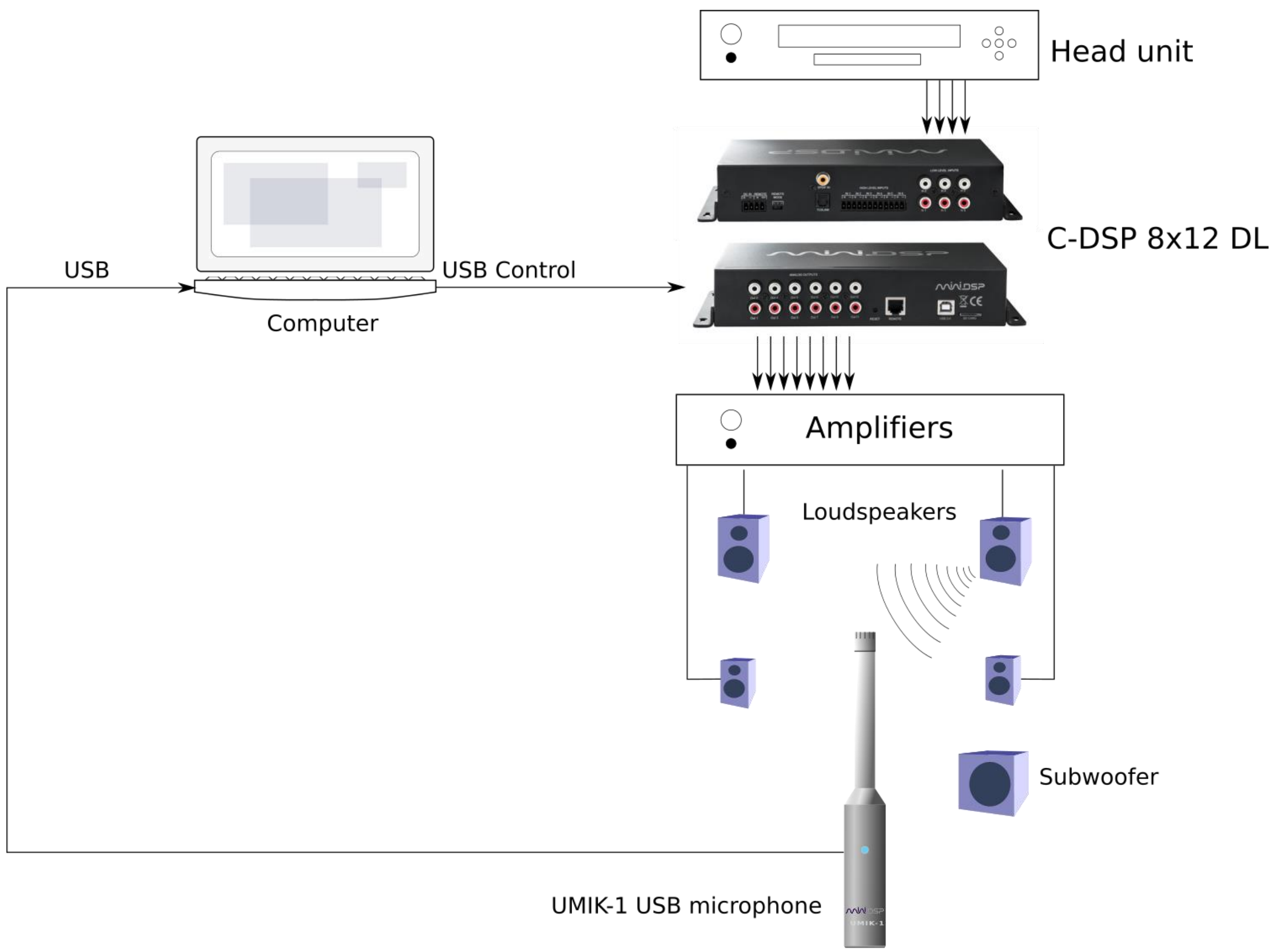
Figure 13. Typical measurement setup

In a vehicle environment, you can choose different sets of measurement positions in order to optimize for different situations i.e. with just the driver or with one or more passengers.

5.2 Connections

The figure below shows a typical connection diagram for performing acoustic measurement for Dirac Live calibration. No changes to the audio connections are needed. Simply:

- Connect the supplied USB (type A to type B) cable from the *C-DSP 8x12 DL* to a USB port on the computer.
- Connect a USB cable (type A to mini type B) from the UMIK-1 to a USB port on the computer.



You will need to arrange a method of holding the microphone in the needed locations and ensure that the USB cable is long enough. If necessary, a USB extension (up to a total USB cable length of 5 meters) can be used. In a vehicle environment, it's usually best to orient the microphone vertically (i.e. pointed at the roof) and use its "90 degree" calibration file (see Mic Config tab on page 43).

5.3 SOME NOTES AND CAUTIONS BEFORE YOU BEGIN

5.3.1 Ensure good measurement conditions

Measurements should be performed under good conditions if possible. While the measurement technique used by Dirac Live is quite robust, low-frequency noise (traffic, machinery, aircraft, storms) in particular can adversely affect measurement accuracy. A high level of ambient noise can degrade signal to noise ratio and prevent the algorithm from analyzing the test sweep signal properly. Minimize the effect of any external noise, ensure that measurement signal levels are adequate, and/or choose a suitable time for performing measurements.

5.3.2 Select the configuration/preset

Prior to running measurements for Dirac Live, you must ensure that you have selected the configuration preset that you will be using with the calibration.

For example, if you have made changes on the **Mixer** and **Outputs** tabs to set up an active crossover, be sure to select the configuration preset that has those settings each time you perform a calibration. If, however, you are not making any changes to the **Mixer** and **Outputs** tabs and all presets remain at the default settings, then it does not matter which one is selected.

5.3.3 Check subwoofer volume

Check that the subwoofer volume is set to zero. (Remember that this volume is relative to the master volume. Zero is typically the best setting to use for calibration.)

5.3.4 Only run one program at a time

Quit the **C-DSP 8x12 DL plugin** before starting **Dirac Live Calibration Tool for miniDSP (DLCT)**.

Be sure to quit the C-DSP 8x12 DL plugin. Running DLCT and the plugin at the same time can result in communication conflicts and odd behavior.

5.3.5 Don’t take shortcuts

It is important that all nine measurements are completed in order to ensure best results from the optimization algorithm. Being patient and thorough will pay audible dividends!

Note: it is good practice to save the project periodically while performing measurements (see below).

5.4 SAVING AND LOADING PROJECTS

Each set of measurements and the associated configuration settings are called a *project*. The project should be saved at regular intervals by clicking on the **Save** button. The default location for project files is **My Documents\MiniDSP\C-DSP\Projects** (Windows) or **Documents/MiniDSP/C-DSP/Projects** (Mac).

A project can be reloaded at any time by clicking on the **Load** button. This enables you to generate new correction filters for different target curves at a later date, or to redo any of the measurements. (Note: if you wish to change between the **Chair, Sofa**, or **Auditorium** listening environments, you will need to start a new project.)

5.5 CONFIGURING FOR MEASUREMENT

The **Dirac Live Calibration Tool for miniDSP (DLCT)** user interface is shown in this annotated screenshot.

Logo and status progress bar

This area shows a progress bar with current status when the program is performing calculations. If the program seems unresponsive at any time, check the status here.

Screen selection tabs

Each tab selects a different step of the calibration process. These are generally worked through in order, from top to bottom. This section covers the first four tabs; the final two are covered in Dirac Live Filter Design.

Load and save a project

A set of measurements can be saved to a file and reloaded at a later time. See Saving and loading projects.

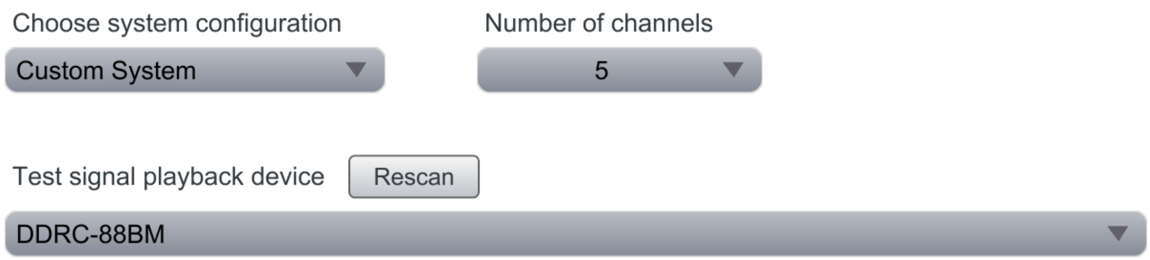
Back / Next

Use these two buttons to advance to the next tab when each is complete, or to go back to the previous tab to make alterations. The tabs at the left can also be clicked on directly.

Help open/close

Click on the small Help divider at the right of the screen to open a pane with help on the currently selected tab. Click on the divider again to close the help pane.

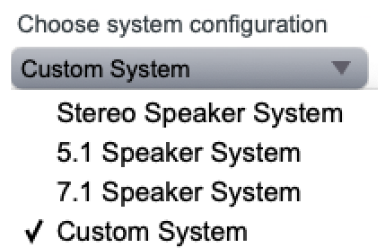
5.5.1 Sound System tab



On the **Sound System** tab, set the following parameters.

Choose system configuration

Use the dropdown menu to select your system configuration.



In a vehicle environment, the **Custom System** configuration is usually the best choice. When this is selected, an additional parameter appears as shown above: **Number of channels**. However, there are a range of possible scenarios; here are some common ones:

- For a stereo source and two speaker channels, select Stereo Speaker System. Do the same if you want to generate a subwoofer drive signal from the left and right input channels.
- For a “straight through” multichannel configuration, use Custom System and set Number of Channels to the number of input channels. For example:
  - Use **4** for a four-channel source and four speakers, or for a four-channel source with 4 speakers and a subwoofer.
  - Use **5** for a 4.1 source and 4 speakers plus a sub.
- For a 5.1 source that applies the “10 dB LFE gain,” use 5.1 Speaker System.

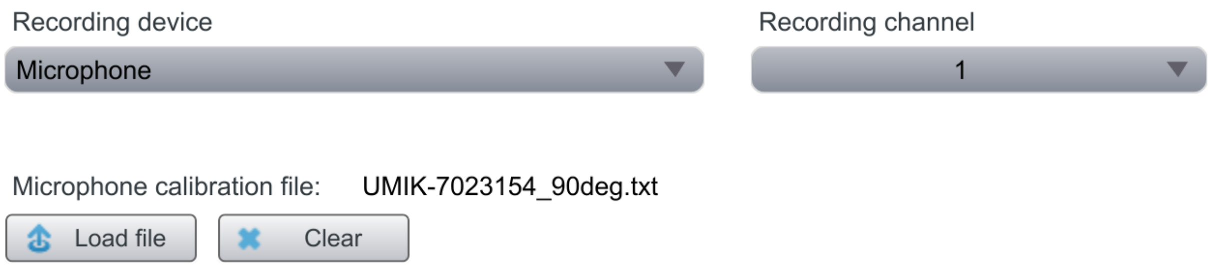
Test signal playback device

Preset to **C-DSP 8x12 DL**. This will ensure that test signals are sent into your audio system via the *C-DSP 8x12 DL* processor.

If the entry for **C-DSP 8x12 DL** is not showing, check that your *C-DSP 8x12 DL* processor is connected via USB and powered on, click the **Rescan** button, and then use the drop-down menu to select **C-DSP 8x12 DL**.

Once you have verified that this tab is correct, click the **Proceed** button.

5.5.2 Mic Config tab



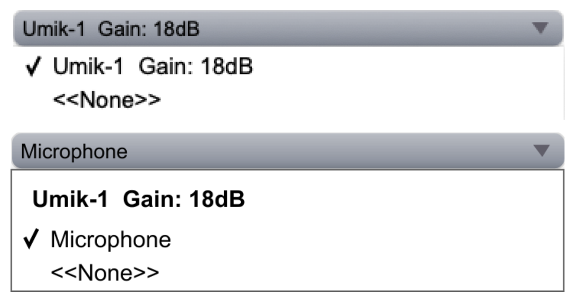
On the **Mic Config** tab, set the following parameters.

Recording device

Preset to the **UMIK-1**.

If the UMIK-1 is not showing, ensure that the UMIK-1 is connected securely to the computer via USB, then go back to the **Sound System** tab and click on **Rescan**. Then use the drop-down menu to select the UMIK-1.

Note: depending on your platform, the display on this screen and dropdown may vary slightly, as shown at right.



A miniDSP UMIK-1 is **required**. You cannot use a different microphone with the miniDSP version of Dirac Live Calibration Tool.

Recording channel

Select **1** from the drop-down menu.

Microphone calibration file

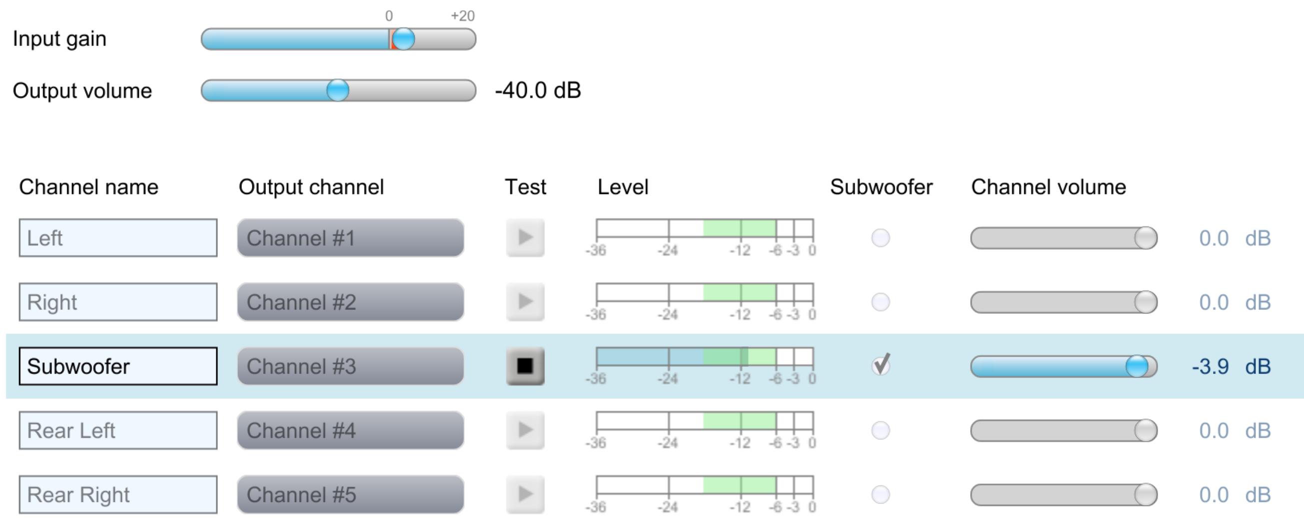
To download the unique calibration files for your microphone, go to the UMIK-1 page and enter your microphone's serial number. It is in the form xxx-yyyy and labelled on the microphone. Ensure that you download both the regular calibration file and the “90-degree” calibration file. The 90-degree file is generated specifically for use with miniDSP’s multi-channel Dirac Live® processors such as the *C-DSP 8x12 DL*, *DDRC-88A/D* and *nanoAVR DL*.

Then click on the **Load File** button and select your calibration file.

For in-vehicle applications, it is usually best to use the 90-degree calibration file as this is created specifically for the vertical microphone orientation. This file is downloaded with the suffix “_90deg” in the file name.

Once you have verified that this tab is correct, click the **Proceed** button.

5.5.3 Output & Levels tab



The **Output & Levels** tab is used to set the signal levels used in the subsequent measurements. For each channel you can set the following parameters:

Channel name

Type any name you like for each channel.

Output channel

By default, each input channel maps to the same numbered output channel (input channel 1 to output channel 1, and so on). The dropdown selectors can be used to change this mapping. Note that DLCT will not let you assign more than one output channel to each input channel.

Subwoofer

The subwoofer checkbox tells the Dirac Live analysis algorithm to use a different method to detect the impulse on that channel, which in turn affects the delay that will be assigned to that channel. This is needed because of the limited frequency response of the subwoofer.

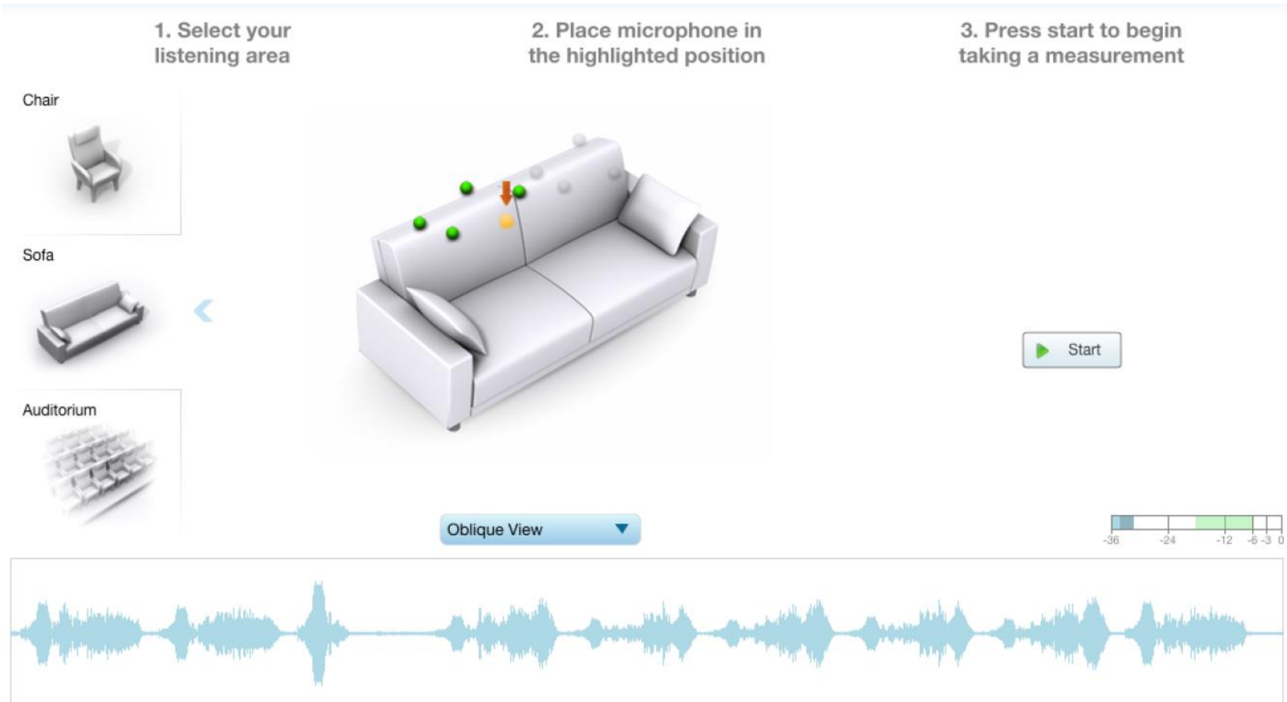
To set levels, we recommend following this procedure:

1. Check that the **Output volume** slider is set at the default starting point of -48 dB.
2. Click on the **Test** button for the left channel and gradually increase **Output volume**. You should hear pink noise playing from the left speaker. Continue to increase volume until it is at a moderate level, such that your voice would have to be raised to converse with someone sitting next to you.
3. Set the **Input gain** slider so that the blue bar on the level meter is about in the middle of the green section, or around -12 dB.
4. Click again on the **Test** button for the left channel to stop the test signal.
5. Click on the **Test** button for each of the remaining channels. If any channel is not in the green zone, use the **Channel volume** sliders to adjust the relative volume of the channels. (Some readjustment of **Input gain** and **Output volume** may also be needed.)

When done, click the **Proceed** button.

5.6 RUNNING THE MEASUREMENTS

Acoustic measurements are performed on the **Measurements** tab.



5.6.1 Select listening environment

The **Measurements** tab presents three different listening environments as a visual guide to positioning the microphone for each of the nine measurements: **Chair**, for a single listening seat; **Sofa**, for multiple listening seats; and **Auditorium**, which is intended for a large venue. Use the icons at the left of the screen to select the listening environment.

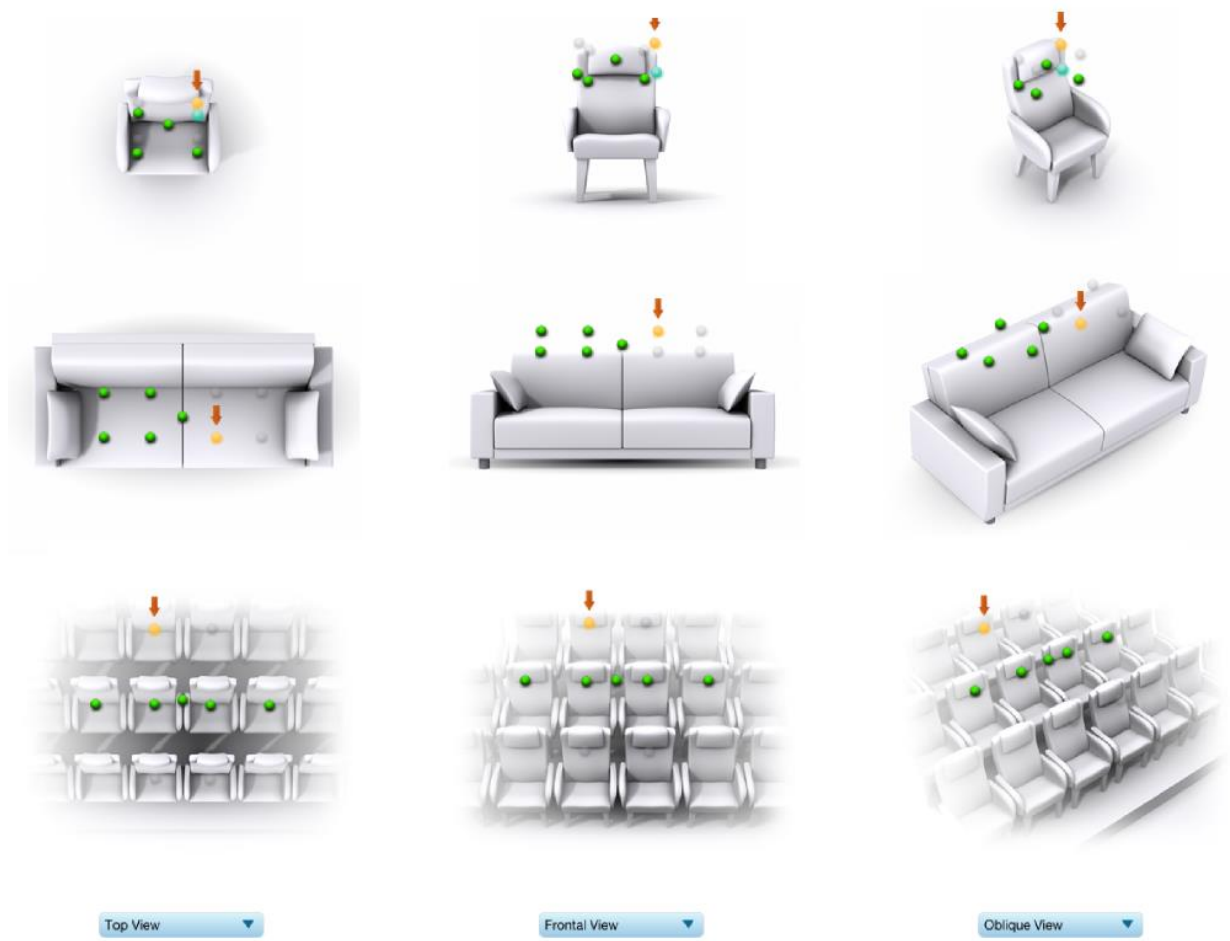
The center of the screen contains a pictorial representation of the selected listening environment, with dots marking the recommended microphone locations. Completed measurements are shown in green, while the next measurement to be done is highlighted in yellow and has a red arrow marker pointing to it. A drop-down menu underneath selects three different views, which should be used to help you place the microphone in a suitable location. (See summary diagram on next page.)

5.6.2 About measurement locations

While the visual guide indicates a suitable set of microphone locations, these locations can be varied to suit individual circumstances. It is, however, imperative that the first measurement is taken in the center of the listening area, as this measurement is used to set the levels and delays of each channel. The subsequent eight measurements should be well spread so that Dirac Live can acquire a good set of. Placing all microphone locations too close to each other may result in “over-correction” that will sound dry and dull.

For example, if using the **Chair** listening environment, spread the microphone positions over a circle with a diameter of a meter (three feet). Vary the height of the microphone up and down by 30 cm (one foot) from the initial position. If using the **Sofa** or **Auditorium** listening environment, again spread the measurement locations over the full listening area and vary microphone height by a significant amount.

You should feel to adjust the locations depending on the circumstances – for example, so avoid placing the microphone in the “shadow” of a seat back. The important thing is to ensure that the measurement locations are spread out over the whole listening area and that the microphone is moved vertically as well as horizontally.



5.6.3 Execute a measurement

With the microphone in place at the central location and pointed vertically (towards the roof), click on the **Start** button. The *C-DSP 8x12 DL* will generate a test signal, audible as a frequency sweep through the front left speaker, then the right front, and so on through all channels. Finally, the frequency sweep plays through the left speaker again.

While the measurement proceeds, the time-domain response graph of the captured audio signal is displayed at the bottom of the measurement tab. (This graph is related to the magnitude response but is not the same display. Its purpose is to verify that the recorded signal level is in a suitable range.)

After the sweeps finish, the status bar will update with a progress indicator while DLCT performs calculations on the measurement. If the measurement was successfully captured, the red arrow marker will advance to the next location to be measured. Move the microphone accordingly and press Start again to run the next measurement.

5.6.4 If you receive an error

If the program indicates that the measurement was not successful, you will need to take corrective action. The most common errors are related to signal level:

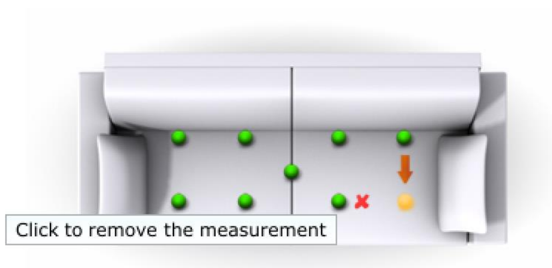
- The measurement signal is too low to ensure a clean capture.
- The measurement signal is too high and the audio signal has exceeded the maximum level (clipping). This is shown in red on the signal graph.

In either case, go back to the **Output & Levels** tab and adjust **Output volume**, **Input gain**, or the **Channel volume** slider for the channel that caused the problem. Then re-run the measurement. (You do not need to redo the measurements you have already successfully completed.)

5.6.5 Viewing and redoing measurements

Click on the green dot for any completed measurement to display its measured time-domain response graph.

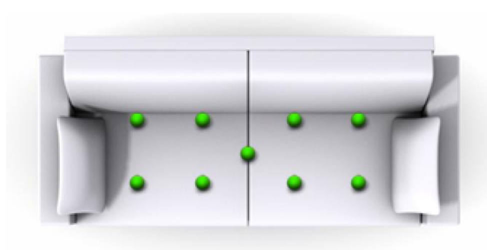
After clicking on a green dot, a small red "X" will appear next it. Click on the "X" to delete the measurement. The status bar will indicate that the program is recalculating parameters.



To redo a measurement, delete it, move the microphone to the appropriate location, and click on **Start**. Note: if more than one measurement is deleted, the marker will move to the lowest-numbered one.

5.6.6 Completing the measurements

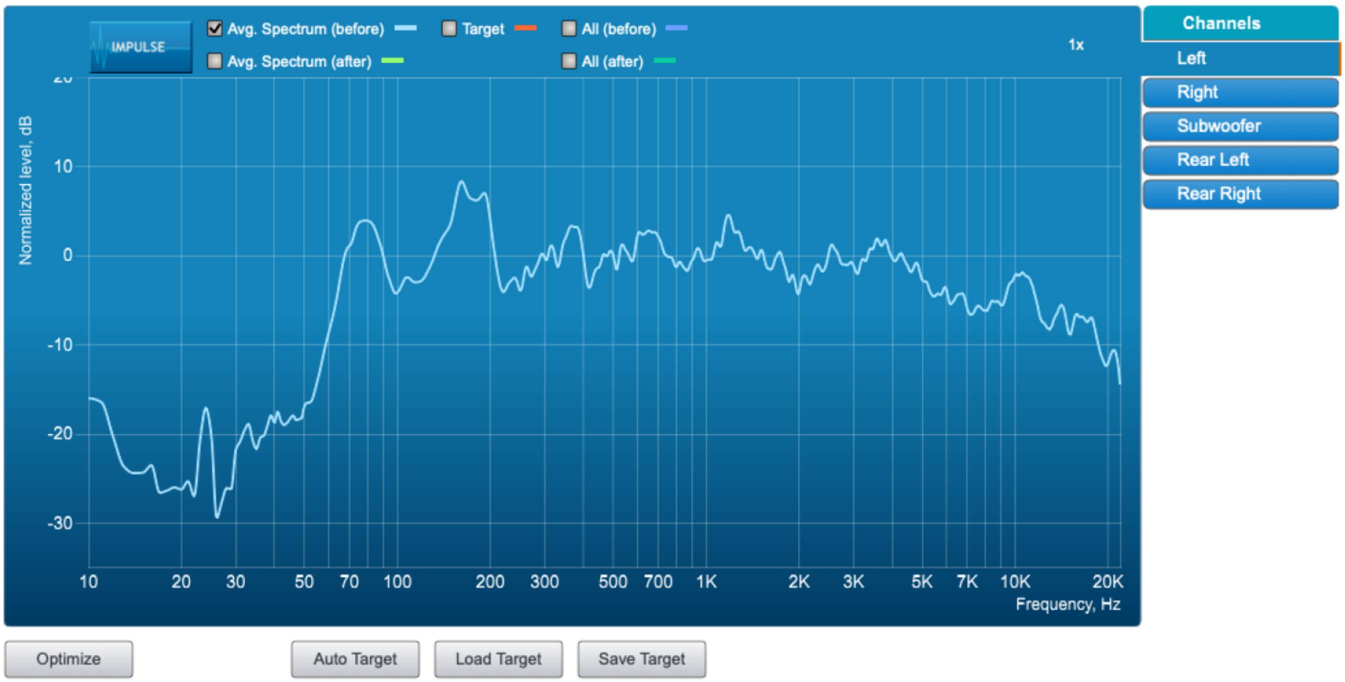
Once all nine measurements have been successfully completed, all locations will be marked in green. Click the **Proceed** button to go the Filters screen.



![miniDSP]

6 DIRAC LIVE FILTER DESIGN

The **Filter Design** tab shows graphs for the various channels. Click on the tabs at the right to display the response graphs for different channels.



For each graph, a number of variants can individually be turned on and off with the checkboxes above the graphs.

Avg. spectrum (before)

The average of the measured magnitude responses. These plots are shown in light blue.

Avg. spectrum (after)

The predicted average magnitude response after correction. These plots are shown in green and can be viewed only after filters have been generated with the **Optimize** button.

Target

The target curve – that is, the desired magnitude response. This curve is user-adjustable so you can fine-tune it to best suit your preferences. See Designing your target curve on page 51.

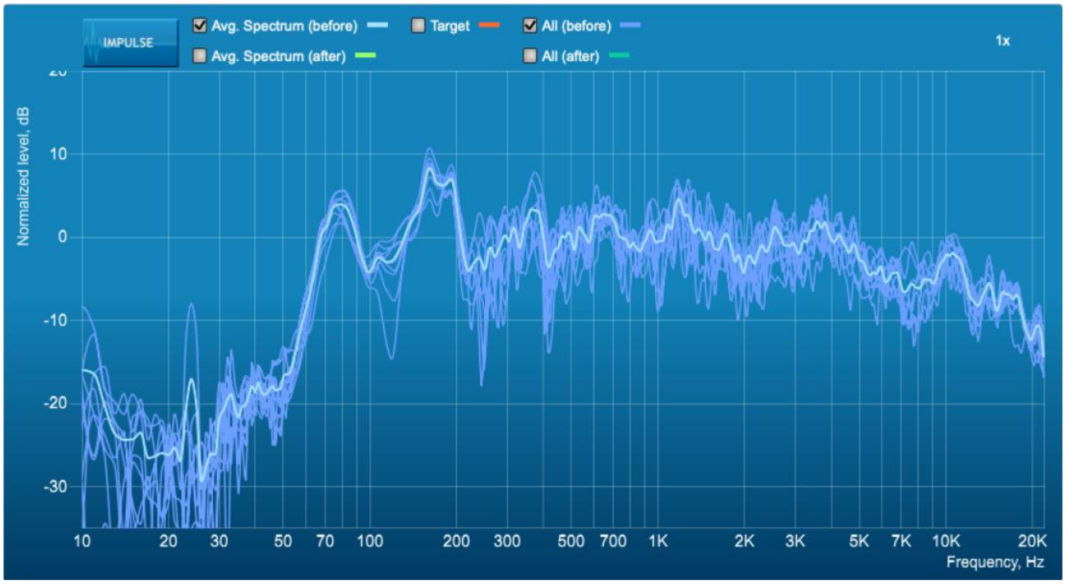
All (before)

All of the measured magnitude responses. These plots are shown dark blue.

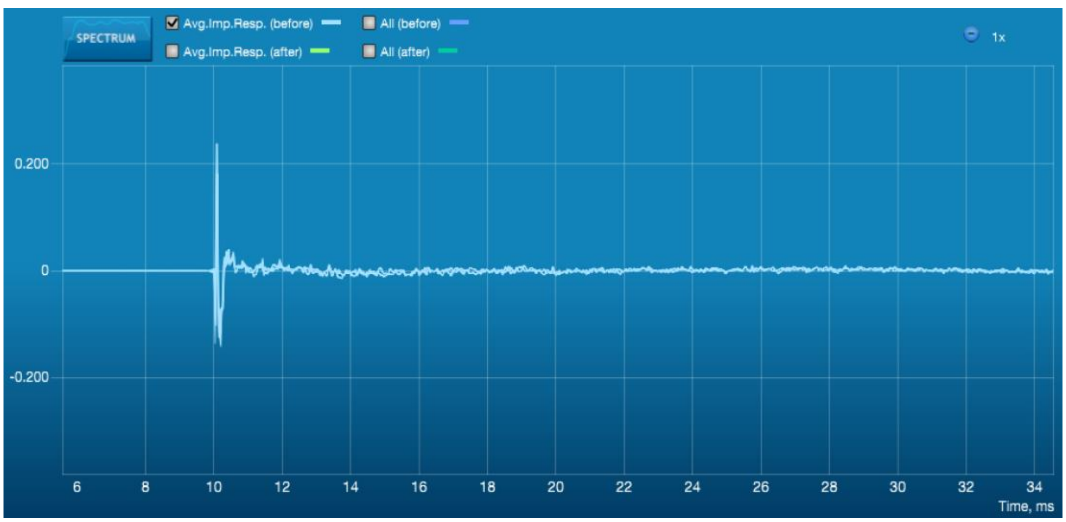
All (after)

All of the predicted magnitude responses after correction. These plots are shown in dark green and can be viewed only after filters have been generated with the **Optimize** button.

The graphs showing all nine measurements are useful for seeing how much variation there is across the listening area:



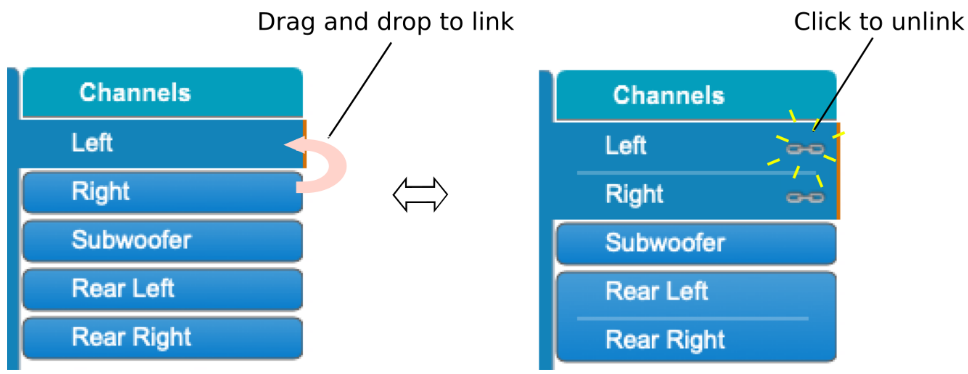
To display the impulse response instead of the magnitude response, click on the **Impulse** button at the top left of the display. All nine individual impulse responses can be shown as well as the average response. The predicted responses after correction can be viewed after filters are generated with the **Optimize** button (see Generating correction filters on page 54).



To return to the magnitude response, click on the **Spectrum** button.

6.1 WORKING WITH GRAPHS

Channels can be displayed individually or in groups. To link channels into a group, drag a channel name onto the channel currently being displayed. When channels are linked, their graphs display together, and they share the same target curve.

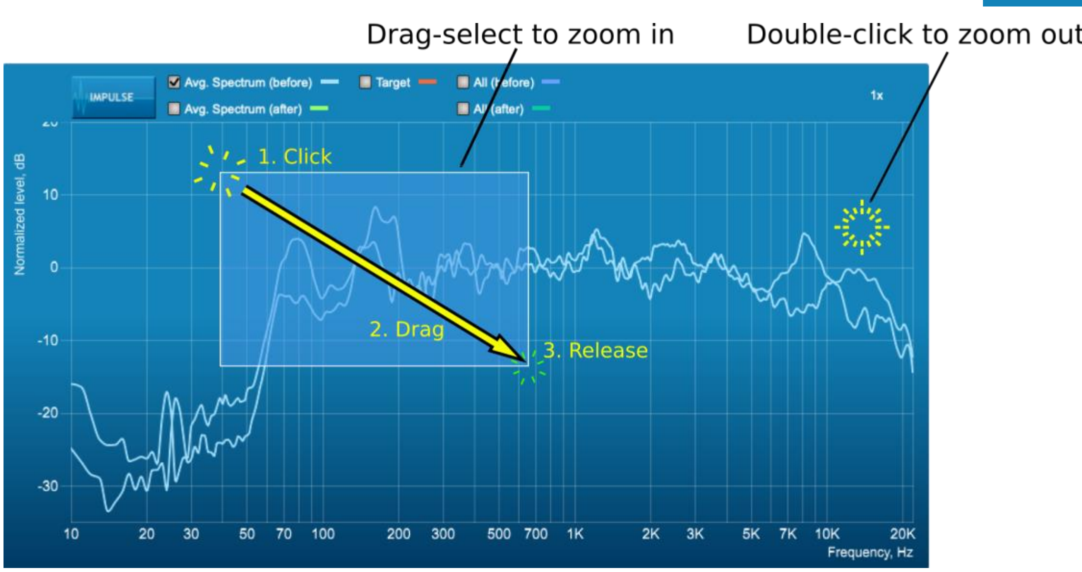


To unlink a channel, click on chain icon to the right of the channel name. It will then be unlinked from the other channels in the group.

Linking channels makes it easier to experiment with target curves. Initially, you may wish to link all speaker channels together. Once you have target curves that sound right to you in your installation, then experiment with different target curves for different groups e.g. for front and rear.

The response graphs can be viewed at a larger scale. To zoom in and out:

- Drag-select a region of the graph to zoom in on it. (Click the left button, move the mouse while holding the button, release the button.) You can then drag-select a region again to zoom in further.
- Double-click on the graph to zoom back out to the previous zoom level, or click on the small "–" sign next to the zoom indicator at the top right of the display.

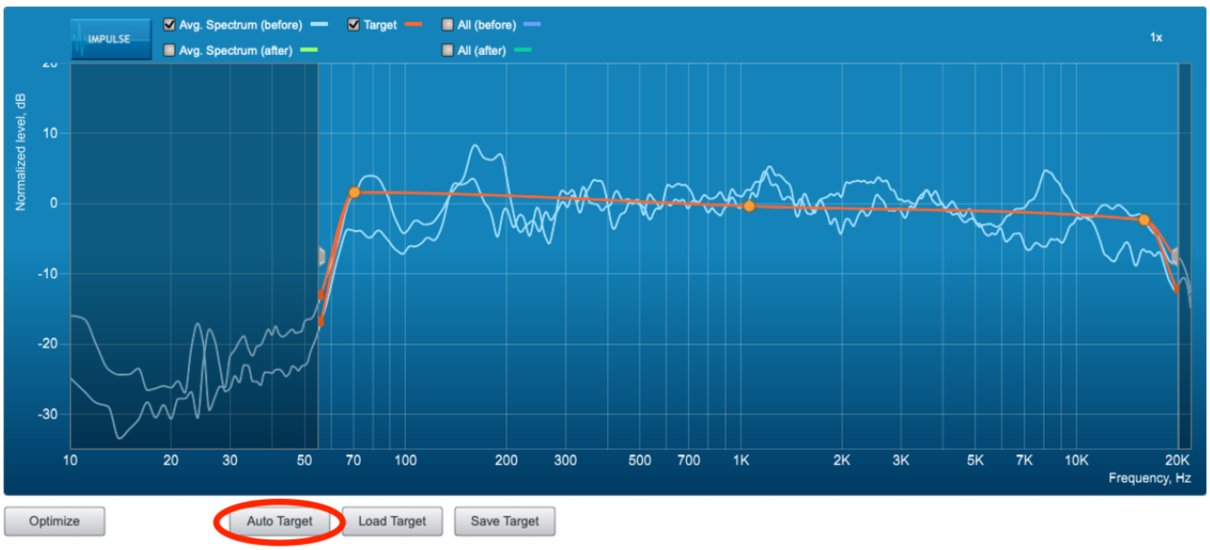


6.2 Designing your target curve

The *target curve* is the desired magnitude response with the *C-DSP 8x12 DL* processor performing digital room correction.

6.2.1 The Auto Target

When first viewing the **Filter Design** tab, an estimated target curve suitable for your speakers is shown as the red curve. This calculated target curve can be restored at any time by clicking on the **Auto Target** button.



Note: restoring the auto target will erase the current target curve. If you wish to keep it, you can save it to a file – see Saving and loading target curves on page 53.

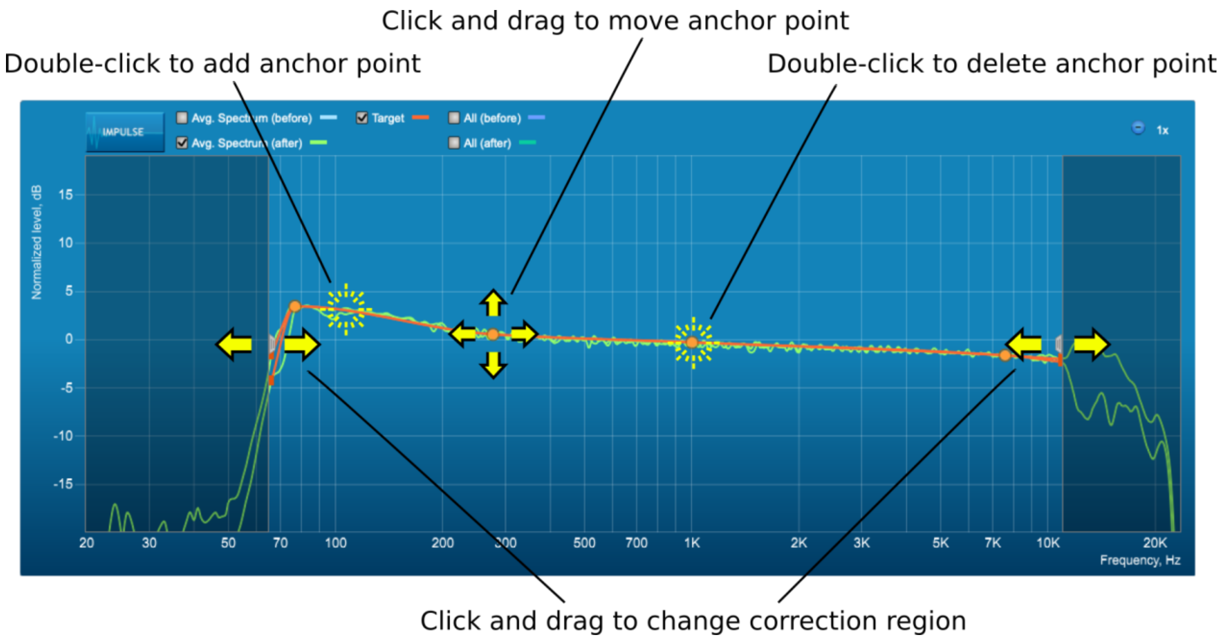
6.2.2 Editing the target curve

You can edit the target curve to produce any desired magnitude response. This is done with the use of *anchor points*, shown as orange dots on the curve:

- Drag an anchor point to move it.
- Double-click on the target curve to add an anchor point.
- Double-click on an anchor point to delete it.

The regions to the left and right of the response graphs that are shaded in a darker color are excluded from magnitude response correction. You can adjust the frequency range for your system and preferences. For example, low-frequency noise (traffic, machinery) may be present when measuring, so it is best to adjust the frequency range to exclude these frequencies from the correction. Or, you can experiment with the upper curtain to see whether you prefer full-range correction or correction over a limited region.

To alter the region of correction, drag the grey handles on either side of the graph. Note that you can't drag these handles over an anchor point, so you may need to move or delete an anchor point that is "in the way."



If channels are linked, the same target curve is used for that group of linked channels. To create a separate target curve for a single channel, unlink it as described in Working with graphs.

6.2.3 Guidelines for target curve design

Care should be taken to create a target curve that works well with your speakers and in your vehicle, as well as suiting your personal preferences. Small changes to the target curve can have significant effects on the tonal quality of the system, so it is important that you experiment with different target curves to find the optimum.

If you initially don't achieve a satisfactory result, please ensure that you have spread your measurements over a sufficiently large area and with sufficient variation in height. The following guidelines will help you understand how to adjust your target curve.

Low-frequency extension and boost

All loudspeakers have a natural low-frequency roll off. Setting the target curve to boost the region below the speaker's natural roll off frequency *may* result in overdriving the speakers, especially with smaller loudspeakers.

The auto-target estimates the low-frequency roll-off and curve. You should determine by listening whether this estimate is suitable for your system and adjust the target curve accordingly.

High-frequency "tilt"

The target curve is the desired measured response of loudspeakers *in your vehicle,* in contrast to measurements made of a loudspeaker during design under anechoic (measured in free space) conditions. The target curve typically has a downward-sloping or "tilting" response at high frequencies, due to the effects of limited dispersion at high frequencies and greater acoustic absorption.

A completely flat target curve is therefore usually not desirable and will tend to sound thin or bright. Start with a target curve that follows the natural measured behavior of your speakers, and then experiment with greater or lesser degrees of tilt in the treble region to obtain the most natural timbral balance.

Low-frequency adjustment

A completely flat response at low frequencies, with complete elimination of peaks due to resonance modes, may sound light in the bass. Especially in a confined environment like a vehicle, the region below 100 Hz should typically be boosted to give a more balanced sound.

Magnitude response dips

In some cases, it may be helpful to adjust the target curve to follow dips in the magnitude response. This can occur where, for example, the direct acoustic path to the speaker is obstructed. In such a case, adjusting the magnitude response to follow the dip may produce the best sound.

Unlinking left and right channels

Usually, the corresponding left and right channels (front left and right, surround left and right, and rear left and right) should remain linked for target curve adjustment, to ensure that both sides produce the same response across the listening area. You can also try unlinking channels and making separate adjustments on each side.

6.2.4 Saving and loading target curves

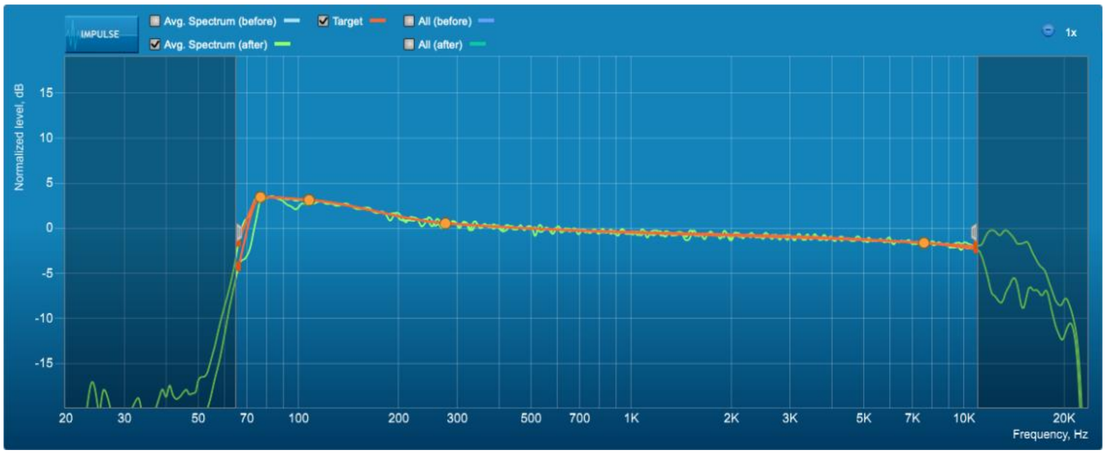
To allow you to experiment with different target curves, you can save a target curve to a file and reload it at a later time. Click on the **Save Target** button to save the target curve of the currently displayed channel or group of channels.

To load a target curve, click on **Load Target**. The currently displayed channel or group of channels will have its target curve updated. (Loading a target will erase the current target curve, so be sure to save it first if needed.)

6.3 GENERATING CORRECTION FILTERS

Once you have a target curve set to your satisfaction, click on the **Optimize** button.

The status bar will update as the algorithm progresses. The entire algorithm may take some time to complete, depending on the speed of your computer. When the algorithm completes, the predicted average magnitude response will be shown in green. (The predicted impulse response can be viewed by clicking on the **Impulse** button.)



Dirac Live Calibration Tool for miniDSP will contact the Dirac license server to verify its license, so you will need to be connected to the Internet to perform this step. If a firewall is in place, it must allow HTTP (normal web traffic) to pass. Otherwise, an error such as the following may appear:

Once the filters are generated, click the **Proceed** button.

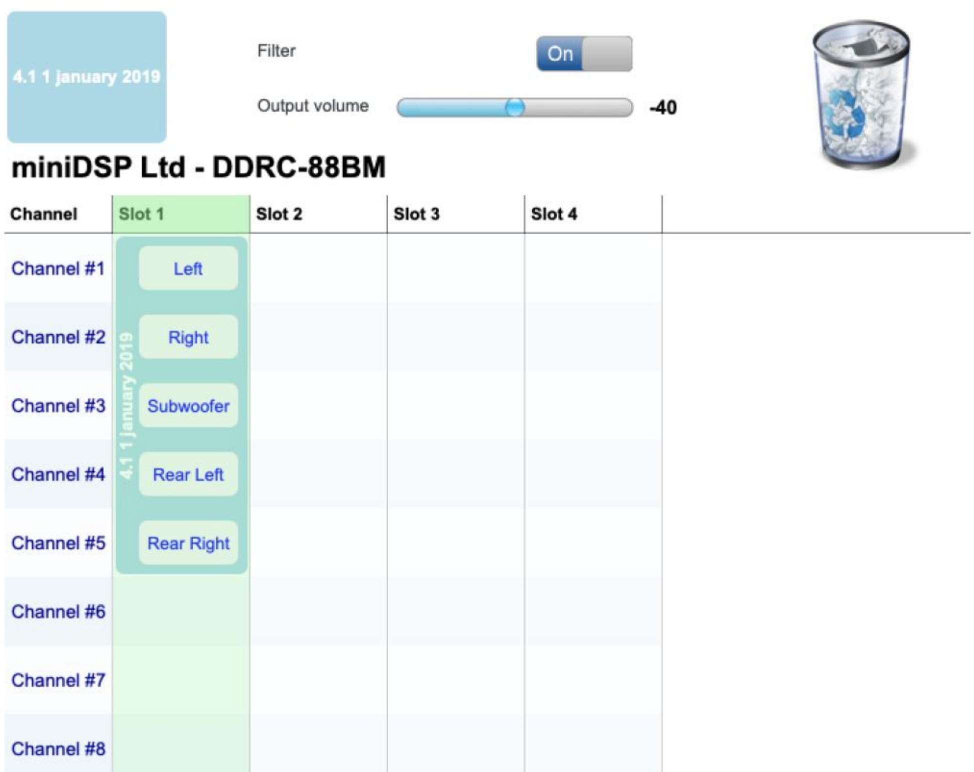
6.4 LOADING FILTER SETS

The **Export** tab initially shows four empty “slots” for filter sets (a filter set is one filter for every channel). Filter sets are managed with a “drag and drop” metaphor:

- To load the most recently generated filter set into the processor, drag the box at the top left (in this example labeled “4.1 1 January 2019”) and drop it onto an empty slot.
- To remove a filter set, click on its name (oriented vertically), drag it from the slot and drop it on the trashcan icon at the top right.

You must load the filters into the same slot as the preset selected when running the Dirac Live calibration measurements.

To load a filter set into a slot that already has filters loaded, first delete the loaded filter set by dragging it onto the trashcan icon, then drag and drop the current filter set onto the now-empty slot.



The two main controls on this tab are:

Filter

Turn this on to enable the Dirac Live® correction filters.

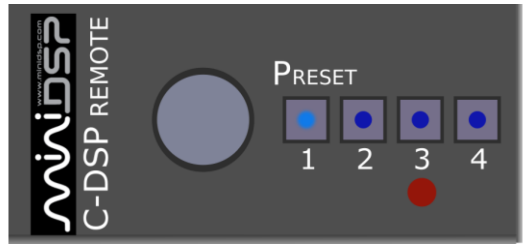
Output volume

Move the slider to adjust the master volume of the processor. Output volume can be also be adjusted by the front panel control knob or an infrared remote control.

7 REMOTE CONTROL

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

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



7.1 STATUS INDICATORS

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

7.2 OPERATION OF THE 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 in now in operation. (Note: The preset cannot be changed with the wired remote while there is a USB connection to the computer. Use the plugin instead.)

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

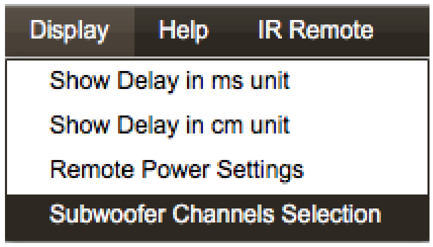
Press button 4 to change the mode. Press buttons 1 and 2 to return to normal operation.

To quickly adjust subwoofer volume while in Master Volume mode:

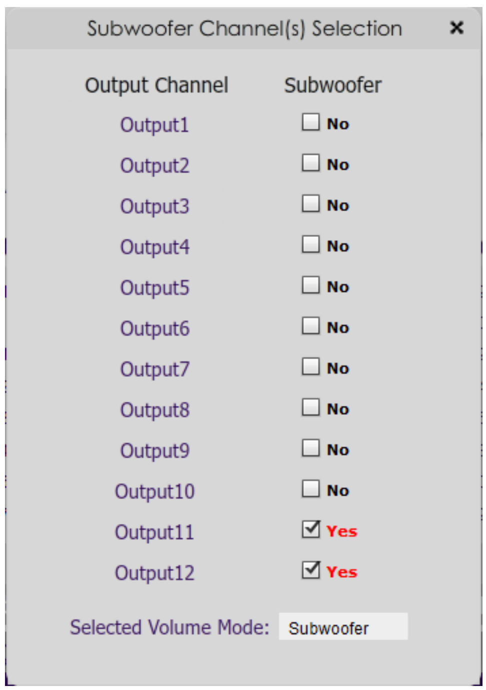
1. Press buttons 1 and 2, then button 4 to switch to Subwoofer Volume mode.
2. Adjust the rotary control until the subwoofer level sounds right.
3. Press button 4 to switch back to Master Volume mode.
4. Press buttons 1 and 2 to return to normal operation (for preset selection).

7.3 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.

7.4 USING THE MINIDSP INFRARED REMOTE

The remote control provided with the processor controls all key runtime functions.

Source

Cycles between the TOSLINK and SPDIF digital input sources. (The analog inputs are always active.)

1, 2, 3, 4

Switches to the selected preset. Note that 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.

[Bell]

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

Vol

Reduce or increase the volume. Each press changes the volume in 0.5 dB. Holding down a button will accelerate volume change to 3 dB steps.

Mute

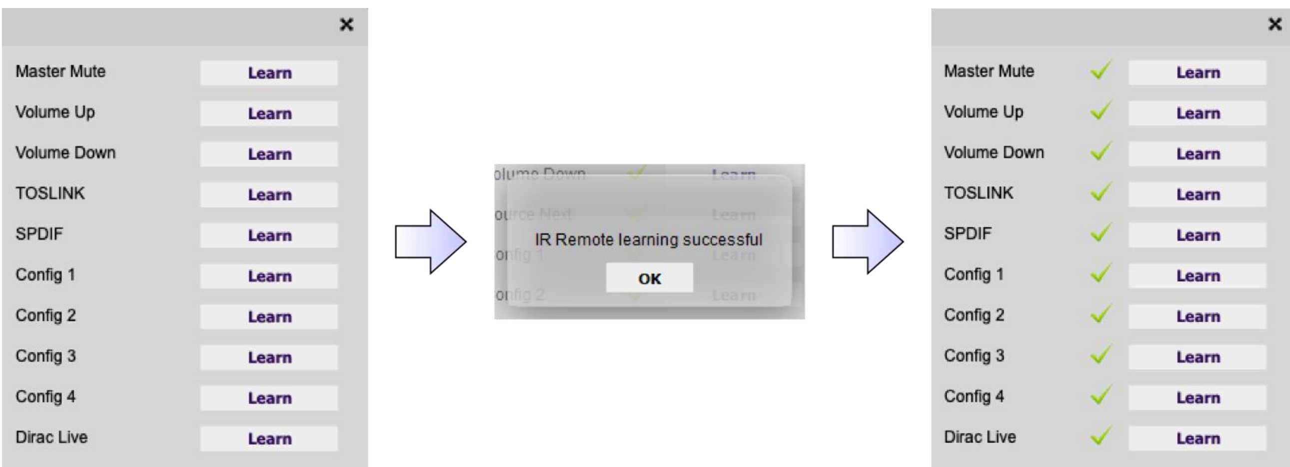
Mutes and unmutes audio output.

7.5 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:

- 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:



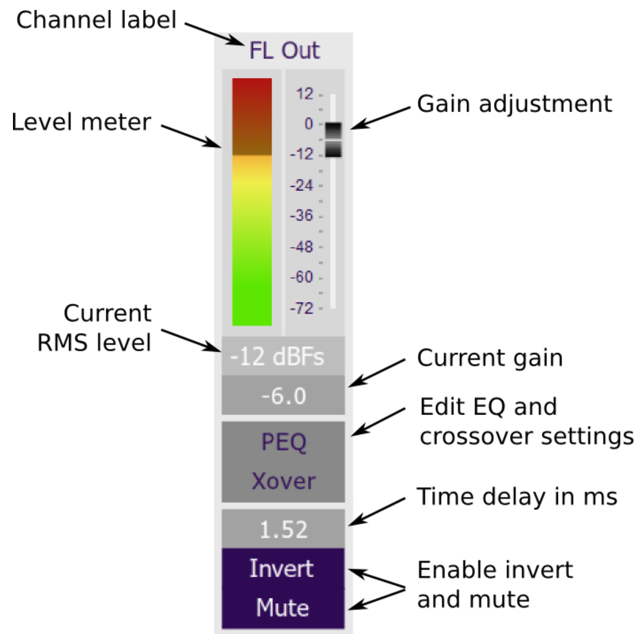
To "unlearn" a command, press the **Learn** button and wait for the plugin to time out. Note that you cannot "learn" the miniDSP remote – it will always work, even if you learn another remote's codes.

8 Plugin Reference

8.1 Output channel processing

8.1.1 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.



8.1.2 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.)

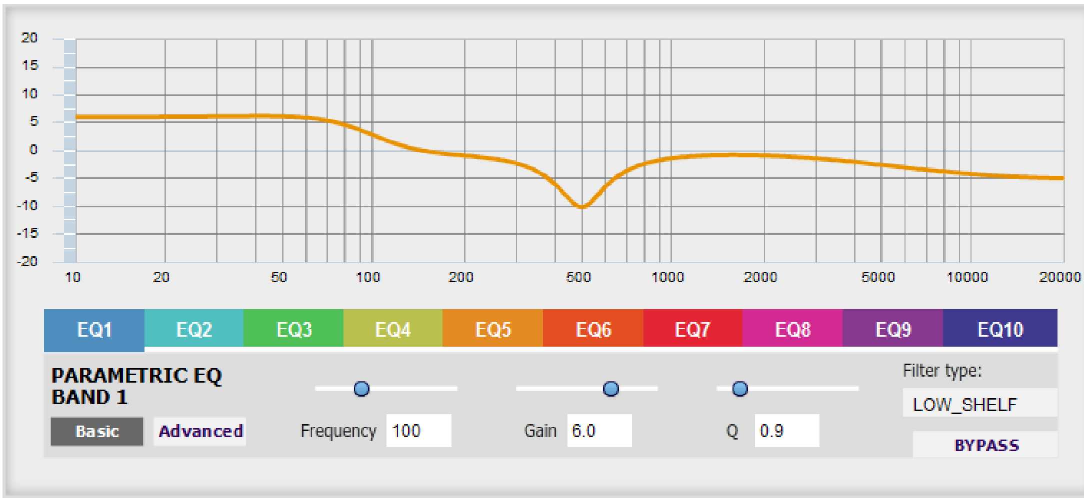
Gain adjustment

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.

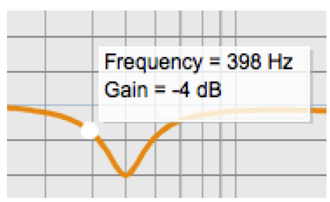
8.1.3 Parametric EQ (PEQ)

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 66.

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

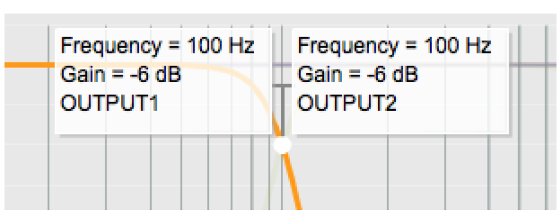
8.1.4 Crossover (Xover)

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 66.

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

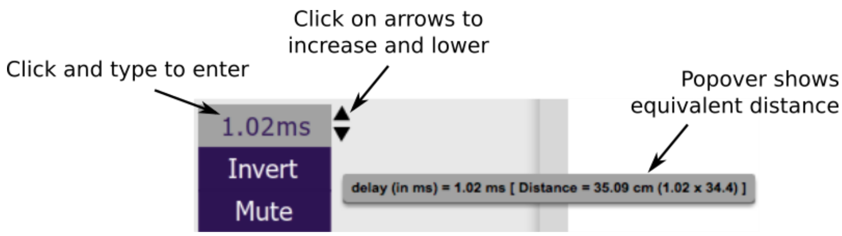
The **Bypass** button enables or disables a crossover filter. The filter is enabled if the button says "BYPASS" and disabled if the button says "BYPASSED" (see screenshot below).

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.

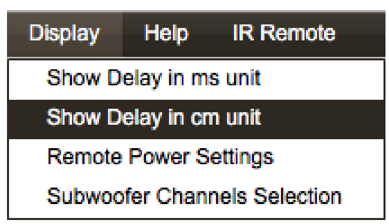
8.1.5 Time delay

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.

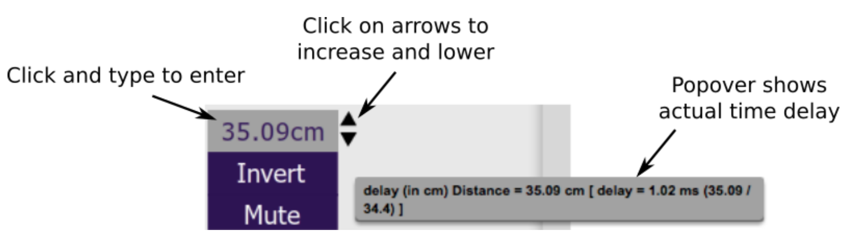


The time delay corresponds to a distance. This distance is shown in cm when the cursor is hovered over the time delay box. The maximum time delay of 20 ms corresponds to a distance of approximately 6.9 meters (about 22.5 feet).

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**.

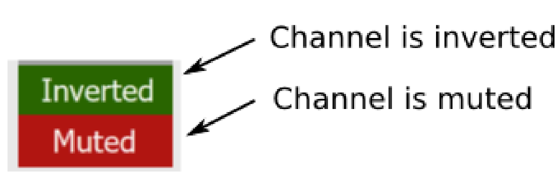


In this case, hovering the cursor over the entry box will display the actual time delay:



8.1.6 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.



8.2 Custom Biquad Programming

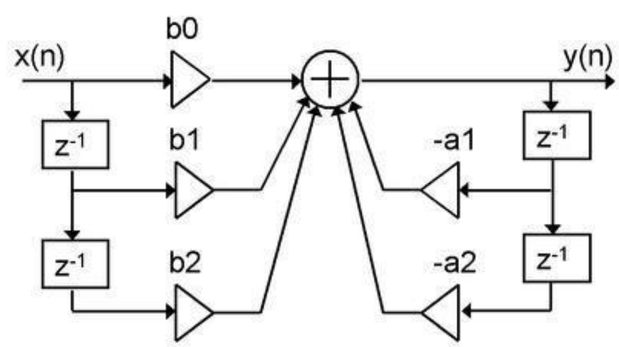
Custom biquad programming is available in the **PEQ** and **Xover** (crossover) blocks. Its purpose is to allow you to directly provide the low-level parameters aka *biquad coefficients* that control the digital filters of the processor, thus providing an almost infinite degree of flexibility.

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.

8.2.1 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:

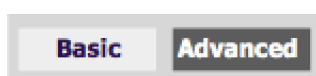
$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}$$



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*.

8.2.2 Using custom biquad programming

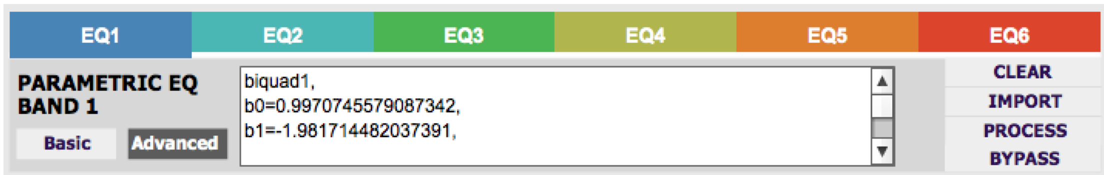
Each crossover block and PEQ filter has a selector that switches it to advanced mode:



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 six.

This example illustrates the correct file format:

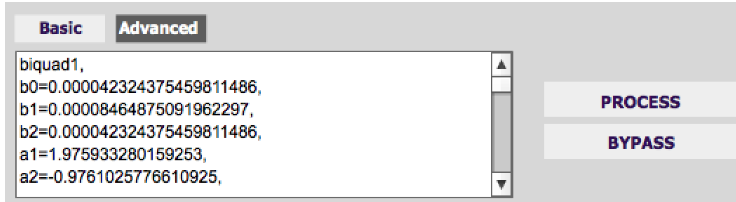
```
biquad1,
b0=0.998191200483864,
b1=-1.9950521500467384,
b2=0.996920046761057,
a1=1.9950521500467384,
a2=-0.9951112472449212,
biquad2,
b0=0.999640139948623,
b1=-1.9981670485581222,
...
biquad3,
...
biquad4,
...
biquad10,
b0=1.0010192374642126,
b1=-1.9950555192569264,
b2=0.9940580112181501,
a1=1.995060938714333,
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.



8.2.3 Biquad design software

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

8.2.3.1 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.

8.2.3.2 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 REW.

Note: as of this writing, REW does not have a dedicated setting for the C-DSP 8x12 DL. To generate filters in the correct format:

- Select the “Generic” filter type.
- Open the **EQ filters** screen and uncheck ten of the provided 20 filters.
- After clicking “Save filter coefficients to file,” select **48000** as the sample rate from the dropdown menu. Save the file with a name ending in “.txt”.
- After saving, open the file and delete the unused filters (all coefficients are set to 0.0 except *b0*, which is set to 1.0). Save the edited file.
- Use the plugin user interface to import the edited file into the desired PEQ filter bank. You should see the filter graph change to be the same as the “Filters” graph in REW (turn off “Show each filter” in REW to see the combined response).

8.3 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.

8.3.1 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.

8.3.2 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.

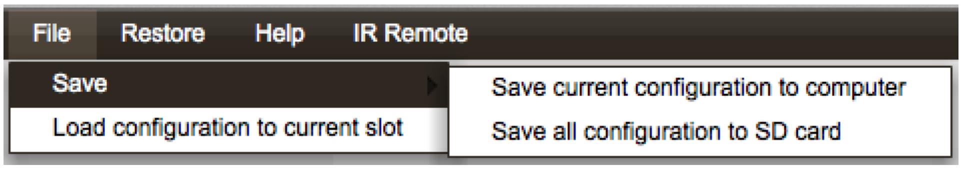
The plugin is in offline mode

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.

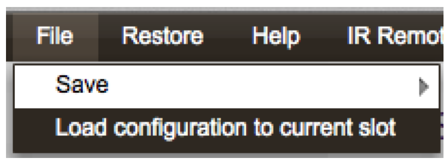
8.3.3 Saving and loading configurations

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**.



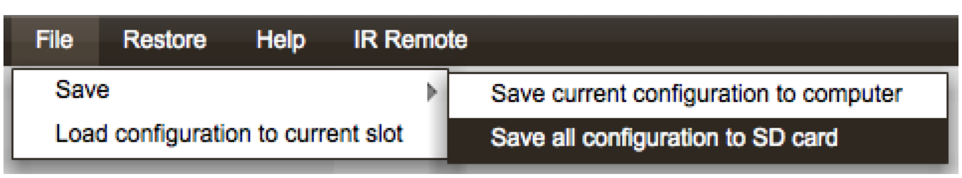
If the plugin is in online mode, the new configuration data will be downloaded to the processor immediately. If the plugin is in offline mode, the configuration will be loaded into the user interface only and will be downloaded to the processor the next time it is synchronized.

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.

8.3.4 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.

8.3.5 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.

8.4 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.

9 ADDITIONAL INFORMATION

9.1 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 Dirac Live Calibration Tool. 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 master volume and 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

9.2 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

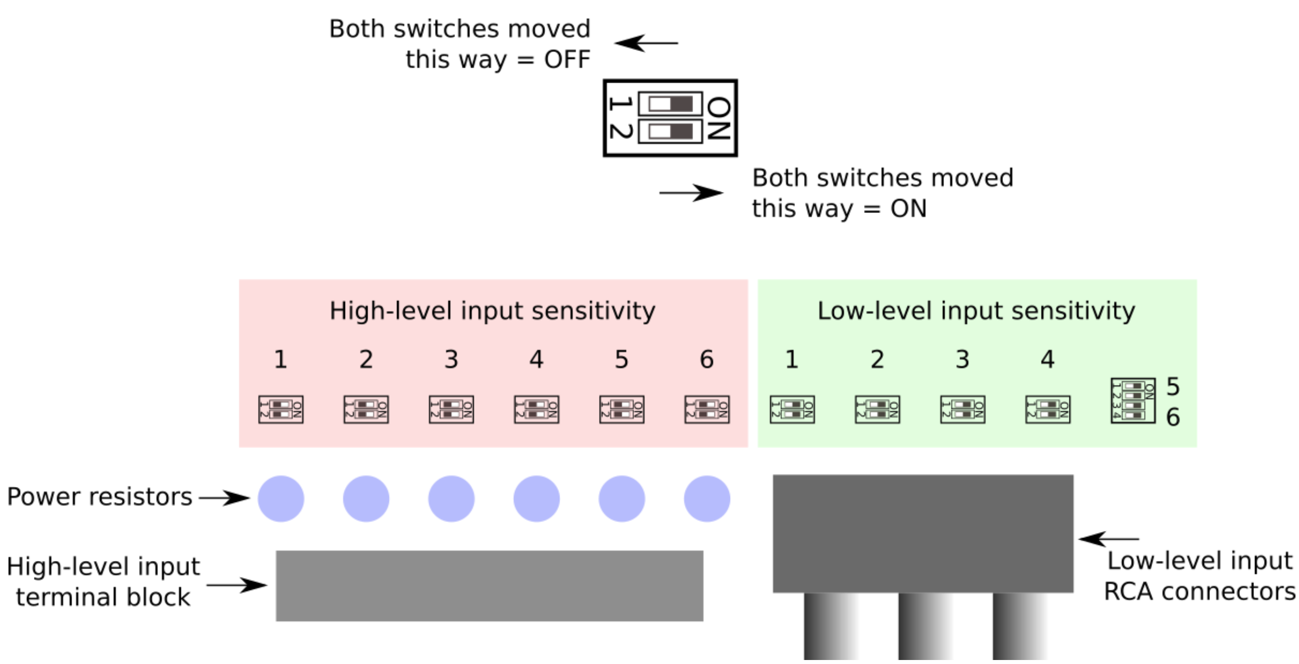
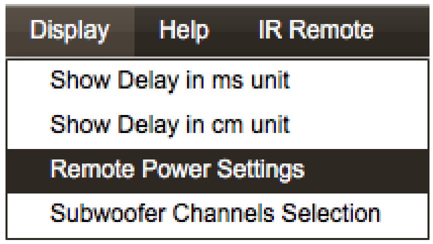


Figure 14. Input sensitivity switch locations

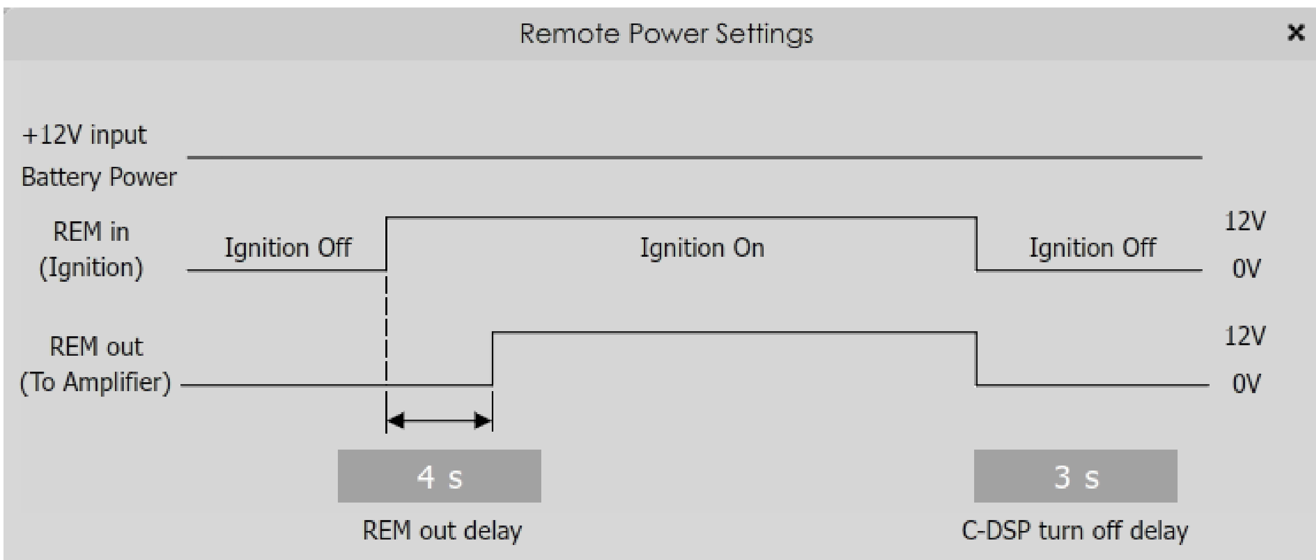
9.3 REMOTE TRIGGER TIMING

When the REMOTE MODE switch (see page 19) 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.

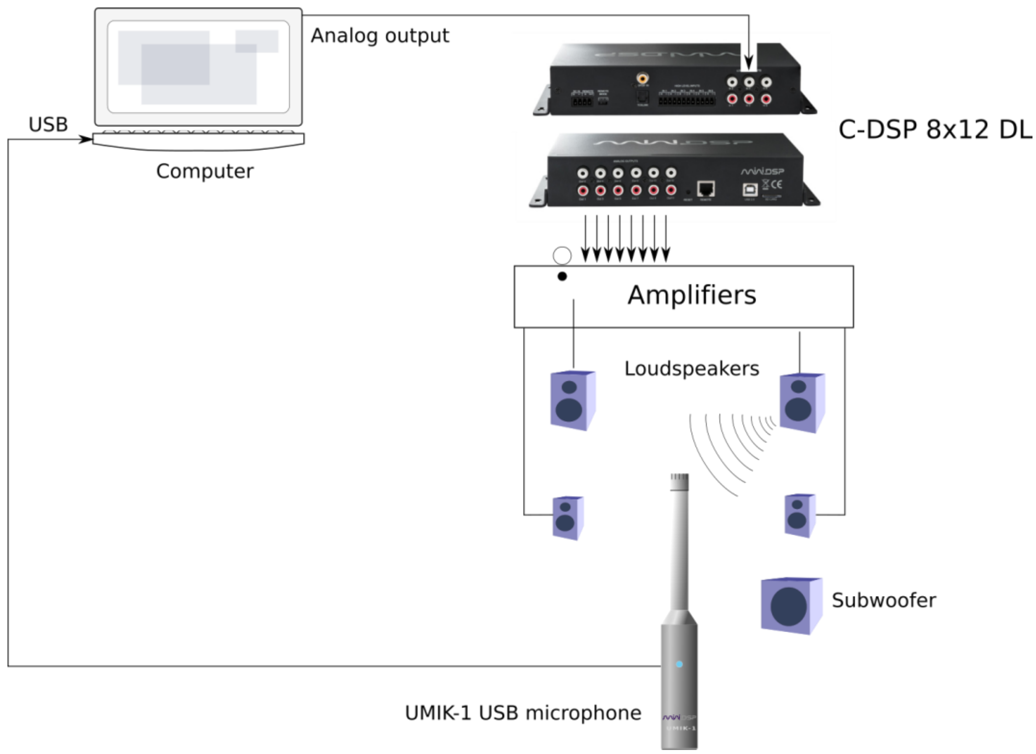
Enter the desired value for each time delay parameter and press the Enter key. When finished, click the close "X" icon at the top right, and the settings will be downloaded to the C-DSP 8x12 DL.

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.

9.4 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

9.5 TROUBLESHOOTING

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

9.5.1 C-DSP 8x12 DL plugin

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 Adobe Air environment may need a network connection the first time you run a plugin. Close the plugin program, ensure that your computer has a network connection, and restart the plugin. b. The Adobe Air environment may require a version update. Download the latest version from http://get.adobe.com/air/.
3	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.

9.5.2 DLCT

1	The C-DSP 8x12 DL doesn't appear in the **Sound System** tab	a. Check that the USB cable to the processor is firmly connected. b. Check that you do not have any other program running that is attempting to communicate with the processor, such as the C-DSP 8x12 DL plugin. c. Check that you have installed and run the miniDSP version of the Dirac Live software, called **Dirac Live Calibration Tool for miniDSP**. d. Go to the **Sound System** tab and click the **Rescan** button.
2	The measurement test signal produces no output	a. Ensure that the processor is connected correctly into the audio system. b. Check that the downstream amplification is powered on. c. Check that the downstream amplification is not muted and doesn't have gain/trim controls set to zero. d. Quit DLCT, open the C-DSP 8x12 DL plugin and click Connect. Connect an analog source to the inputs and confirm that signal levels are seen on input and output meters. e. Check that master mute is not enabled.
3	No input from measurement microphone	a. Check that the USB cable to the UMIK-1 is securely seated. b. Check that the UMIK-1 is selected in the **Mic Config** tab. c. Remove any USB hubs and extensions.
4	Insufficient recording level	a. Increase microphone level in the **Output & Levels** tab. b. Go to the Control Panel and view the Recording tab of the Sound pane. Select the UMIK-1 and view its Properties. In Levels, set the gain to 100. c. Increase system output volume.
5	Unable to generate correction filters (**Optimize** button)	a. Check that your computer is connected to the Internet and able to pass HTTP (web) traffic. b. Check that you do not have any other program running that is attempting to communicate with the processor, such as the C-DSP 8x12 DL plugin.

9.6 Firmware upgrade

See "Upgrade instructions for C-DSP2.txt" in the software download.

9.7 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.