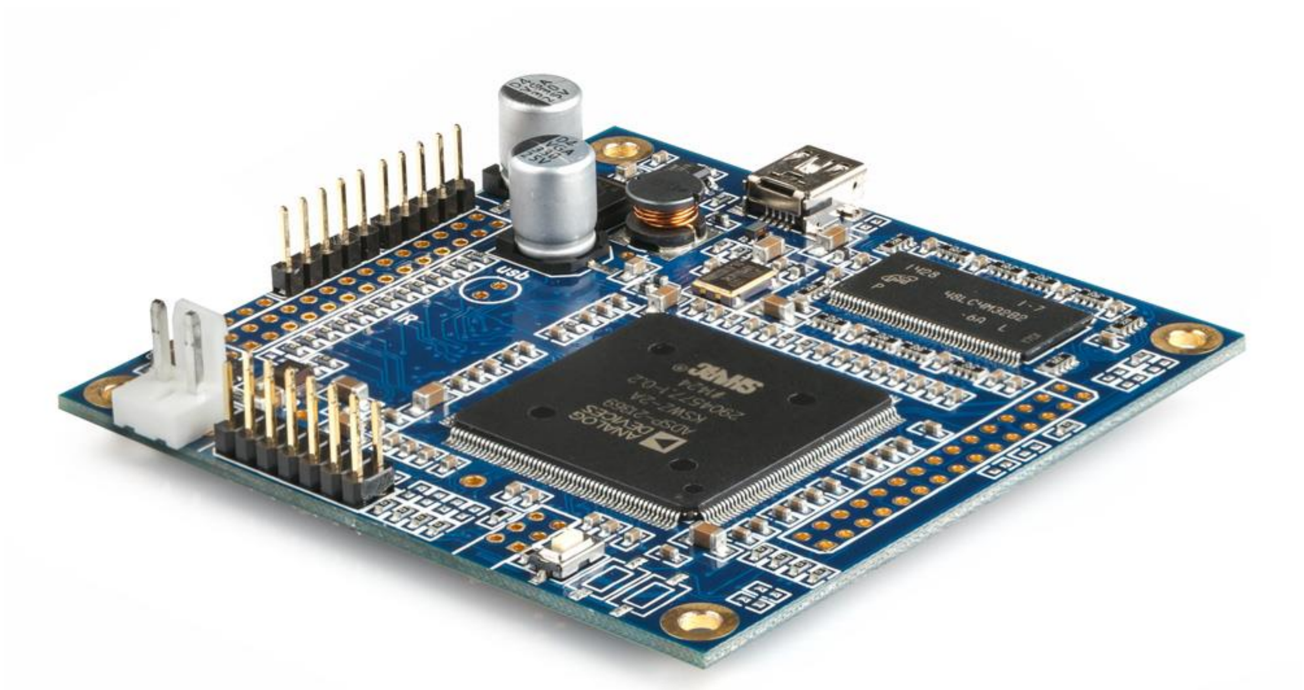
MINISHARC

COMPACT 4-IN 8-OUT AUDIO PROCESSOR BOARD WITH FIR CAPABILITY

User Manual



Revision history

Revision	Description	Date
V1.0	Initial revision	10-02-2013
V1.1	Added section for AN-FP connectivity	28-08-2013
V1.2	Added clarifications for routing of I2S/SPDIF	10-12-2013
V1.3	I2S slave configuration added	24-02-2014
V1.4	I2S master/slave clarifications added	17-03-2014
V1.5	Clarifications for SPDIF/I2S channel assignments	29-12-2014
V2.0	Revised full manual	31 July 2016

CONTENTS

IMPORTANT INFORMATION

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

SYSTEM REQUIREMENTS

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

Windows

- PC with 1GHz or higher processor clock speed. Intel® Pentium®/Celeron® family, or AMD K6®/AMD Athlon®/AMD Duron® family, or compatible processor recommended.
- 512 megabytes (MB) of RAM or higher
- Keyboard and mouse or compatible pointing device
- USB 2.0 port
- Microsoft• ® Windows® Vista® SP1/ XP pro SP2/Win7/Win8.1/Win10
- Microsoft• ® .NET framework v3.5 or later
- Adobe AIR environment (latest version)
- Adobe Flash player (latest version)

Mac OS X

- Intel-based Mac with 1 GHz or higher processor clock speed
- 512 megabytes (MB) of RAM or higher
- Keyboard and mouse or compatible pointing device
- USB 2.0 port
- Mac OS X 10.8 or higher
- Adobe AIR environment (latest version)
- Adobe Flash player (latest version)

DISCLAIMER/WARNING

miniDSP cannot be held responsible for any damage that may result from the improper use of this product or incorrect configuration of its settings. As with any other product, we recommend that you carefully read this manual and other technical notes to ensure that you fully understand how to operate this product. The miniDSP audio processor is a powerful tool, and misuse or misconfiguration, such as incorrectly set gains or excessive boost, can produce signals that may damage your audio system.

As a general guideline, you should perform the initial configuration of the miniDSP audio processor before enabling audio through any connected output device or amplification. Doing so will help ensure that the software is correctly configured.

Finally, note that the miniDSP audio processor is a very flexible device, and many of the questions we receive at the tech support department are already answered in this user manual and in the online application notes on the miniDSP.com website. So please take the time to carefully read this user manual and the online technical support. Thanks for your understanding!

Warranty Terms

miniDSP Ltd warrants this product to be free from defects in materials and workmanship for a period of one year from the invoice date. Our warranty does not cover failure of the product due to incorrect connection or installation, improper or undocumented use, unauthorized servicing, modification or alteration of the unit in any way, or any usage outside of that recommended in this manual. If in doubt, contact miniDSP prior to use.

FCC Class B Statement

This device complies with Part 15 of the FCC Rules. Operation is subject to the following two conditions:

- This device may not cause harmful interference.
- This device must accept any interference received, including interference that may cause undesired operation.

Warning: This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

Notice: Shielded interface cable must be used in order to comply with emission limits.

Notice: Changes or modification not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

CE Mark Statement

The miniSHARC has passed the test performed according to European Standard EN 55022 Class B.

A note on this manual

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

1 Product Overview

Thank you for choosing the miniSHARC audio processor board. The miniSHARC is an extremely compact yet powerful and flexible digital signal processing circuit board for audio applications. It provides audio input and output via I2S and S/PDIF (logic-level). Various accessory boards are provided for a rapid build or for prototyping, and the system integrator can implement their own boards for I2S input and output.

The audio processing functionality of the board is implemented via the use of a software *plugin*. Depending on the plugin and accessory boards used, the miniSHARC currently supports up to 4 input channels and 8 output channels. For more information on the available plugins, see Choosing a plugin on page 8.

The miniSHARC is a tried and proven hardware platform for audio processing applications. It is used as the processing core of several miniDSP products. For those seeking a ready-made “in the box” solution, please see our OpenDRC Series and Dirac Series.

1.1 The miniDSP concept

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

Hardware unit

In this case, the hardware unit is the miniDSP miniSHARC board. Various add-on boards can be purchased for the miniSHARC. You can also develop your own I2S input/output boards.

Software plugin

The software plugin is installed on your PC or Mac, and determines the processing that the DSP will perform. It provides a friendly user interface, and downloads instructions into the miniDSP hardware unit that tell it how to process the audio signal. See Choosing a plugin on page 8 for more information on plugins.

1.2 CHOOSING A PLUGIN

There are three plugins that operate with the miniSHARC board. These are summarized below.

Table 1. miniSHARC plugins

	miniSHARC 4x8	**miniSHARC 4x8 96k**	**OpenDRC 2x2**
Internal sample rate	48 kHz	96 kHz	48 kHz
Number of input channels	4	4	2
Number of output channels	8	8	2
Access to I2S input[1]	✓	✓	✓
Access to SPDIF input[2]	✓	✓	×
Access to I2S output[1]	✓	✓	✓
Access to SPDIF output[2,3]	✓	✓	×
Number of FIR taps (total)	9600	4300	12288
Minimum FIR taps per channel	6	6	6144[4]
Maximum FIR taps per channel	2048	2048	6144[4]
PEQ bands per input channel	10	10	6
PEQ bands per output channel	10	10	×
Compressor block	✓	✓	✓
Max per-channel time delay	3000 ms	1500 ms	3000 ms
Applications	Active crossover up to 4-way stereo or 8-way mono		Digital room correction, 2-way mono crossover

1. Requires suitable hardware/interface circuitry.
2. Requires level convertor for connection to external S/PDIF line.
3. Derived from I2S output channels 7 and 8.
4. Taps are not reassignable.

1.3 ORDERING PLUGINS

Plugins must be ordered in addition to the hardware:

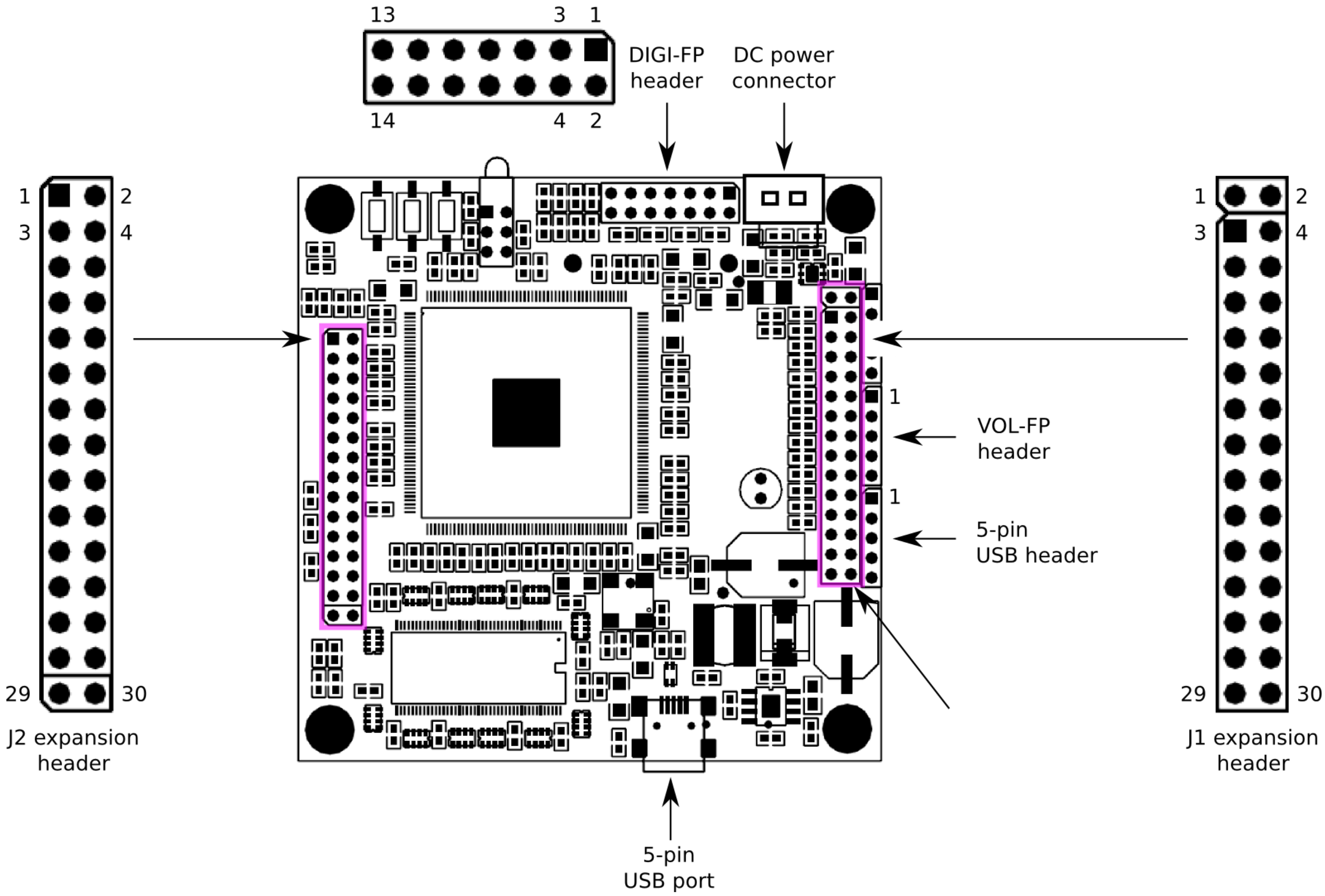
- miniSHARC plugins. Ordering from this page includes both the **miniSHARC 48k** and **miniSHARC 4x8 96k** plugins.
- OpenDRC 2x2 plugin. Ordering from this page includes just the **OpenDRC 2x2** plugin.

2 BOARD OVERVIEW AND CONNECTIVITY

This section describes the board and its interfaces.

2.1 BOARD LAYOUT

This diagram shows the layout of the miniSHARC board.



Note: the miniSHARC is delivered with two different header configurations. By default:

- If the miniSHARC is ordered alone, it will be delivered without the 30-pin headers for J1 and J2 soldered. This is so that you can solder your own headers or connections. Straight headers for USB, VOL-FP, and DIGI-FP will be soldered in.
- If the miniSHARC is ordered together with a miniDAC8, it will be delivered with straight 30-pin headers for J1 and J2 soldered. This is so that it interfaces directly to the miniDAC8 in the standard piggyback configuration (page 17). In addition, right-angle headers will be soldered in for the USB, VOL-FP and DIGI-FP headers.

If you require a header configuration other than the default, please put a clear request in the order form when you place the order.

2.2 DC Power

The miniSHARC accepts a regulated DC power supply on the 2-pin connector. A short cable from a 2.1 mm panel mount DC power socket is included with the kit purchase.

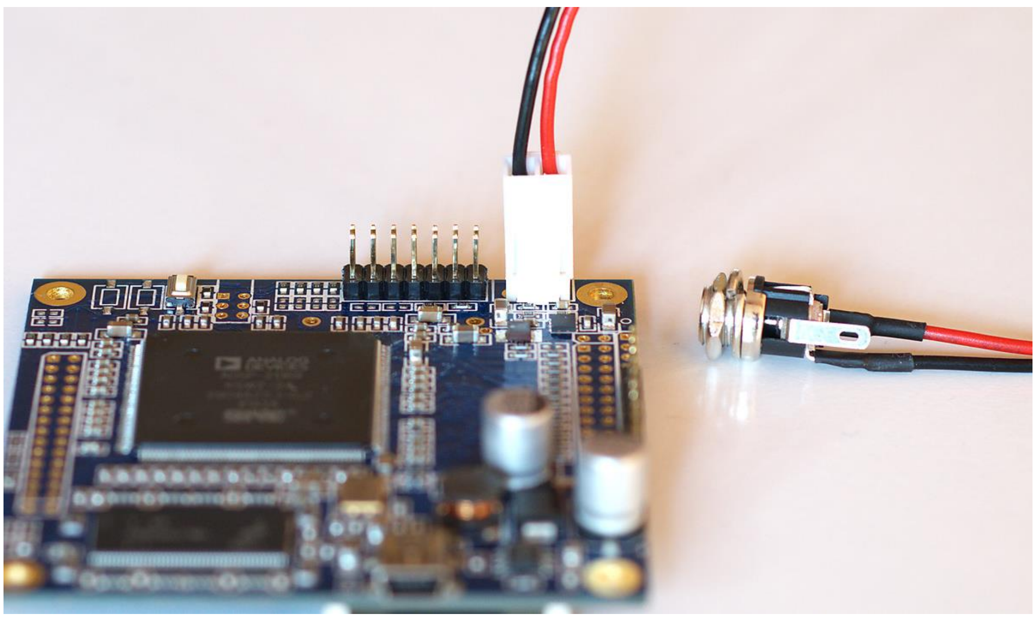
Standalone use

When the miniSHARC is not used with any of the miniDSP accessories that take power from the miniSHARC board, the miniSHARC can be powered with a DC supply in the range of 5 to 24 V.

Use with powered accessories

When the miniSHARC is used with any of the miniDSP accessories that take power from the miniSHARC board (VOL-FP, DIGI-FP, DA-FP, AN-FP, and miniDAC8), the miniSHARC **must** be powered from a regulated 5V supply **only**. *Any other supply voltage will cause malfunction and possible damage to boards that will not be covered under warranty.*

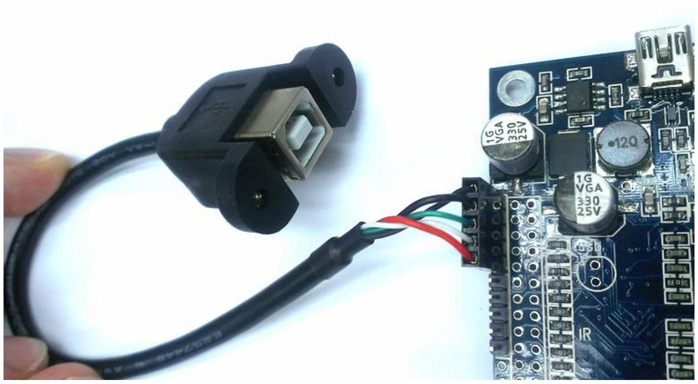
The connection to the DC power header is as shown below. Note that a slightly different wiring scheme is used when the AN-FP is used in conjunction with the miniSHARC – see page 16.



2.3 USB

The miniSHARC board has an onboard USB Mini-B socket. This can be connected directly to a computer with a standard Type A to Mini-B cable.

For installation into a chassis, the miniSHARC kit has a 5-pin header (standard 0.1"/2.54mm pitch). This can be connected to a panel-mount USB connector via a commonly available cable assembly.



Although this is a commonly available cable (typically used in computer hardware), you should nonetheless confirm the pin-out with the manufacturer, or you could easily cause damage to your computer or miniDSP kit. In most cases, pins 4 and 5 can be identified by a black wire, but once again, DO check the manufacturer's spec of the connector to prevent any short-circuits.

To make things simple, we provide a suitable cable for purchase in the accessories section of our website.

The miniDSP accessory boards DIGI-FP, AN-FP, and DA-FP incorporate a panel-mount USB Type B connector and include a cable to interface to the miniSHARC 5-pin header.

2.4 J1 EXPANSION HEADER

The user-accessible pins in the J1 header are listed in Table 2 below. Only a small number of pins are intended to be user-accessible. Please do not connect to any pins marked "—".

Table 2. J1 expansion header pinout

Pin	Function	Pin	Function
1	—	2	—
3	GND	4	GND
5	GND	6	GND
7	—	8	I2S input slave select (pull low)
9	I2S output slave select (pull low)	10	—
11	—	12	—
13	—	14	—
15	—	16	—
17	—	18	—
19	—	20	—
21	—	22	—
23	—	24	—
25	—	26	—
27	—	28	—
29	—	30	—

2.5 J2 EXPANSION HEADER

The user-accessible pins in the J2 header are listed in Table 3 below. This header contains all of the I2S input and output signals.

Table 3. J2 expansion header pinout

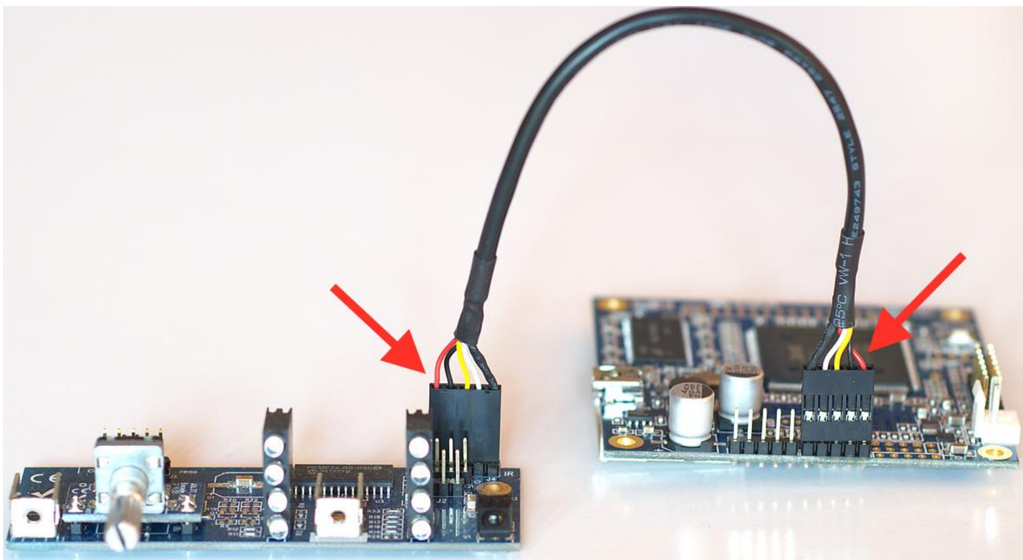
Pin	Function	Pin	Function
1	SPDIF RX (3.3V compliant)	2	SPDIF TX (3.3V compliant)
3	LRCLK_SPDIF (Not used in current firmware)	4	BCLK_SPDIF (Not used in current firmware)
5	GND	6	GND
7	GND	8	GND
9	MCLK	10	I2S_DATA_IN1&2
11	I2S_DATA_IN3&4	12	I2S_DATA_IN5&6
13	I2S_DATA_IN7&8	14	I2S_IN_LRCLK
15	I2S_IN_BCLK	16	I2S_DATA_OUT1&2
17	I2S_DATA_OUT3&4	18	I2S_DATA_OUT5&6
19	I2S_DATA_OUT7&8	20	I2S_OUT_LRCLK
21	I2S_OUT_BCLK	22	3.3V
23	GND	24	3.3V
25	GND	26	GND
27	NC	28	NC
29	NC	30	NC

3 OFF-THE-SHELF ACCESSORIES AND CONNECTION

3.1 VOL-FP

VOL-FP is a small board intended for mounting on the front panel of an enclosure. It contains a rotary encoder, infrared receiver, and status LEDs. It provides volume control, preset selection, and digital input selection (applies when used together with DIGI-FP or DA-FP).

After mounting in the enclosure, connect the supplied 5-pin cable between the VOL-FP board and the miniSHARC board as shown in the photograph below. Note carefully the orientation of the connectors as shown in the photograph.



3.1.1 Operation of VOL-FP

To change the volume

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

To change the selected source

Briefly press the control button. The **Digital Source** LED blinks quickly. Rotate the control knob until the desired source LED is blinking. Press the control knob again, and the selected LED will remain steady. The preset LED will now blink.

To change the selected preset

After changing source selection, rotate the control knob until the desired preset LED is blinking. Press the control knob again, and the selected LED will remain steady.

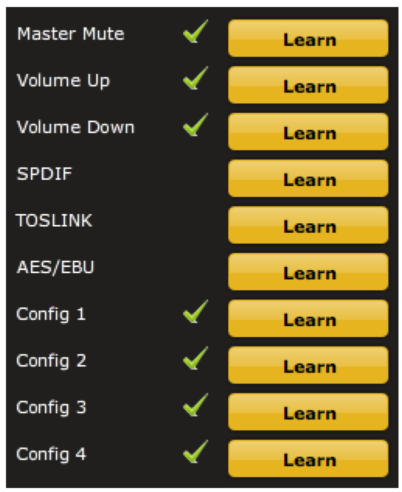
3.1.2 Infrared remote control

The miniSHARC plugins can “learn” the control codes of your current remote if it supports one of the following remote control codes:

- NEC
- Sony
- Philips RC6

To initiate the learning process, drop down the IR Remote menu and select **IR learning**. Click on the **Learn** button for an operation, and then press the desired button on the remote control. If the code is accepted, the status will change to show a tick.

This screenshot shows the IR learning screen:



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

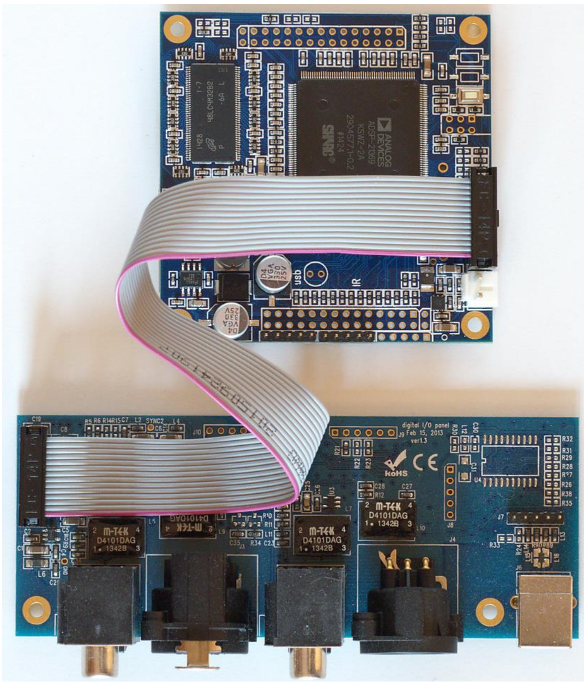
3.2 DIGI-FP

DIGI-FP is an audio interface board intended for panel mounting that provides stereo digital input and output in three formats:

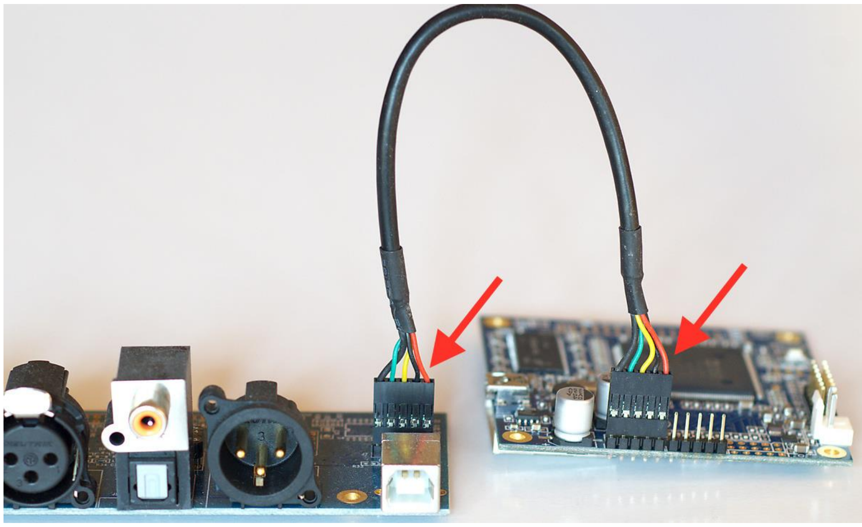
- S/PDIF (RCA)
- Optical (TOSLINK)
- AES/EBU (XLR)

The DIGI-FP contains an on-board asynchronous sample rate convertor (ASRC) that converts the input signal to the sample rate set by the miniSHARC plugin (48 or 96 kHz). In addition, it provides a USB Type B socket.

The DIGI-FP requires that two cables (supplied with the DIGI-FP) be connected to the miniSHARC. A 14-pin flat ribbon cable carries power, I2S and control signals. Connect it as shown in the photograph at right.



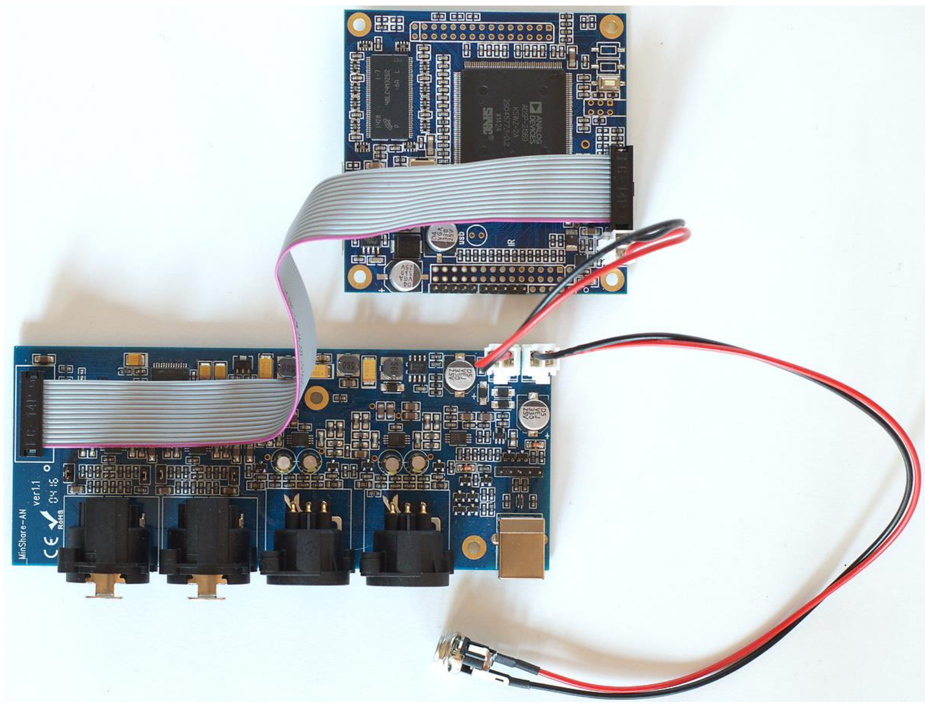
A 5-pin cable carries USB between the miniSHARC and the panel-mount USB connector. Connect it as shown in the photograph at right, taking careful note of the orientation of the connector as indicated by the arrows.



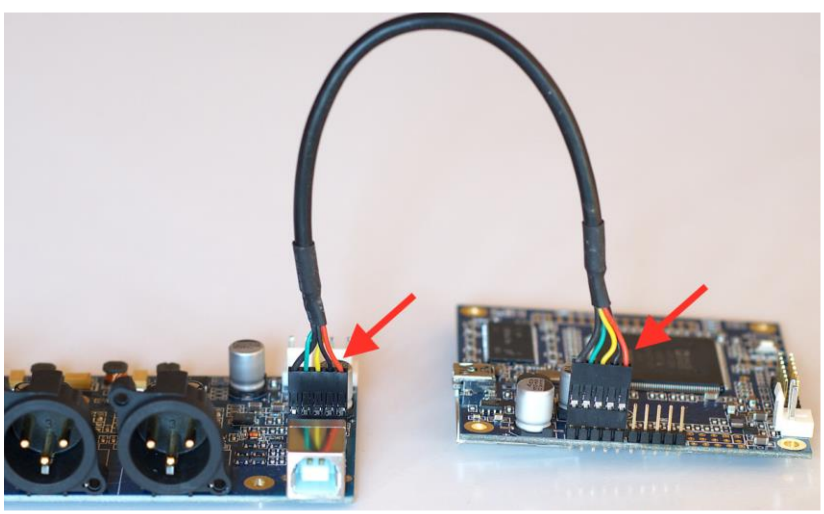
3.3 AN-FP AND DA-FP

AN-FP is an audio interface board that provides stereo balanced analog I/O on Neutrik panel-mount connectors. DA-FP provides the same set of digital inputs as the DIGI-FP (previous page), and stereo balanced analog output on Neutrik panel-mount connectors. Both boards also provide a USB Type B socket.

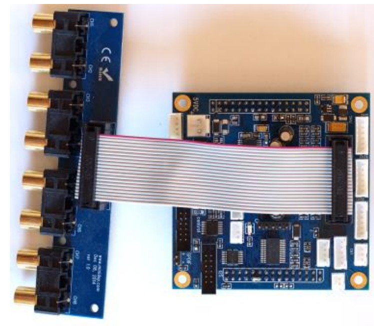
The AN-FP and DA-FP require that three cables (supplied with the board) be connected to the miniSHARC. A 14-pin flat ribbon cable carries I2S and control signals. The AN-FP and DA-FP also require separate 5V power. To facilitate a simple wiring scheme, the AN-FP and DA-FP provide two power connectors, which can be used as shown in the photograph below.



A 5-pin cable carries USB between the miniSHARC and the panel-mount USB connector. Connect it as shown in the photograph below, taking careful note of the orientation of the connector as indicated by the arrows.

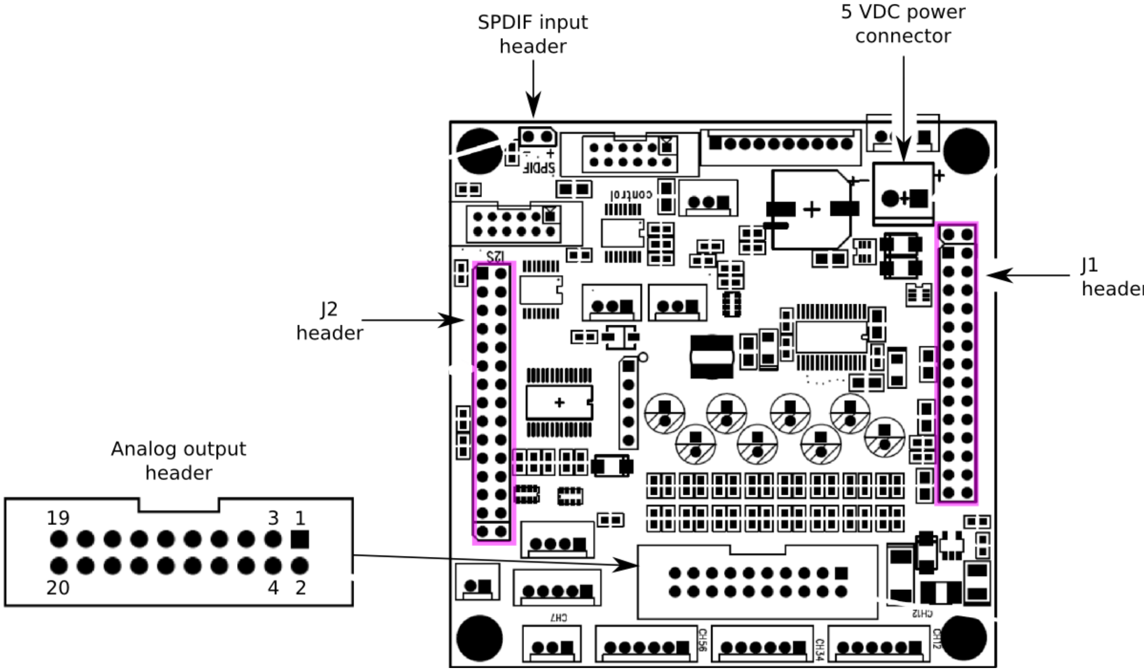


3.4 MINIDAC8

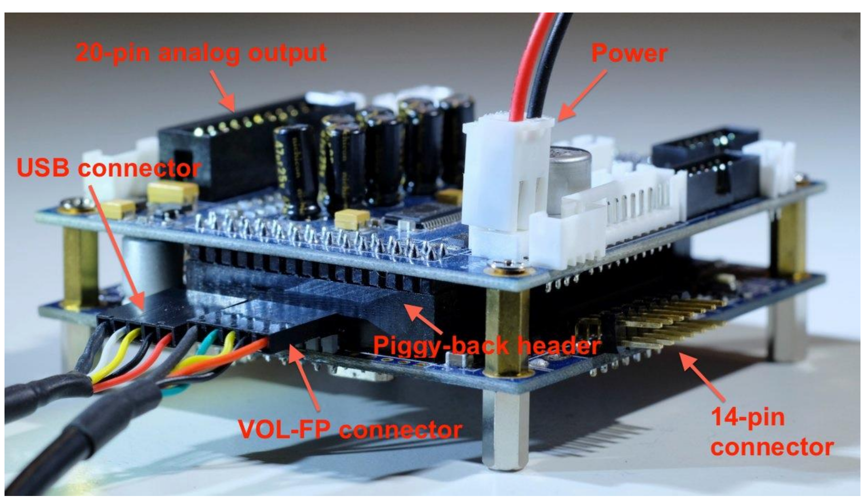


The miniDAC8 is an 8-channel DAC. It is supplied as a daughterboard that "piggybacks" onto the miniSHARC, and a separate panel-mount board carrying eight RCA jacks. It also contains an SPDIF input level shifter and connection point. The 5 VDC power connection to this board powers the miniSHARC.

To mount the miniDAC8 onto the miniSHARC, the miniSHARC must have straight 30-pin headers installed for both J1 and J2. The USB and VOL-FP headers must be right-angle types to ensure adequate clearance between the boards.



This photograph shows the key connectivity points:



Note: one particular usage of the miniDAC8 requires a custom firmware – see page 22 for full details.

4 I/O INTERFACES

Interfacing to the miniSHARC is accomplished through I2s, also known as Inter IC Sound, an electrical serial bus used to interface digital audio devices at the chip and circuit board level. The miniSHARC circuit board provides audio input and output via I2S pins on the J2 30-pin header. Prebuilt SPDIF interfaces are also provided on J2.

4.1 I2S OVERVIEW

An I2S interface consists of up to three clocks, and a data line for each pair of channels.

There are three types of clock:

MCLK The master clock that the miniSHARC uses internally. This clock is always provided as an output by the miniSHARC, and connected circuitry can choose whether or not to use it.

LRCLK The frame synchronization clock, also known as the word clock. This clock is equal to the sampling frequency (Fs) of the audio signal.

BCLK The bit clock (also known as shift clock or system clock). This is always equal to 64 x Fs.

Table 4 summarizes the relation between the clocks for the 48 and 96 kHz plugins.

Table 4. I2S clock ratios

Plugin sample rate (LRCLK)	Master clock (MCLK)	Bit clock (BCLK)	MCLK/LRCLK	BCLK/LRCLK
48 kHz	24.576 MHz	3.072 MHz	512	64
96 kHz	24.576 MHz	6.144 MHz	256	64

The timing of data lines is determined by the bit clock and the word clock, as illustrated in the following diagram:

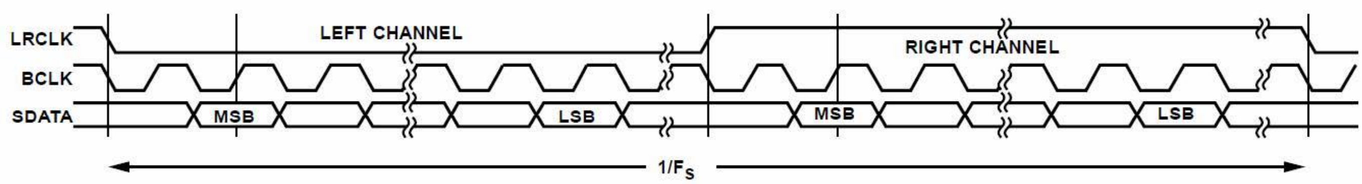


Figure 31. I²S Mode—16 Bits to 24 Bits per Channel

The miniSHARC board has four input I2S data lines and four I2S output data lines. Note, however, that the input data lines for channels 5 though 8 are not accessible in the current firmware and plugins. The I2S data line for channels 3 and 4 is used by the SPDIF input, and is accessible by the miniSHARC 4x8 plugins (both 48 kHz and 96 kHz versions).

Together, the word clock and bit clock are referred as a clock *domain*. The miniSHARC board has three clock domains:

- I2S input
- I2S output
- SPDIF input

Depending on the I2S clock mode, the clocks may be provided by the miniSHARC or external circuitry, and the data line may or may not be subject to sample rate conversion.

4.2 I2S USAGE NOTES

Note that I2S is not a "plug and play" protocol. It requires attention to technical details such as clocking and wire layout. It is a solution for OEMs and *advanced* DIYers (or professionals) with suitable knowledge, skills and measurement equipment. For off-the-shelf completed units utilizing the miniSHARC board, please refer to the miniDSP OpenDRC Series and the Dirac Series.

Be sure to take the following precautions when designing your I2S interface and wiring:

General I2S usage notes

- Unbuffered I2S lines must be kept short to ensure clock and data integrity.
- If driving longer lines, buffers may be required for the clock signals (MCLK, LRCLK, and BLCK).
- Observe correct grounding and shielding, and keep analog and digital grounds separated.
- Ensure that the clock ratios (as listed in Table 2) are compatible with connected circuits.

3.3V logic level

All lines use a 3.3V logic level. Ensure that connected circuits use a compatible level (1.8V, for example, will not work).

4.3 I2S CLOCK MODES

The miniSHARC can act in either **master** or **slave** mode with respect to I2S clocking. Master/slave mode can be set independently for I2S input and output.

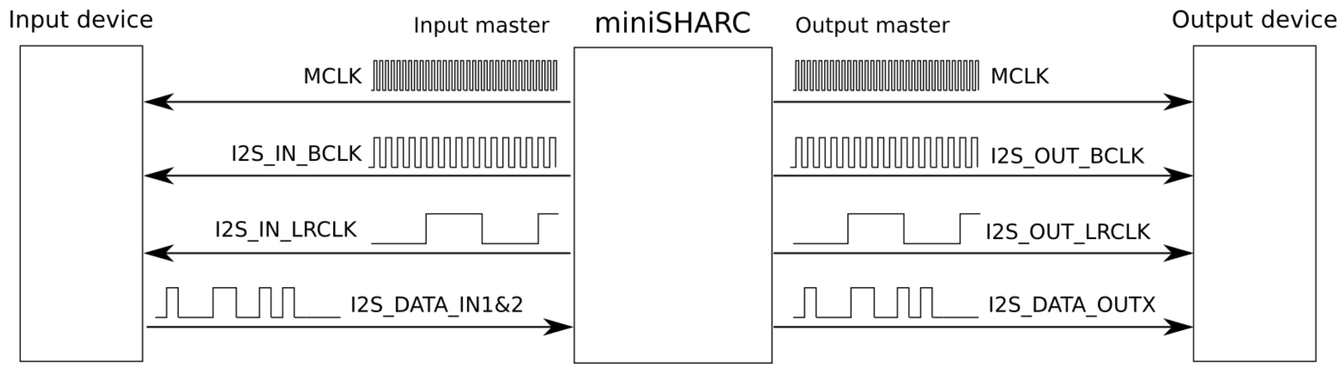
In master mode (for either input or output), the miniSHARC provides the I2S clocks, and the connected device is expected to transmit or receive the data signal using those clocks. The clocks will match either a 48 kHz or 96 kHz sample rate, depending on the plugin loaded onto the miniSHARC.

In slave mode, the connected device provides the I2S clocks, and the connected device is assumed to be providing or consuming the data signal using those clocks.

4.3.1 Master mode

This is the default mode for the miniSHARC. The miniSHARC provides all clocks. The connected input and output devices must therefore provide or consume data at the rates defined by those clocks. A typical connection scenario is shown in Figure 1.

1. The connected devices may or may not use MCLK. This is dependent on the specifics of the devices.
2. The miniSHARC must be in master mode to use any of the off-the-shelf peripheral boards AN-FP, DIGI-FP, and DA-FP. The DIGI-FP and DA-FP have an onboard asynchronous sample rate convertor to convert the incoming sample rate to the clock domain provided by the miniSHARC.



Note 1: Clock ratios are not accurate, waveforms are for illustration only

Note 2: I2S_IN_BLCK == I2S_OUT_BLCK, and I2S_IN_LRCLK == I2S_OUT_LRCLK

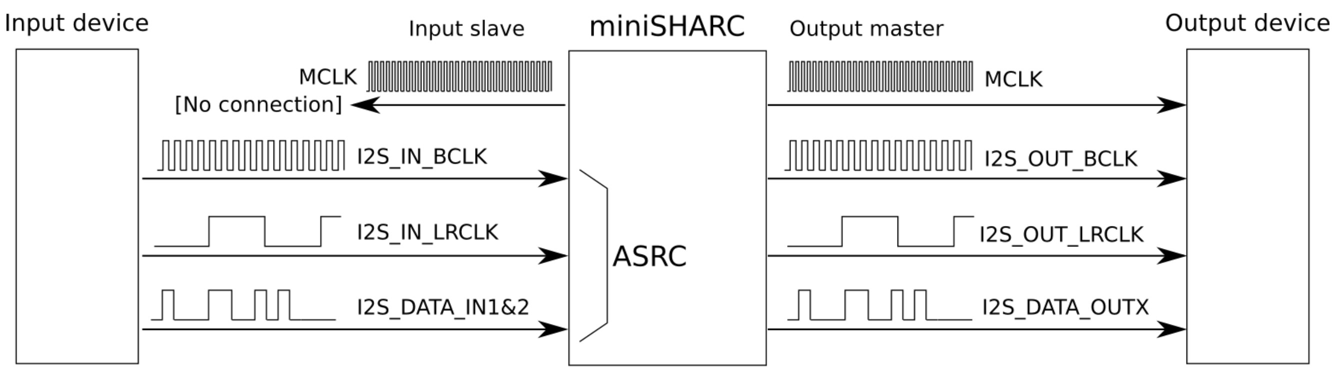
Figure 1. Master mode

4.3.2 Input slave mode

This mode is used when the input device provides its own clock. One example is the miniDSP miniStreamer and USBStreamer products, which provide clocks determined by the sample rate of the audio stream received over USB.

When the miniSHARC is put into input slave mode, I2S_IN_BCLK and I2S_IN_LRCLK become inputs. The miniSHARC clocks the I2S_DATA_IN1&2 signal according to the clocks provided on I2S_IN_BCLK and I2S_IN_LRCLK, and applies asynchronous sample rate conversion to convert the data stream to its internal clock domain (48 or 96 kHz). The connections are shown in Figure 2 below.

To put the miniSHARC into input slave mode, pull pin 8 of the J1 expansion header low. This is easily accomplished by connecting pin 8 to pin 6 (GND).



Note 1: Clock ratios are not accurate, waveforms are for illustration only

Note 2: ASRC converts from input clock domain to miniSHARC clock domain

Figure 2. Input slave mode

4.3.3 Output slave mode

Output slave mode is used when the miniSHARC is connected to external circuitry that:

a. Acts as both I2S data source and receiver, and

b. Generates its own clocks.

In this mode, the connected device must provide a bit clock and word clock at the sample rate of the loaded plugin – that is, 48 or 96 kHz. The clocks must be provided to both sets of clock lines, as illustrated in Figure 3 below.

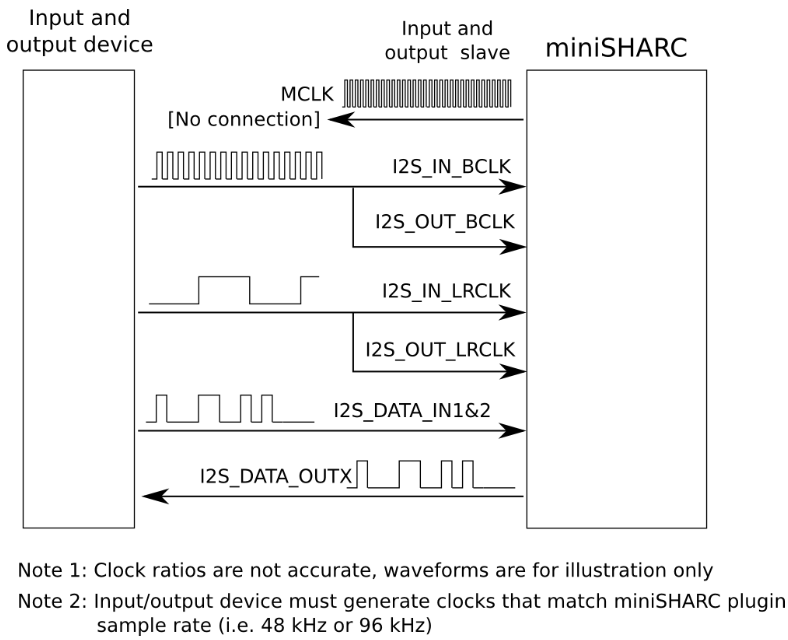


Figure 3. Output slave mode

To put the miniSHARC into output slave mode, pull both pin 8 and pin 9 of the J1 expansion header low. This is easily accomplished by connecting them to pin 6 and/or pin 5 (GND).

4.3.4 Summary of I2S clock modes

Table 5 summarizes the various I2S clock modes. The table indicates the source of each clock domain for the mode, and the settings of the relevant J1 pins.

Note that there is a fourth mode, in which J1 Pin 8 is N/C and J1 Pin 9 is pulled low. This mode does not have any practical application so is not supported.

Table 5. Summary of modes and clock domains

Mode	Input clock domain	Output clock domain	J1 Pin 8	J1 Pin 9
Master	miniSHARC 48/96k	miniSHARC 48/96k	N/C	N/C
Input slave	Input device 32–192k	miniSHARC 48/96k	Pull low	N/C
Output slave	I/O device 48/96k	I/O device 48/96k	Pull low	Pull low

4.4 Usage with miniDAC8 in I2S input slave mode

In the particular case where the miniSHARC uses the miniDAC8 as output device **and** the miniSHARC is to be run in I2S input slave mode, special rules apply. In that case (and **only** that case), **all** of the following are required:

1. Pin 8 of J1 is pulled low.
2. A custom firmware version is loaded onto the miniSHARC.
3. Input connection is made to the DIGI-FP header instead of to J2.

Pull J1 pin low

Tie Pin 8 of J1 to Pi 6 (GND). Because the J1 header is no longer accessible once the miniDAC8 is mounted, this is best accomplished by soldering a small jumper between the pins on the rear of the board.

Custom DSP firmware

Because of the way that the miniSHARC uses the J1 header, a custom DSP firmware needs to be loaded.

1. Download the custom DSP firmware from the User Downloads section of the minidsp.com website, in the section OpenDRC/miniSHARC plugins and with the heading “MiniSHARC 4x8 DSPFW I2S CLK.”
2. Unzip the download, and follow the instructions in the file “readme.txt.”

Input connection

Since the J2 header has the miniDAC8 on it, an alternate input path to the DIGI-FP header has been provided. Figure 4 shows the pin numbers of the DIGI-FP header to connect to.

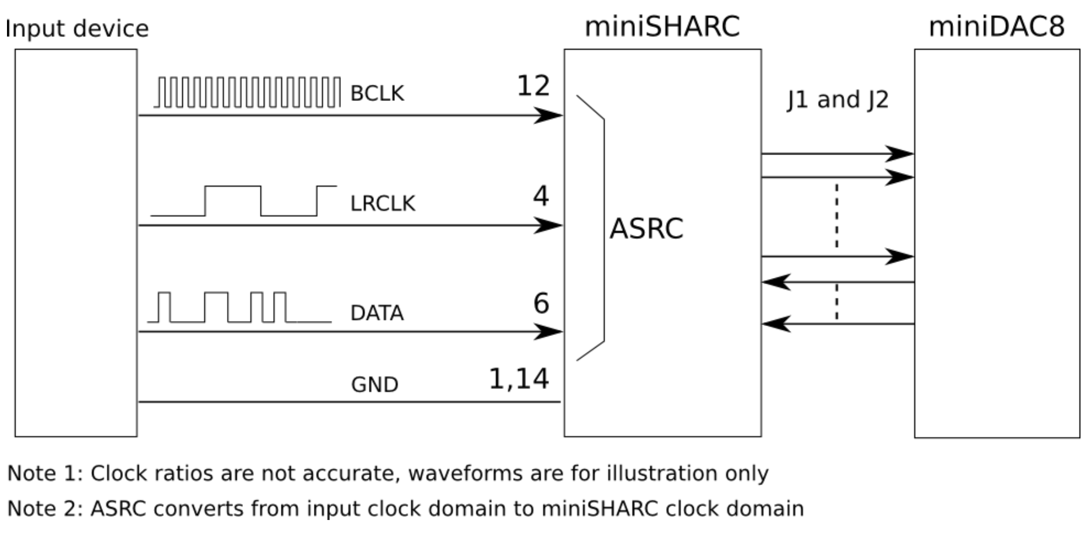


*Figure 4. Input connections to DIGI-FP header, input slave mode with miniDAC8 **only***

Note: do not use this connection method for any case other than input slave mode with miniDAC8 as output device. Malfunction and possible damage to the board may result.

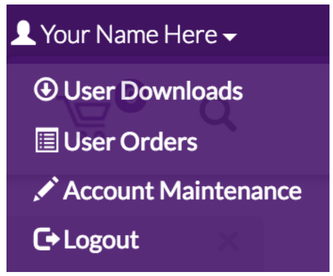
4.5 S/PDIF CONNECTIVITY

Pins 1 and 2 of J2 are S/PDIF input and output signal lines respectively.

- These pins cannot be connected directly to S/PDIF input or output connectors. They use a 3.3V logic level, in contrast to the S/PDIF level of 0.5 – 0.6 V (peak to peak). A suitable level convertor circuit must be implemented to connect to these pins.
- Pin 1, S/PDIF input, produces the I2S data line I2S_DATA_IN3&4. This data line is accessible in the **miniSHARC_4x8** and **miniSHARC_4x8_96k** plugins as the "SPDIF" input. It is not accessible in the **OpenDRC2x2** plugin.
- I2S_DATA_IN3&4 always has asynchronous sample rate conversion applied, from the SPDIF clock domain to the miniSHARC's clock domain, regardless of the I2S clock mode. The SPDIF input can therefore be driven at any sample rate from 32 to 192 kHz.
- Pin 2, S/PDIF output, is derived from I2S data line I2S_DATA_OUT7&8. This data line is driven as output channels 7 and 8 in the **miniSHARC_4x8** and **miniSHARC_4x8_96k** plugins. It cannot be driven by the **OpenDRC2x2** plugin.

5 PLUGIN INSTALLATION

When your order ships, your ordered plugin (or plugins) will be available from the User Downloads section of the miniDSP website. You will need to be logged into the website with the account you created when purchasing. The User Downloads link is visible from the dropdown menu at the top right of the website page:



Navigate to the **OpenDRC Plug-ins** section of User Downloads, then download the zip file (or files) under the heading **miniSHARC 4x8 48kHz**, **miniSHARC 4x8 96k**, or **OpenDRC 2x2** Double-click on it to unzip it. Then follow the installation procedure below according to your computer type.

Note: the Adobe Air framework may need a network connection the first time the plugin is used. If the plugin does not start properly, see Troubleshooting.

5.1 WINDOWS

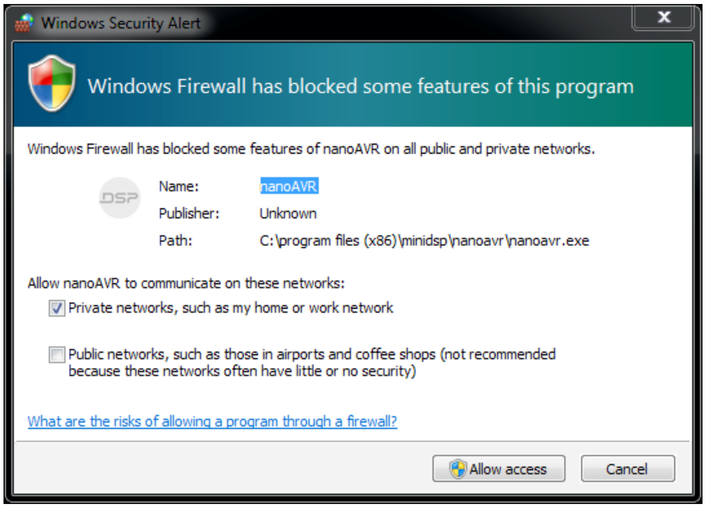
Prior to installing the miniDSP software, download and install the following programs. You will need to accept the license agreements in order to successfully complete the installation.

- Microsoft .NET framework (version 3.5 or later)
- Latest version of Adobe Flash
- Latest version of Adobe Air

If you haven't updated these recently, you should download and install the latest versions prior to running the miniDSP install program.

To install the miniDSP software, open the **Windows** folder of the download and double-click on the installation program: **miniSHARC_4x8.exe**, **miniSHARC_4x8_96k.exe**, or **OpenDRC_2x2.exe**. We recommend accepting the default installation settings.

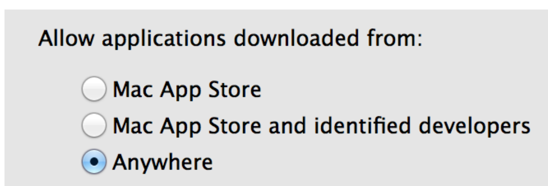
Once installation is complete, the plugin will automatically start. Since the plugin checks for a network connection when starting up, a warning such as the following may appear. In this case, click on **Allow access**.



5.2 Mac OS X

On versions of OS X from 10.7 (Lion) and later, you will need to inform the GateKeeper program that it is OK to install and run this software. Go to System Preferences, then click on Security & Privacy and select the General tab:

1. Click on the padlock icon in the lower left corner and enter your password, in order that you can make changes to the settings.
2. Under the text “Allow Applications downloaded from:”, click on “Anywhere.”



Then, download and install the following programs. You will need to accept the license agreements in order to successfully complete the installation:

- Latest version of Adobe Flash
- Latest version of Adobe Air

If you haven’t updated these recently, you should download and install the latest versions prior to running the miniDSP install program.

To install the miniDSP software, open the **Mac** folder of the download, and double-click on the disk image (“.dmg”) file to open it in a new window. Then double-click on the installer program, **Install miniSHARC 4x8.app**, **Install miniSHARC 4x8 96k.app**, or **Install OpenDRC-2x2.app**. Once installation is complete, the plugin will automatically start.

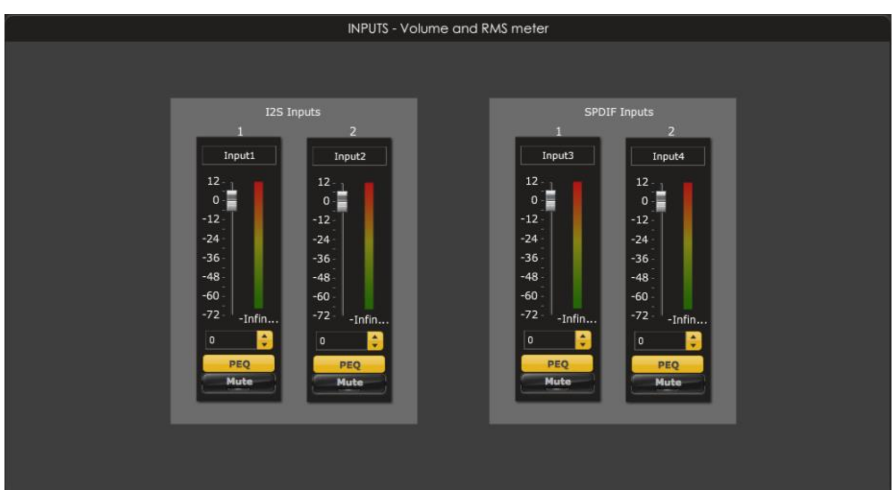
6 PLUGIN ARCHITECTURE

6.1 MINISHARC 4X8 PLUGINS

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

6.1.1 Input tab

The **Input** tab displays a row of input channel control strips: two for I2S input channels 1 and 2, and two for the SPDIF inputs (I2S channels 3 and 4). The input controls are described on page 29.



6.1.2 Routing tab

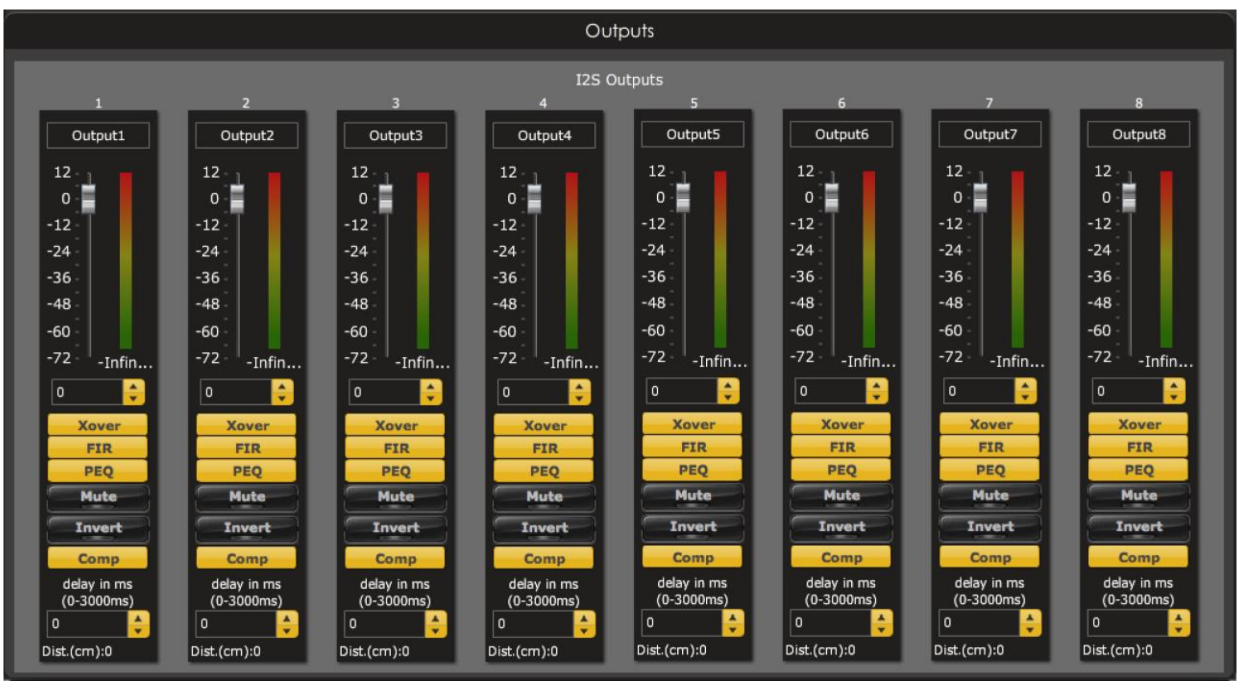
The **Routing** tab displays the matrix mixer, which sets up the routing from input channels to output channels. The input channels are labeled along the left, and the output channels along the top. Each input-output channel assignment is turned on and off by clicking the button corresponding to the two channels.

The labels displayed for the input and output channels can be changed on the **Input** and **Output** tabs. Note that both **I2S In** (channels 1 and 2) and **SPDIF In** (I2S channels 3 and 4) are active simultaneously.

	I2S Out							
I2S In	Output1	Output2	Output3	Output4	Output5	Output6	Output7	Output8
Input1	On	On	On	On	Off	Off	Off	Off
Input2	Off	Off	Off	Off	On	On	On	On
SPDIF In	Output1	Output2	Output3	Output4	Output5	Output6	Output7	Output8
Input3	On	On	On	On	Off	Off	Off	Off
Input4	Off	Off	Off	Off	On	On	On	On

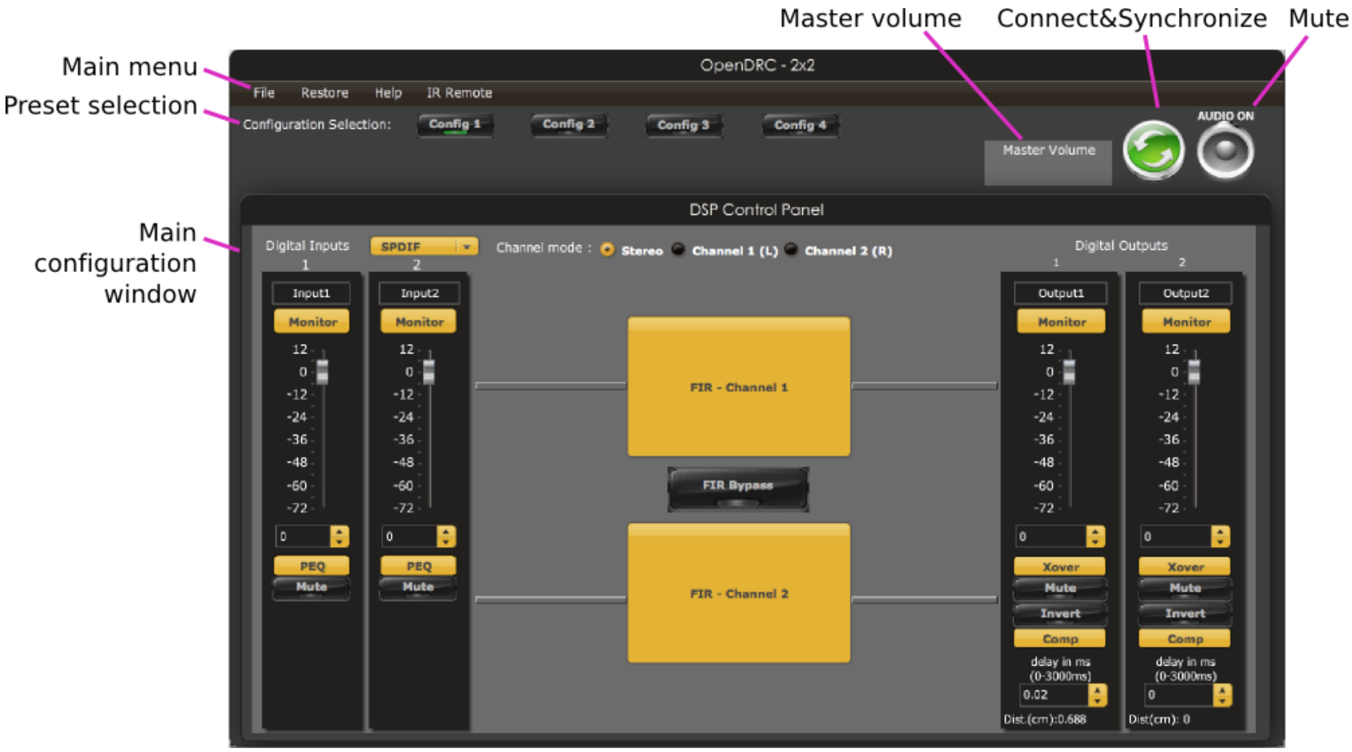
6.1.3 Output tab

The **Output** tab displays a row of output channel control strips. All channels are identical. The output controls are described on page 30 and in Section 7.



6.2 OpenDRC-2x2 plugin

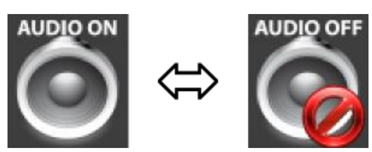
This annotated screenshot shows the main areas of the OpenDRC 2x2 plugin:



The two input channels are as provided in I2S data 1&2, and the output channels are as produced on I2S data line 1&2. (The S/PDIF input and output are not accessible from this plugin.) The input and output controls are described on pages 29 and 30. The FIR filter blocks are described on page 39.

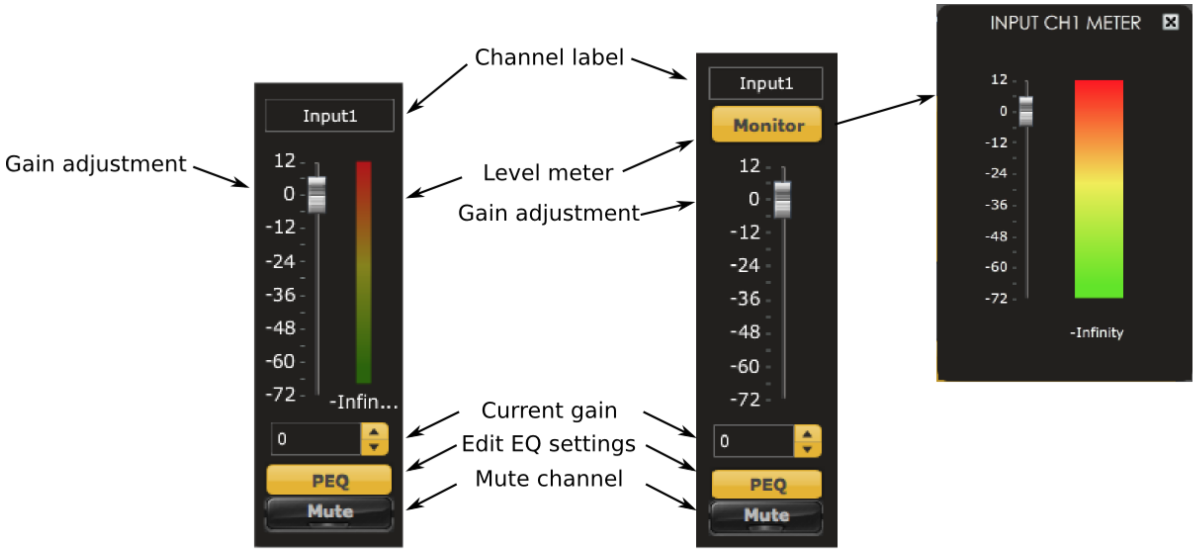
6.3 Common features

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



The Master volume display shows the current volume setting:

6.4 INPUT CHANNEL STRIPS



miniSHARC 4x8 OpenDRC 2x2

Channel label

Each input channel has a customizable label, which is shown at the top of the channel strip. This label also appears on the **Routing** tab (in the miniSHARC plugins). To change the label, click on it, type a new label (up to eight characters), and press the Return key.

Level meter

Displays the current signal level in real time. In the miniSHARC 4x8 plugins, the meter is displayed alongside the gain adjustment. In the OpenDRC 2x2 plugin, the Monitor button brings up an overlay that displays the level meter. The plugin must be in online mode to display signal levels.

Gain adjustment

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

PEQ settings

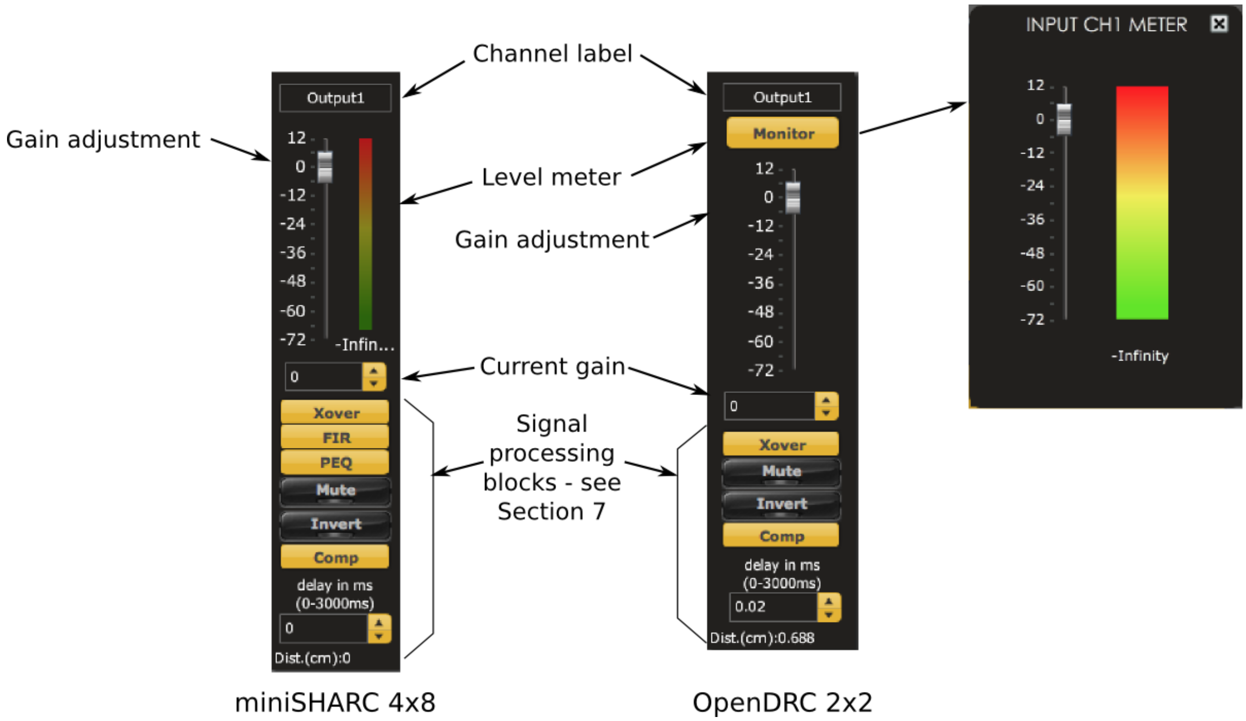
Click on this button to open the parametric EQ settings window for that channel. The miniSHARC 4x10 plugins have ten parametric EQ filters on each input channel; the OpenDRC 2x2 plugin has six parametric EQ filters on each input channel. For more details, see Parametric EQ on page 37.

Mute

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

6.5 OUTPUT CHANNEL STRIPS

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



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 (in the miniSHARC 48 plugins). To change the label, click on it, type a new label (up to eight characters), and press the Return key.

Level meter

Displays the current signal level in real time. In the miniSHARC 4x8 plugins, the meter is displayed alongside the gain adjustment. In the OpenDRC 2x2 plugin, the Monitor button brings up an overlay that displays the level meter. The plugin must be in online mode to display signal levels.

Gain adjustment

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

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.

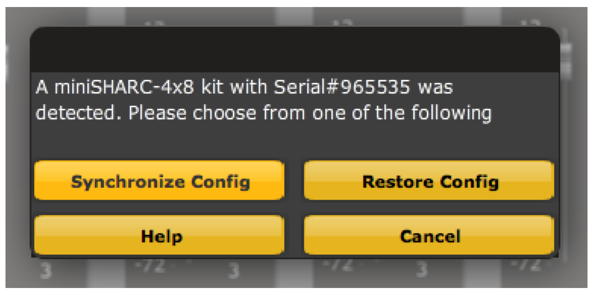
The bottom section of the control strip is used to access the main signal processing blocks of each output channel. Refer to Section 7 for full details.

6.6 SYNCHRONIZING WITH THE PROCESSOR

Communication with the processor takes place over a USB connection. Note that USB connection or to on the miniSHARC board is used for control purposes only. Audio data cannot be streamed to the processor over USB, except by using an external USB-to-I2S interface such as the miniStreamer or USBStreamer.

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

The first time you connect, or if you have made any changes to any data in the user interface, the following dialog box will appear:



The options are:

Synchronize Config

Download the currently selected configuration into the corresponding configuration preset of the processor. After downloading the configuration data, the plugin is in *online* mode and any changes to processing parameters will be downloaded immediately in real time. That is, the user interface is now “live.”

Synchronize and Upgrade

This is similar to Synchronize Config, but also upgrades the internal data of the processor. This option may appear after downloading and installing an updated version of the plugin.

Restore Config

Restore the data in the currently selected configuration to the factory defaults. When using this option, any connected equipment should be muted or powered off until you have set the configuration to a working state. Note that the configuration data will be lost, so if needed, ensure that you have saved the configuration to a file prior to using this option.

Cancel

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

6.7 Working with configurations

The data that controls the audio processing is called a *configuration*. The processor stores four configuration presets in its internal memory, which can then be selected from the front panel or via remote control.

6.7.1 Online and offline mode

Initially, the plugin is in *offline* mode. When the **Connect&Synchronize** button is used, the plugin downloads configuration data into the processor and goes into *online* mode. Changes made in the plugin user interface therefore fall into two categories:

The plugin is in online mode

The plugin user interface is "live" – that is, any changes made to the audio processing parameters in the user interface are immediately downloaded to the processor. The effect of these changes will thus be audible as the changes are made.

The plugin is in offline mode

Changes made to audio processing parameters in the plugin user interface will be made locally only. The next time the plugin is synchronized to the processor, the parameters will be downloaded to the processor (as long as the **Synchronize Config** button is selected).

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

6.7.2 Selecting a configuration

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

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

The plugin is in online mode

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

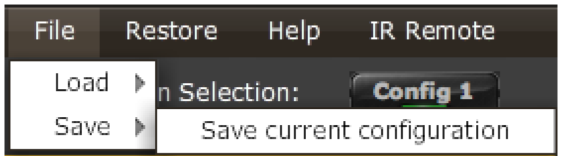
The plugin is in offline mode

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

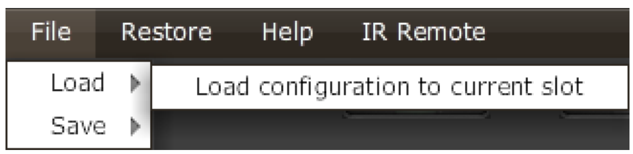
6.7.3 Saving and loading configurations

Configurations can be saved to and loaded from files. Each configuration is stored in a separate file. It is strongly recommended that each configuration programmed into the processor be saved to a file, to ensure that the configuration is not lost if the processor is inadvertently reset.

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



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



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

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

6.7.4 Restoring to defaults

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

Factory Default

Reset all four configuration presets to the factory default settings.

Current Configuration Only

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

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

6.8 KEYBOARD SHORTCUTS

The **miniSHARC 4x8** plugin supports the use of the keyboard for many operations.

Tab

The Tab key moves the focus from the current user interface element to the next. A blue-grey surrounding box usually indicates the user interface element with the focus. Shift-Tab moves the focus in the opposite direction.

Up/down arrows

The up/down arrow keys (and in some cases, the left/right arrow keys) adjust the value of many parameters, if they have the focus:

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

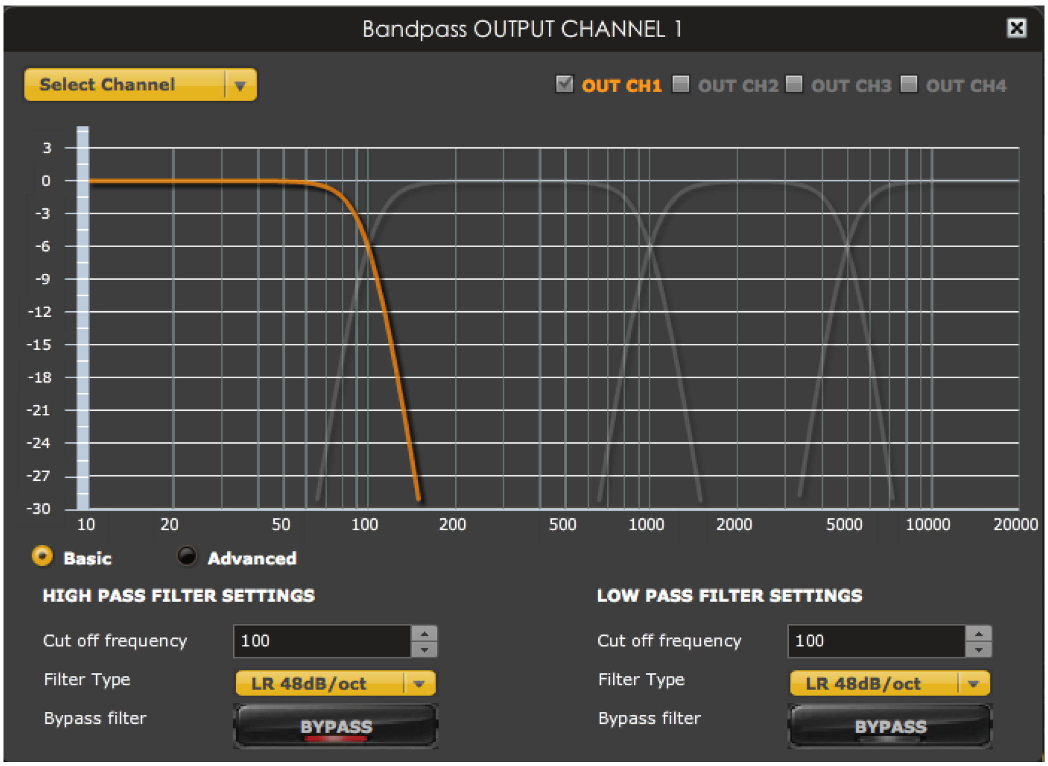
Space

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

7 Signal Processing Blocks

7.1 Crossover

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

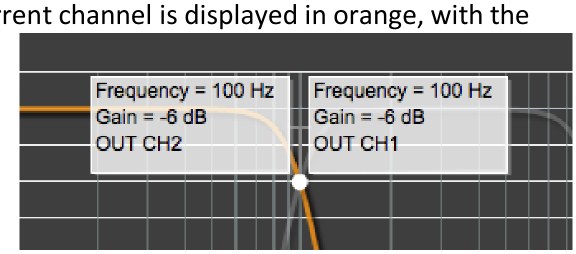


Crossovers "split" the frequency band to send to different drivers. In a two-way loudspeaker, for example, a *low pass* filter is used to remove high frequencies from the signal sent to the woofer, and a *high pass* filter is used to remove low frequencies from the signal sent to the tweeter. In a three-way speaker, the midrange driver will use both the high pass and low pass filters. Crossover filters can also be used to limit low frequency content delivered to a speaker or subwoofer, to help protect it from over-excursion.

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

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

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



Basic/Advanced

By default, the crossover is in basic mode, and shows the controls described below. Advanced mode enables custom biquad programming for almost infinite flexibility in crossover filter implementation. This is described in Custom biquad programming on page 43.

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

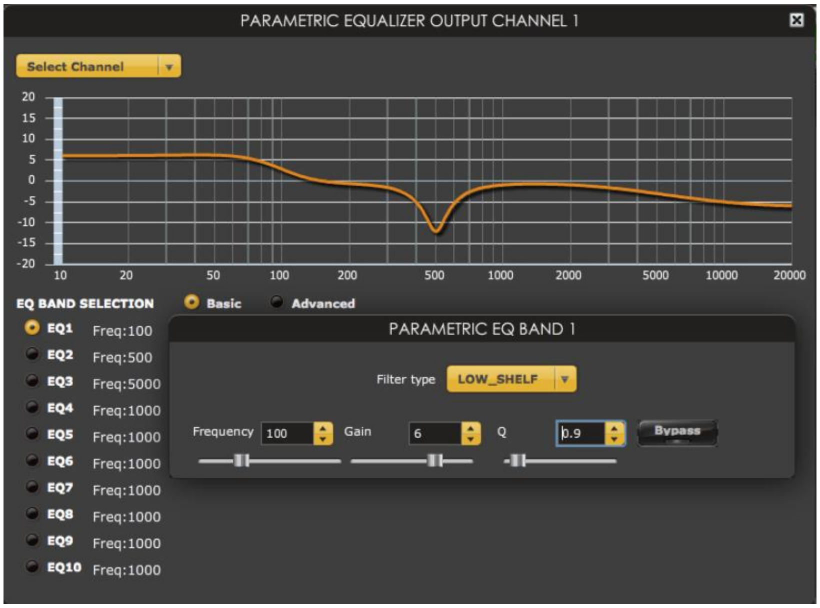
Clicking on the **Bypass** button disables or enables that high pass or low pass filter. The filter is bypassed when the button is "lit".

Enabled Bypassed

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

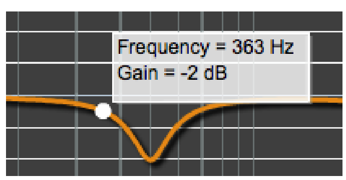
7.2 PARAMETRIC EQ

Parametric equalization (PEQ) is a flexible type of equalization filter. It can be used to correct for errors in loudspeaker output, to compensate for acoustic room effects, and to tailor the overall system response for best sound. Click on the PEQ button to open the parametric equalizer settings window:



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

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



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

EQ band selection

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

Basic/Advanced

By default, each filter is in basic mode, and shows the controls described below. Advanced mode enables custom biquad programming for almost infinite flexibility in filter implementation. This is described in Custom biquad programming on page 43.

Filter type

Selects the type of filter:

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

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

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

SUB_EQ Create a dip or a peak in the frequency response at low frequencies (10 to 50 Hz). This filter type is similar to PEAK but gives more accurate results for low frequencies. Note that activating any SUB_EQ filter reduces the number of available filters on that channel by one (from 10 to 9 for miniSHARC plugins, or from 6 to 5 for the OpenDRC plugin.)

Frequency

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

Gain

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

Q

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

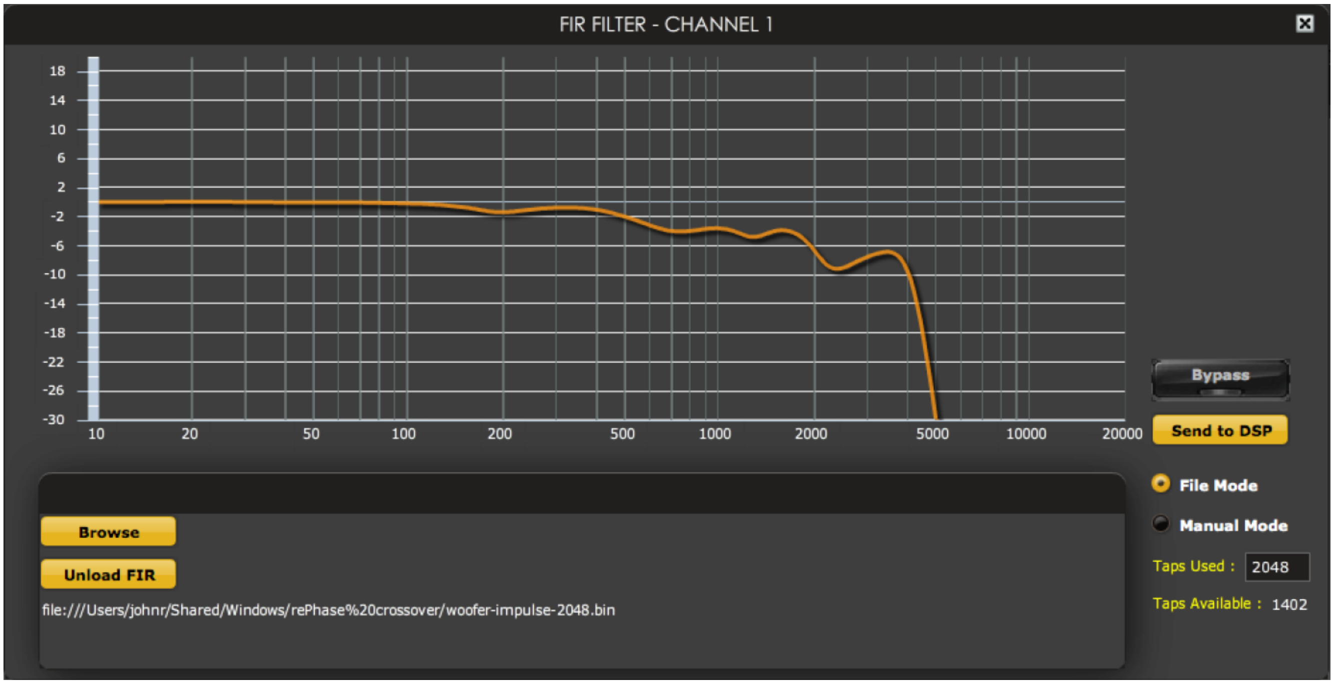
Bypass

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

Enabled Bypassed

7.3 FIR FILTERING AND DESIGN

FIR filtering is a powerful and advanced feature of the miniSHARC and associated plugins. It allows construction of complex arbitrary equalization and crossover filters with independent control of amplitude and phase. The parameters of each FIR filter are set in the **FIR** settings window:



Browse Opens a filter browser to select a file containing FIR filter coefficients. (See FIR filter file format below).

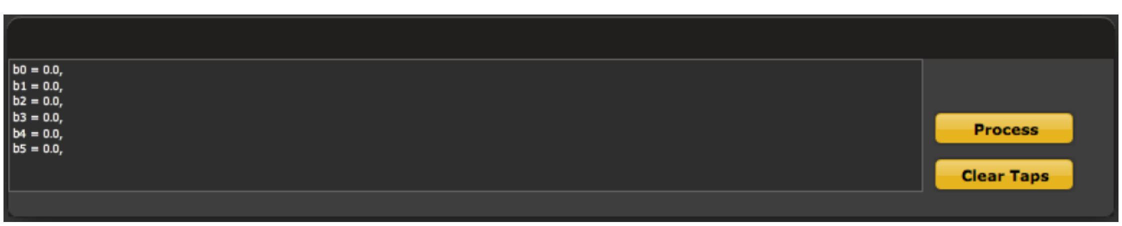
Unload FIR Deletes the currently loaded filter from the display and from the DSP memory.

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

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

File Mode / Manual Mode

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

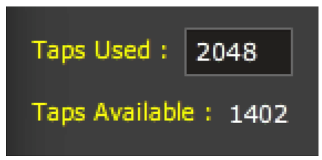


7.3.1 FIR filtering overview

FIR ("finite impulse response") filtering differs from the IIR ("infinite impulse response") filters used in the PEQ and crossover blocks. Technically speaking, IIR filters are *recursive*, meaning that each output value is partially calculated from earlier output values as well as from input values. An FIR filter is specified by a large array of numbers, whereas an IIR filter requires only a fairly small of values to be specified.

These numbers are conventionally referred to as "taps." The tap numbers for each plugin are given in Table 1.

- In the OpenDRC 2x2 plugin, each channel has a fixed number of taps (6144).
- In the case of the miniSHARC 4x8 plugins, taps can be distributed across the eight output channels, with the limitation that each output channel must have 6 or more taps and can have no more than 2048 taps. The decision on how many taps to allocate to each channel up to you, and should be determined after working with an FIR filter design program (see below). The number of taps is set in the lower right corner (click on the text entry box and type the desired number of taps, then press **Tab** or **Return**):



7.3.2 FIR filter design software

The filter coefficients must be created with the aid of filter design software. miniDSP does not provide any such software, instead referring you to the many software packages available for this purpose (both freeware and commercial). Please see the [FIR filter tools] page on our website.

7.3.3 Filter file format

The filter coefficient file loaded in **File Mode** uses [IEEE 754 single-precision binary floating-point format]. The number of entries in the file must not exceed the allocated number of taps.

In Manual Mode, the coefficients must be plain text in this format:

```
b0 = 1,
b1 = -1,
b2 = 0.5,
b3 = -0.5,
b4 = 0.2,
b5 = 1,
```

7.3.4 Loading filter coefficients

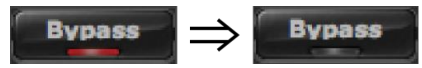
In **File Mode**:

1. Click **Browse**, navigate to the file containing the filter coefficients, and open it. A dialog will appear confirming the number of coefficients loaded.
2. Confirm that the response curve is as you expect.
3. Press **Send to DSP**. This will write the coefficients into the DSP's memory.
4. To clear the filter coefficients, click **Unload FIR** and then **Send to DSP**.

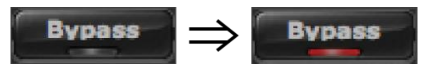
In **Manual Mode**:

1. Cut and paste the coefficients from the text output of the design program.
2. Press the **Process** button.
3. Confirm that the frequency response graph is as you expect.
4. Press **Send to DSP**. This will write the coefficients into the DSP's memory.
5. To clear the filter coefficients, click **Clear Taps** and then **Send to DSP**.

If, after selecting a filter file or setting coefficients, the frequency response graph does not change as expected, make sure that the Bypass button is turned off.



If you clear the filter taps, make sure that you also bypass the filter, otherwise there will be no audio through that channel.



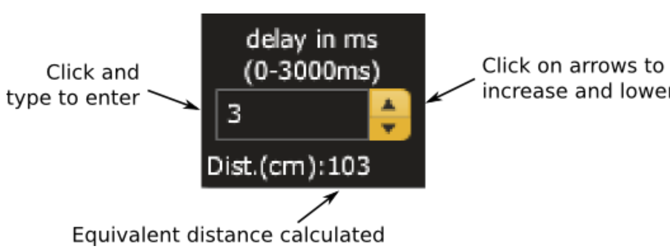
7.4 Invert and Mute

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

7.5 Time Delay

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

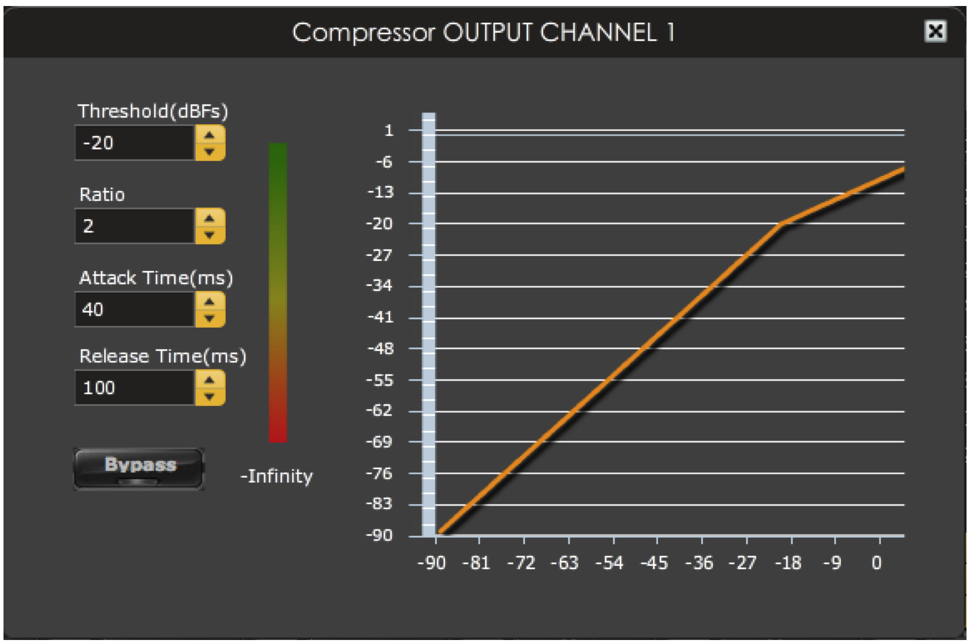
The time delay corresponds to a distance. This distance is shown in centimeters below the entry box.



7.6 COMPRESSOR

The compressor reduces the gain of an output channel when the audio signal reaches a certain level as specified by the **Threshold** parameter. The gain of the channel will be progressively reduced as the signal increases above the threshold, according to the **Ratio** parameter. This can be used to limit the power delivered to speakers and thus reduce the risk of damage from overdriving.

This screenshot shows an example Compressor setting:



(Note that the compressor algorithm is bypassed by default, so click on the **Bypass** button to see the curve as shown here.)

In this example, the threshold is set to -20 dB, so the compressor will activate when the signal on that channel reaches -20 dB (relative to full output). The ratio is set to 2, so if the input signal level to the compressor then increases by 10 dB, the output level will increase by only 5 dB. If the input signal level to the compressor is at full scale (0 dB), then the output level will be limited to -10 dB.

Two additional parameters control the action of the compressor: the attack time and the release time. These two parameters govern how quickly the compressor activates when the signal level exceeds the threshold, and how quickly it deactivates when the signal level reduces. The optimum settings may need to be tuned by ear. For more information, see the Wikipedia article [Dynamic range compression].

7.7 CUSTOM BIQUAD PROGRAMMING

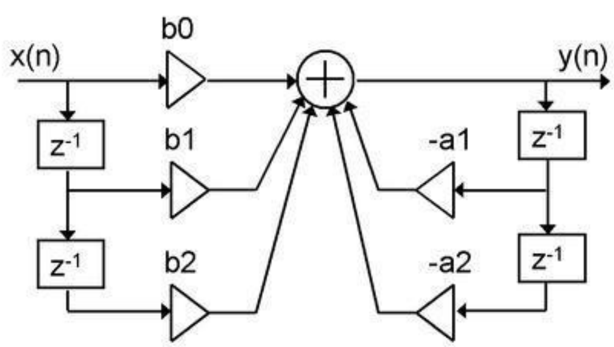
Custom biquad programming is available in the PEQ and Crossover blocks. Its purpose is to allow you to directly provide the low-level parameters aka *biquad coefficients* that control the digital filters of the processor, thus providing an almost infinite degree of flexibility.

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

7.7.1 What's a "biquad?

A biquad is the basic unit of processing that is used to create digital filters. It can be described either with an equation or with a signal flow diagram, as shown here:

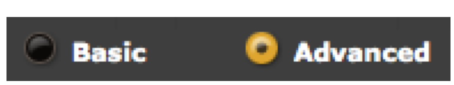
$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}$$



A single biquad like this can perform a great many functions, including all of the functions of a single parametric EQ filter, one 6 or 12 dB/octave high pass or low pass filter, and more. Biquads are combined in series (cascaded) to create more complex filters. The function that each biquad performs is determined by just five numbers: *a1*, *a2*, *a0*, *b1*, and *b2*. These numbers are called the *coefficients*.

7.7.2 Using custom biquad programming

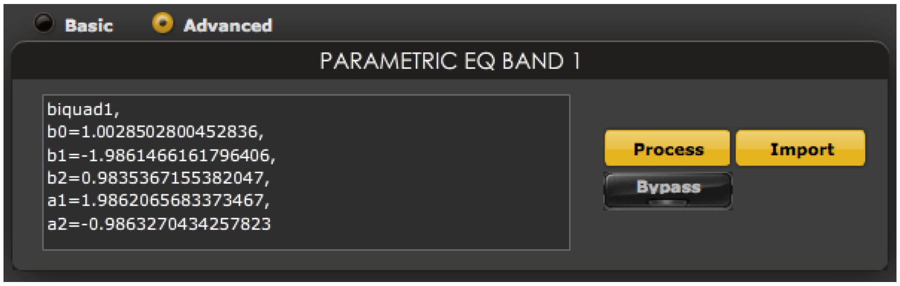
Each crossover block and PEQ filter has a selector that switches it to advanced mode:



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

Parametric EQ advanced mode

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



Parametric EQ file import (REW integration)

Multiple biquads in the parametric EQ block can be set at once by importing a coefficient file. This file can be generated by Room EQ Wizard (REW) or by other programs. The design program must be set for a **48 kHz** sample rate if using the **miniSHARC 4x8** plugin, and a **96 kHz** sample rate if using the **miniSHARC 4x8 96k** plugin. The number of filters is limited to a maximum of ten.

This example illustrates the correct file format:

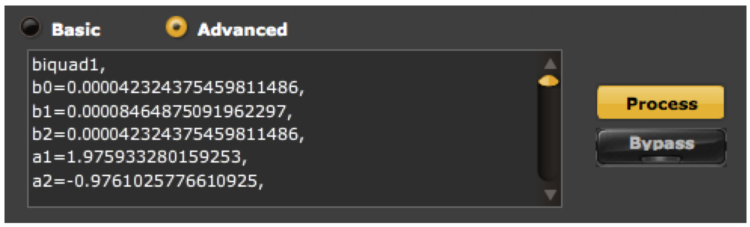
```
biquad1,
b0=0.998191200483864,
b1=-1.9950521500467384,
b2=0.996920046761057,
a1=1.9950521500467384,
a2=-0.9951112472449212,
biquad2,
b0=0.999640139948623,
b1=-1.9981670485581222,
b2=0.9985489719847982,
a1=1.9981670485581222,
a2=-0.9981891119334211,
biquad3,
...
biquad4,
...
biquad5,
b0=1.0010192374642126,
b1=-1.9950555192569264,
b2=0.9940580112181501,
a1=1.995060938714333,
a2=-0.9950718292249559
```

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

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

Crossover advanced mode

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



7.7.3 Biquad design software

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

7.7.3.1 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

7.7.3.2 Room EQ Wizard (REW)

Room EQ Wizard is a free acoustic measurement and analysis tool, available for Windows, Mac and Linux platforms. It includes the ability to automatically generate a bank of parametric EQ biquads based on a measurement. These coefficients can be saved to a file from REW and loaded directly into a PEQ bank in a miniDSP plugin. Room EQ Wizard can be downloaded here:

- http://www.roomeqwizard.com/#downloads

For guidance on using this feature, please refer to the app note Auto EQ with REW.

8 ADDITIONAL INFORMATION

8.1 SPECIFICATIONS

Computer connectivity	Driverless USB 2.0 control interface for Windows and Mac OS X
Audio resolution	24-bit input and output **miniSHARC 4x8** and **OpenDRC 2x2** plugins: 48 kHz internal sample rate **miniSHARC 4x8 96k** plugin: 48 kHz internal sample rate
Audio processing	Analog Devices SHARC 32-bit floating-point processor Specific processing functionality depends on loaded plugin. See Sections 6 and 7 of this manual.
Storage/presets	All settings controllable in real time from software user interface. Up to 4 presets stored in local flash memory.
Infrared remote (when used with optional accessory board VOL-FP)	"Learning remote" capabilities (NEC, Philips, Sony) Controls master volume, mute, digital input selection, preset selection.
Power supply	5–24V DC if used standalone 5 VDC @ 600mA if using any accessory boards (VOL-FP, DIGI-FP, AN-FP, DA-FP, miniDAC8)
Dimensions (board only, H x W x D)	75 x 75 x 17 mm

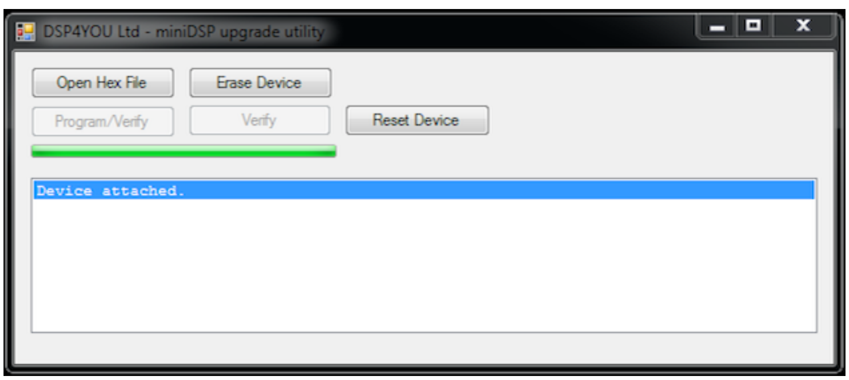
8.2 MCU FIRMWARE UPDATE

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

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

To update the MCU firmware:

1. Download and install the latest version of either the **miniSHARC 4x8** or **OpenDRC 2x2** plugin from the **User Downloads** section of the miniDSP website.
2. Connect the processor to your computer via USB.
3. Start the plugin (if it is not already running).
4. Click on the **Connect&Synchronize** button and select the **Synchronize** option from the dialog.
5. From the menus, select **Restore -> Upgrade Firmware**. The processor will be put into boot loader mode and the miniDSP upgrade utility will start. The status line should display "Device attached".



6. Click on the **Open Hex File** button and select the .hex file included in the download. It will have a name like **ENC_miniSHARC_Ver2.18_SS.hex**.
7. Click on the **Program/Verify** button. The status bar will indicate progress. **Do not disconnect or power off** the processor during the firmware upgrade!
8. After the status indicates that the verification step has completed successfully, click on the **Reset Device** button, and then quit the upgrade utility.
9. Return to the **miniSHARC 4x8 or OpenDRC 2x2** plugin. (If there is a dialog informing "Connection to DSP closed," click on **OK**.) Click on the **Connect** button, and then on **About this product** from the **Help** menu to verify the new firmware version.

8.3 TROUBLESHOOTING

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

Item#	Symptoms	Troubleshooting recommendation
1	Cannot install software	a. Confirm that you downloaded and installed the required frameworks first (see Software Installation).
2	Software running in background but not showing	a. The Adobe Air environment may need a network connection the first time you run a plugin. Close the plugin program, ensure that your computer has a network connection, and restart the plugin. b. The Adobe Air environment may require a version update. Download the latest version from http://get.adobe.com/air/.
3	Cannot connect to the board by USB	a. Reset the processor by power-cycling the unit. b. Make sure the processor is seen in the device manager as a HID device.
4	Cannot reload a configuration	a. Confirm the file format of your file (.xml). b. Confirm the version of the file.

8.4 OBTAINING SUPPORT

1. Check the forums on miniDSP.com to see if this issue has already been raised and a solution provided.
2. Contact miniDSP via the support portal at minidsp.desk.com with:
    a. The specific product you are having an issue with (in this case, miniSHARC board or accessories).
    b. A clear explanation of the symptoms you are seeing.
    c. A description of troubleshooting steps (see Troubleshooting above) performed and your results.

Please note that miniDSP is only able to provide support for the hardware and functions documented in this manual, and only for problems specifically related to the miniDSP hardware and software functions. Any other items, such as designing or debugging your I2S interface circuitry or layout or interfaces to third-party hardware, are specifically excluded from the scope of miniDSP support.

If you are not confident in your ability to design, debug and interface external circuitry to the miniSHARC, please consider using our of-the-shelf interface accessories (section 3) or our pre-assembled "box" units in the OpenDRC and Dirac Series.