

HA-DSP

DSP-ENABLED PORTABLE HEADPHONE AMPLIFIER



User Manual



Revision history

Revision	Description	Date
V0.2	Preliminary version	12 June 2017
V0.3	Updated for released plugin	28 June 2017

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IMPORTANT INFORMATION

Please read the following information before use. In case of any questions, please contact miniDSP via the support portal at 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 HA-DSP has passed the test performed according to European Standard EN 55022 Class B.

A NOTE ON THIS MANUAL

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

1 PRODUCT OVERVIEW

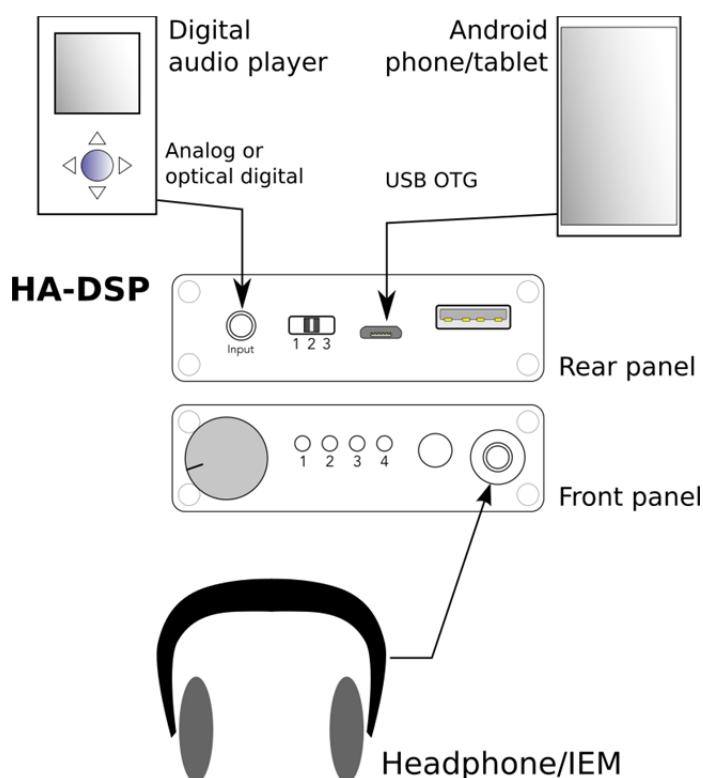
Thank you for choosing the miniDSP HA-DSP DSP-enabled portable headphone amplifier. The HA-DSP is a:

- **DSP processor.** The HA-DSP incorporates a powerful set of features ranging from gain, bass, and treble controls, through 10-band parametric EQ on each channel, to a powerful cross-feed FIR (finite impulse response) filter array for advanced headphone equalization and cross-feed processing.
- **DAC.** The digital audio data is fed to a 32-bit ESS Sabre DAC. The USB input also supports direct DSD playback, where DSD audio (DoP) is fed directly to the DAC chip. (DSP processing is disabled while playing DSD.)
- **Headphone amplifier.** A Texas Instruments TPA6120 provides up to 100 mW in each channel into 32 ohms with very low distortion and a signal-to-noise ratio of 112 dB.

Four DSP processing configurations are stored on-board and can be selected using the button on the front panel. The HA-DSP features three inputs – USB, digital optical, and analog – allowing it to be used with almost any portable device.

1.1 TYPICAL USAGE EXAMPLE

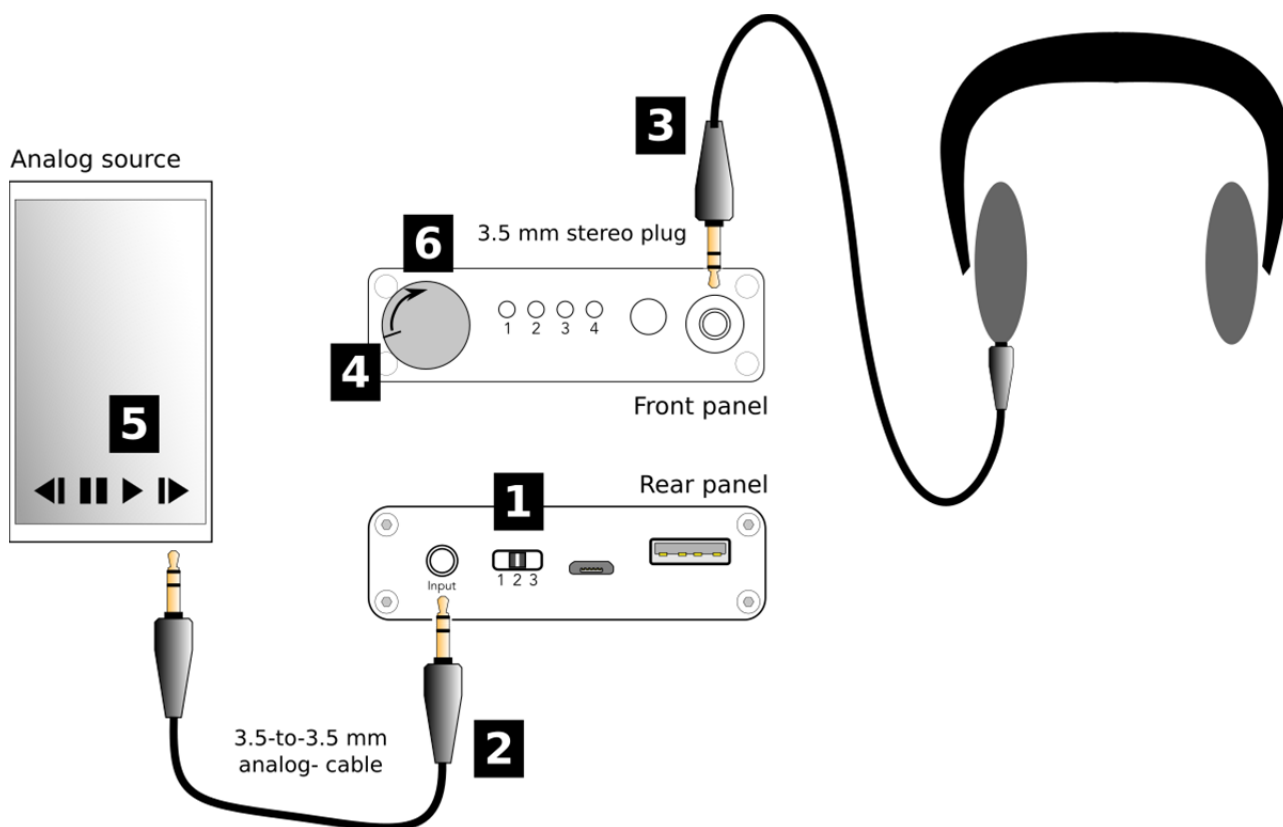
The HA-DSP has a number of uses. This diagram illustrates one typical usage mode.



2 QUICK-START GUIDE

While we do recommend that you install the miniDSP software on your computer and familiarize yourself with the powerful features of the plugin sooner rather than later, you *can* simply play audio through your HA-DSP when you receive it.

1. Check that the rear panel switch is set to position 2.
2. Connect an audio source to the analog input. (If connecting to the digital optical input or to the micro-USB port, see Section 4.)
3. Connect your headphones to the headphone output.
4. Turn on the HA-DSP with the main knob. Leave the volume knob at a low level at this point.
5. Wait until one of the LEDs on the front panel of the HA-DSP turn on. Start playing audio from your source.
6. Gradually turn up the volume knob.



3 INSTALLING THE SOFTWARE

For full control over your HA-DSP, you will need to install software as follows:

Windows

- The HA-DSP plugin software that runs on your computer.
- The ASIO driver needed to stream audio.

See page 10 for Windows installation.

Mac

- The HA-DSP plugin software that runs on your computer.

See page 11 for Mac installation.

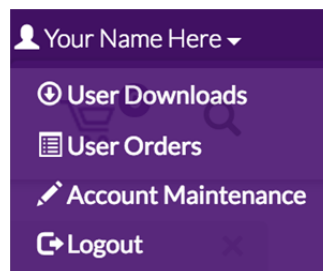
3.1 DOWNLOADING THE SOFTWARE

If you purchased your product directly from miniDSP, your software will be available from the [User Downloads](#) area 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 “HA-DSP” 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 **HA-DSP plug-ins** section and download the zip file under the heading **HA-DSP**.

3.2 INSTALLATION ON WINDOWS

The miniDSP software installer will automatically download and install additional software that it requires. You must therefore be connected to the Internet when you run the installer.

3.2.1 Special note for Windows XP and Vista

For Windows XP and Vista, download and install the following frameworks first.

- [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)

Windows XP and Vista also use a different version of the USB driver. Follow the instructions below.

3.2.2 Plugin installation

To install the plugin:

1. Unzip the downloaded software: right-click on its icon and select “Extract All...”.
2. Navigate to the **Windows** folder of the unzipped download.
3. Double-click on the **HA-DSP.exe** installer program to run it. We recommend that you accept the default installation settings.
4. The plugin will start automatically (if you accepted the default installation settings). To make it quicker to run in future, right-click on its icon in the taskbar and select “Pin to taskbar.”

3.2.3 USB driver installation

The USB driver is required for the plugin to communicate with the HA-DSP. It is also required for full audio streaming functionality.

1. Navigate to the **WinDrivers** folder of the software download and double-click on the appropriate installer (note: the version number embedded in the filename may be different):
 - **miniDSP_UAC2_v2.29.0_ForWinXP_Vista.exe** for Windows XP and Vista
 - **miniDSP_UAC2_v4.11.0_2017-06-19_setup.exe** for Windows 7, 8, and 10
1. We recommend accepting the default installation location. Once the driver installation completes, click the **Finish** button.
2. To complete installation of the driver, the HA-DSP will need to be connected and turned on. You can do this now, or connect the HA-DSP later and turn it on to allow Windows to finish the installation.

3.2.4 Additional installation notes

Note 1: The Adobe Air framework may need to connect to the Internet the first time you run the plugin. It's strongly recommended that you are connected to the Internet the first time you start 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.”

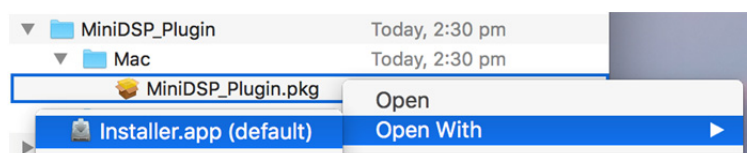
3.3 INSTALLATION ON MAC

To install the plugin:

1. Unzip the downloaded software by double-clicking on the zip file.
2. Navigate to the **Mac** folder of the unzipped software download.
3. The installer program is named **HA-DSP.pkg**. To run it, double-click on it, or right-click and open as described below. We recommend that you accept the default installation settings.
4. To run the plugin, locate **HA-DSP.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 1: 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 **HA-DSP.pkg**, not **MiniDSP_Plugin.pkg** as shown in the screenshots):

1. Right-click on the installer (or click while pressing the Control key).
2. On the menu that pops up, move the mouse over the “Open With” item and then click on “Installer (default).”



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



Note 2: The Adobe Air framework may need to connect to the Internet the first time you run the plugin. It's strongly recommended that you are connected to the Internet the first time you start the plugin.

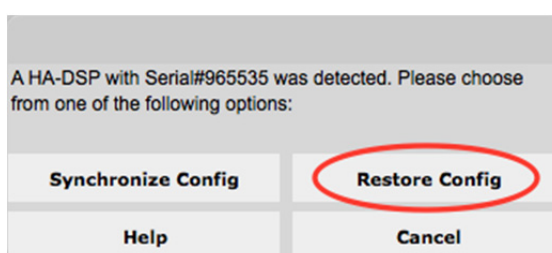
3.4 FIRST-TIME CONNECTION

When you start the plugin, the user interface screen appears as shown below:



To connect to the HA-DSP:

1. Connect the supplied USB cable from your computer to the micro-USB port on the HA-DSP.
2. Turn on the HA-DSP with the volume knob. Don't connect your headphones yet.
3. Click on the **Connect** button at the top right of the window. The **first time** the following dialog pops up, click on "Restore Config". This ensures that both the user interface and the HA-DSP are reset to



defaults.

4. The Connect button will change to show "Connected". At this point, the plugin is in online mode. This is also called "synchronized" – any changes that you make in the user interface will take effect



immediately.

4 CONNECTIONS

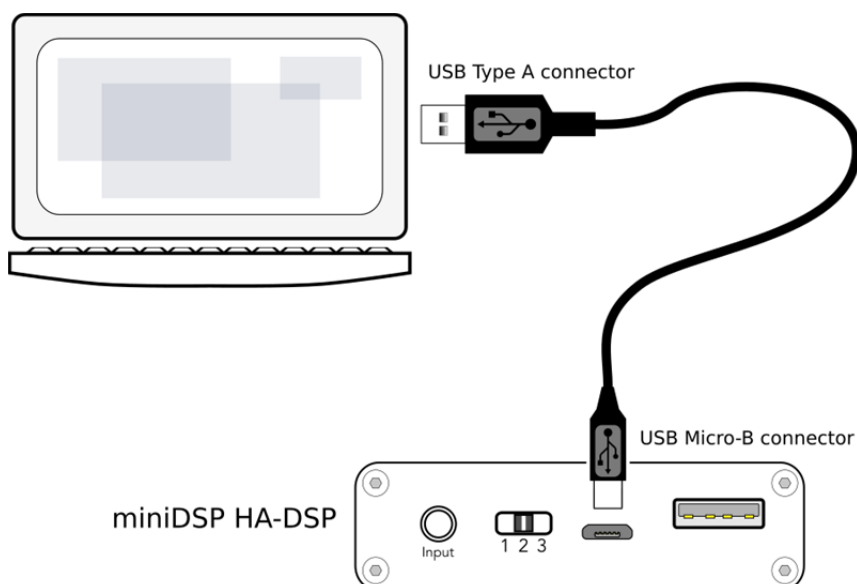
4.1 INPUTS

The HA-DSP accepts input over USB, analog, or optical digital. For normal operation, the rear panel switch should be left in position 2.

4.1.1 USB

4.1.1.1 Computer

USB connection from a computer is used for both configuring the plugin with the plugin GUI, and for streaming audio. Use the supplied USB cable to connect from a USB port (must be USB2.0 compatible) on the computer to the micro USB connector on the HA-DSP.

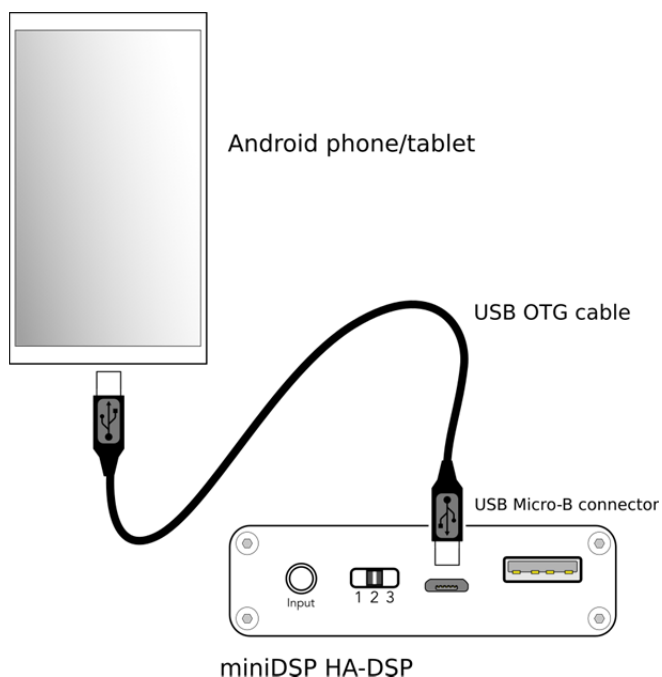


For information on configuring with the plugin, see later sections of this manual.

4.1.1.2 Android

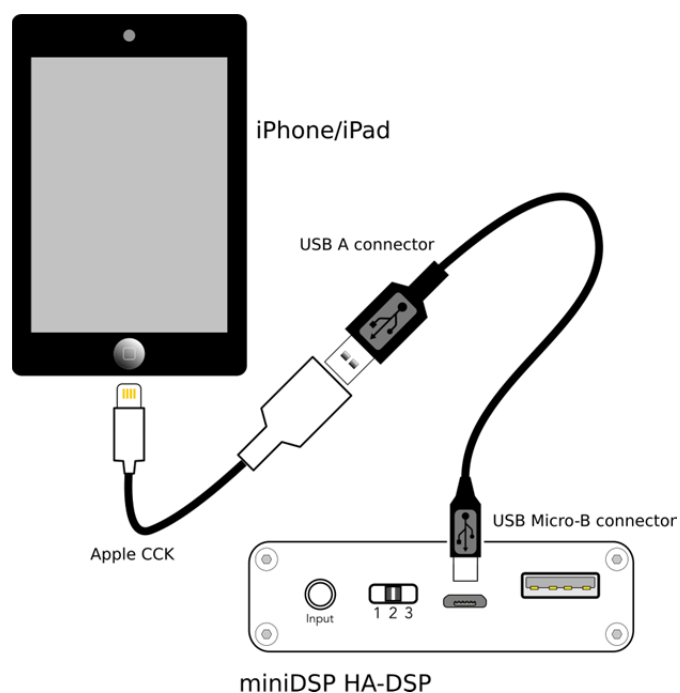
To connect an Android device to USB, use the supplied OTG (“on the go”) cable to connect from your Android device’s micro USB port to the micro USB port on the HA-DSP.

Your Android device must support the USB OTG standard to play audio to the HA-DSP over USB. Check the specifications of your device and/or its user manual. (If your device does not support OTG, then you can still play audio through the HA-DSP using the analog input.)



4.1.1.3 IOS (Apple)

To connect an iOS device (iPhone or iPad) to the HA-DSP via USB, use the Apple Camera Connection Kit (CCK) and the supplied USB type A to micro USB cable.

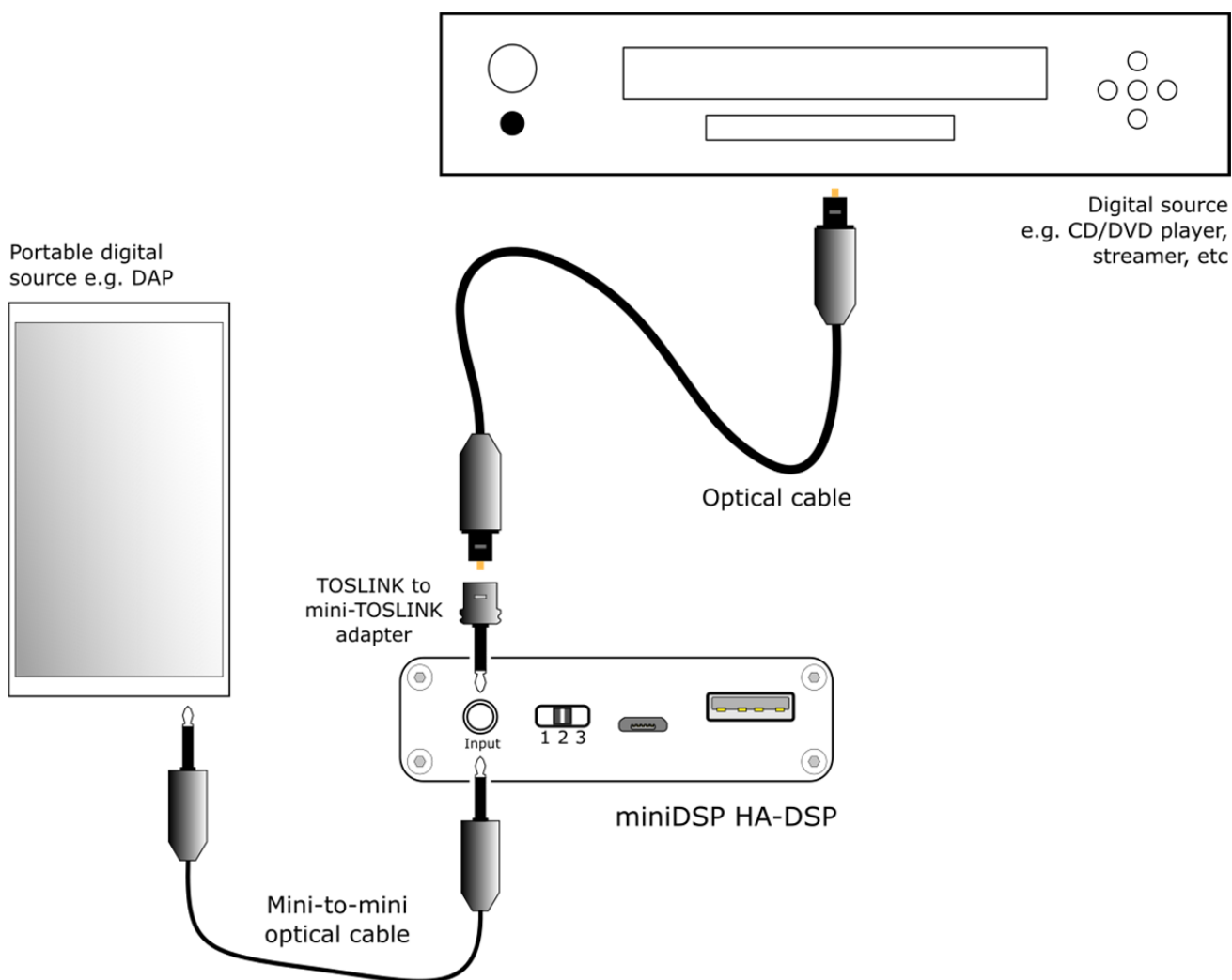


4.1.2 Digital optical

To connect a digital source, plug a 3.5mm optical connector into the input jack. The HA-DSP will automatically detect the optical input and switch to it. USB Audio will be disabled. You can use:

- The supplied TOSLINK to mini-TOSLINK adapter with a regular TOSLINK cable, or
- A cable that has a mini-TOSLINK on the end.

This diagram shows two typical connection examples:

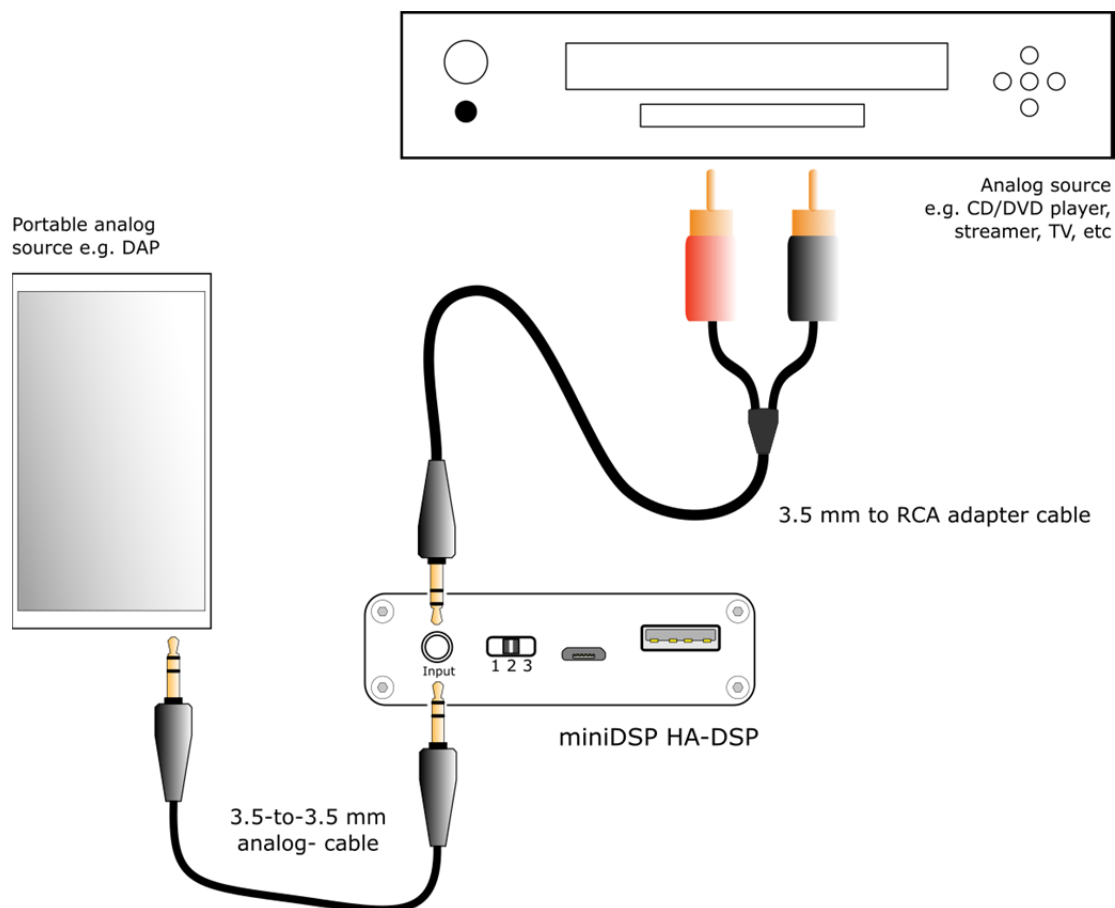


4.1.3 Analog

To connect an analog source, plug a 3.5 mm stereo plug into the input jack. The HA-DSP will automatically detect the analog input and switch to it. USB Audio will be disabled. You can use:

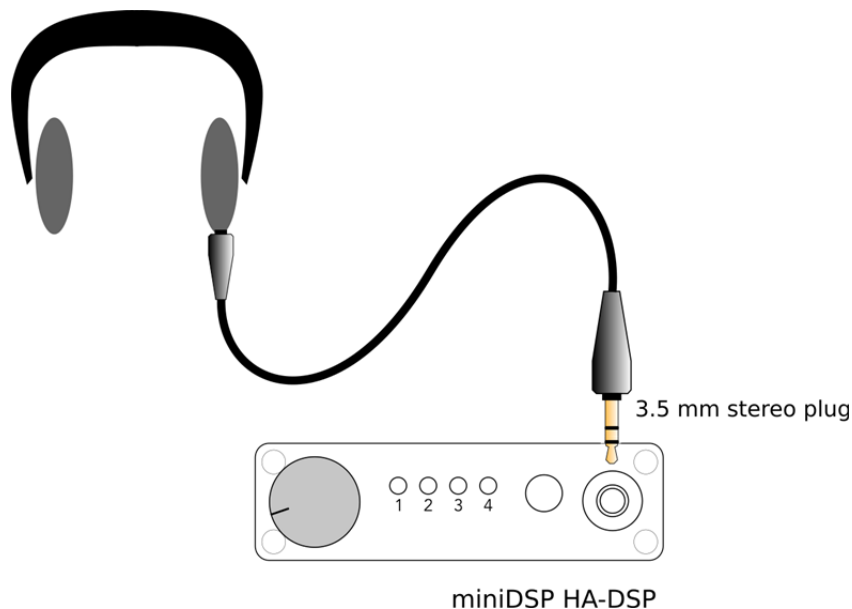
- The supplied 3.5 mm to 3.5 mm cable. This short cable is perfect for connecting to a phone.
- Any other stereo cable with a 3.5mm stereo connector. For example, in some cases, the source may have RCA jacks. In that case, use an RCA to 3.5 mm adapter cable.

This diagram shows two typical connection examples:



4.2 OUTPUT

To connect headphones, earbuds, or in-ear monitors (IEMs), simply plug them into the 3.5mm jack on the front panel.



If your headphones have a 6.35 mm (1/4-inch) plug, you will need to use an adapter plug or pigtail.

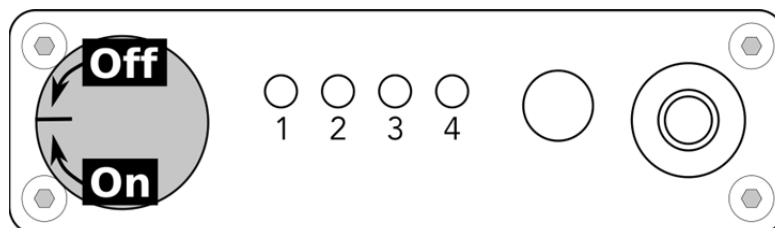


Turn the volume knob to the lowest position (without necessarily turning off the amplifier) when inserting the headphone plug.

4.3 TURNING THE AMPLIFIER ON AND OFF

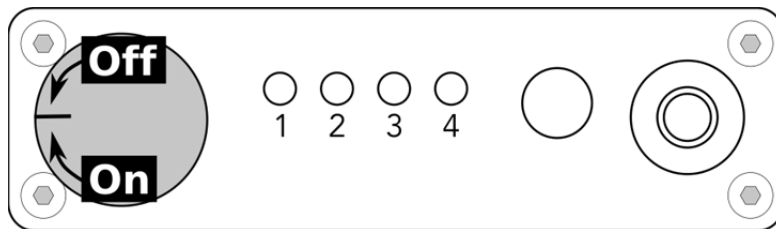
To turn the amplifier on, rotate the knob clockwise. The DSP will take a few seconds to initialize, after which one of the LEDs (numbered 1 to 4) corresponding to the selected DSP preset will turn on.

To turn the amplifier off, rotate the knob counterclockwise until it clicks and the LEDs are all out.



4.4 FRONT PANEL PRESET CONTROL AND LED STATUS

The HA-DSP is fitted with a front panel momentary switch to control the active DSP preset. Internal Flash memory will store up to 4 x DSP preset banks.



To change to a new preset:

- Perform a short press to toggle to the new preset bank
- Perform a long press (2sec) to confirm the selection. If a confirmation isn't performed, the LED will revert to the currently loaded preset.

LED Status and their meanings:

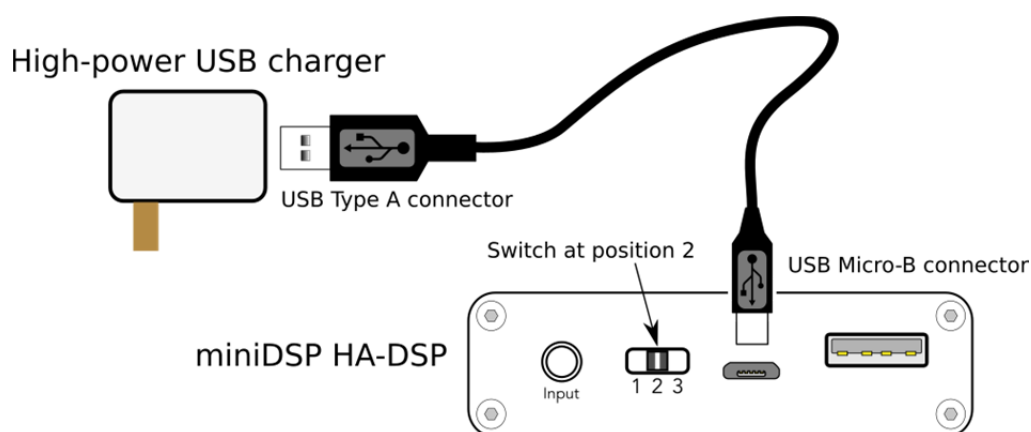
- LED1: Preset1 selected
- LED2: Preset2 selected
- LED3: Preset3 selected
- LED4: Preset4 selected
- LED 1&4: Bypass DSP
- LED 2&3: DSD mode (automatically selected when playing DSD file)

4.5 CHARGING AND POWER

4.5.1 Fast-charging the HA-DSP

USB ports supply only limited current. Therefore, while connected to a device such as a computer, the HA-DSP will only trickle-charge. If it is in use at the same time to play audio, it may even continue to drain the battery.

To charge the HA-DSP more quickly, connect it to the dedicated 2A USB charger provided along the HA-DSP. A USB hub with at least 2A capacity can be used to play audio at the same time as fast charging.



4.5.2 Checking battery level / Upcoming Firmware.

1. Turn the amp off (if it is not already)
2. Press and hold the control button.
3. After a second or two, between 1 and 4 of the LEDs will light up. 1 LED means the battery is very low, while 4 LEDs means that the battery is fully charged.
4. Turn the amplifier back on if required.

5 DSD PLAYBACK MODE

The HA-DSP detects DSD (Direct Stream Digital) audio sent to it over the USB connection and switches to play it automatically. When in DSD playback mode, **all audio processing except for input selection and final stage volume is disabled**.

5.1 WHAT IS DSD?

DSD is a relatively recent audio format. It is the form in which data is stored in SACDs. (Regular CDs, in contrast, store data in PCM - pulse code modulation - format.)

DSD audio will be contained in files ending in the suffix “.dsf” or “.dff”. You will be playing DSD only if you have purposely downloaded DSD files and use a DSD-capable audio player to play them. Regular audio from your computer will always be in PCM format. Common audio format such as FLAC, WAV, ALAC, AIFF, and MP3 are all PCM format.

5.2 HOW DO I PLAY DSD?

By using a DSD-capable audio player. These days, most audiophile-level music players are DSD-aware. Typically, you will need to tell the player (e.g. in the Preferences settings) that the HA-DSP can play DSD using DSD-over-PCM (DoP).

5.3 WHY IS AUDIO PROCESSING DISABLED IN DSD PLAYBACK MODE?

The DSD format is not conducive to audio processing. When the HA-DSP detects DSD audio, it simply bypasses the digital signal processing (DSP) chip and sends the DSD data directly to the DAC chip.

5.4 WHAT IF I WANT TO KEEP THE PROCESSING WHILE PLAYING DSD FILES?

You will need to tell the player to convert the DSD to PCM before sending it to the HA-DSP. Some players will do this automatically if you have not told it to send DSD to the HA-DSP; others may need a specific setting to convert to PCM.

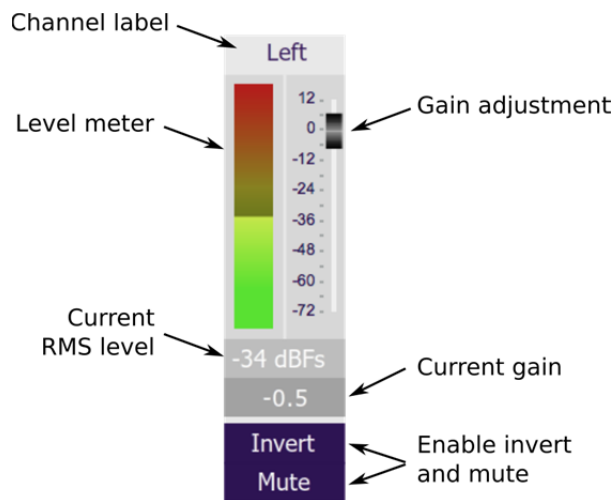
5.5 WHY IS IT IMPORTANT THAT I KNOW THAT PROCESSING IS DISABLED IN DSD MODE?

Any EQ or cross-feed that you have set up in the HA-DSP will no longer function. In addition, the gain settings in the input section will (in effect) be set to 0 dB. If you have applied a lot of attenuation here, you will get a jump in volume when you switch from playing a PCM file to playing a DSD file. Usually, this is not a problem as the input attenuators are not set very far below zero. You just need to be aware that the volume *could* change.



In DSD playback mode, all audio processing is disabled. This may result in a sudden change in volume.

6 INPUT CONTROLS



Channel label

Each input channel is labeled. The label can be changed: to do so, click on it, type a new label (up to eight characters), and press the Return key.

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.)



Gain adjustment is disabled (along with all other processing) in DSD playback mode. If you have this set to a low value, be aware that it will effectively be at 0 dB for DSD playback.

Invert

Press this button to invert that channel. The button color and label changes to show that the channel is inverted.



Mute

Press this button to mute that channel. The button color and label changes to show that the channel is muted.



7 LINKED MODE

Put the plugin into linked mode by clicking on the linking icon.



In linked mode, the following controls are linked for both channels:

- Input gain, invert and mute
- Bass and treble settings
- Parametric EQ

The FIR filters and the routing matrix are *not* linked.

Generally, it's best to start using the plugin in linked mode. When you find a reason to have separate settings for each channel, at that time you can unlink them. The settings of the two channels after unlinking will initially be the same.

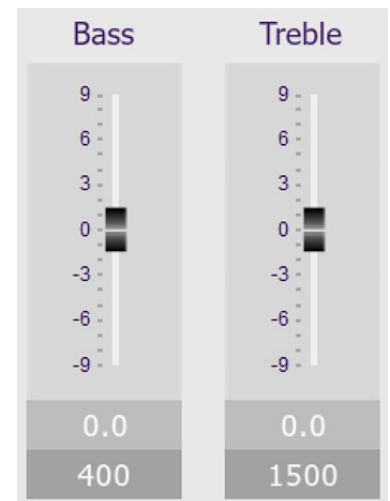
8 BASS AND TREBLE CONTROLS

The input screen includes bass and treble controls to allow a quick way to boost the high or low range of the frequency spectrum. Adjust the level of boost or cut simply by moving the slider. Alternatively, the amount of boost or cut can be entered directly in the entry box underneath the slider.

The frequency range over which the controls take effect can be set in the lower box:

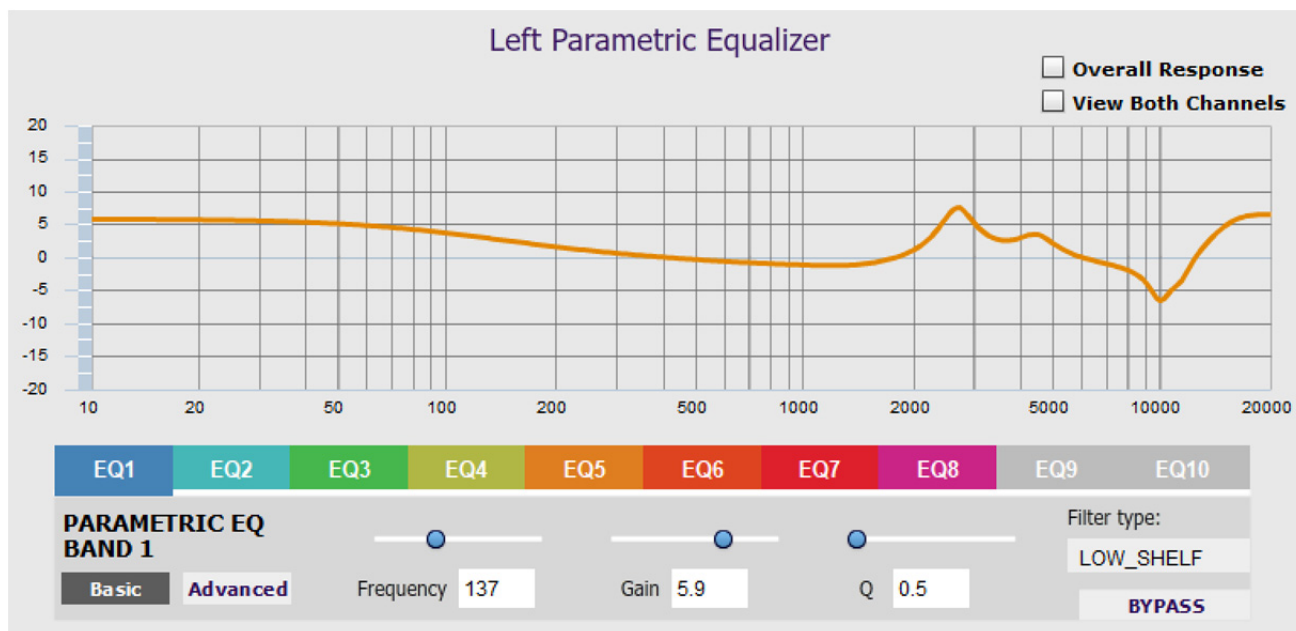
- Bass can be adjusted from 200 to 750 Hz. The default is 400 Hz.
- Treble can be adjusted from 750 to 300 Hz. The default is 1500 Hz.

The effect of these controls can be viewed on the parametric EQ screen, by checking the "Overall Response" checkbox.



9 PARAMETRIC EQ (PEQ)

Parametric equalization (PEQ) is a flexible type of equalization filter. To open the parametric equalizer settings window, click on Left Parametric EQ or Right Parametric EQ button. Here is an example window with one of the preset headphones settings:

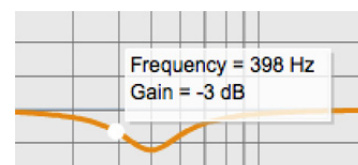


If you are in Linked Mode, the graph and the settings apply to both channels. Otherwise, they apply to either the left or right channel.

9.1 OVERVIEW

There are ten parametric EQ filters on each channel. Click on any of the colored labels EQ1, EQ2 and so on to select the controls for that filter. The graph in orange represents the frequency response of all filters combined. If the button “Overall Response” is checked, this also includes the effect of the Bass and Treble controls in the Input Control block.

You can hover the mouse over the orange curve to bring up an overlay showing the frequency and the corresponding gain:



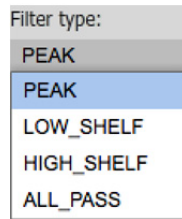
PEQs are specified either in Basic mode or Advanced mode. For Advanced mode, see page 31.

To disable an individual filter, click on the **Bypass** button at the lower right. The filter is enabled if the button says “BYPASS” and disabled if the button says “BYPASSED”:



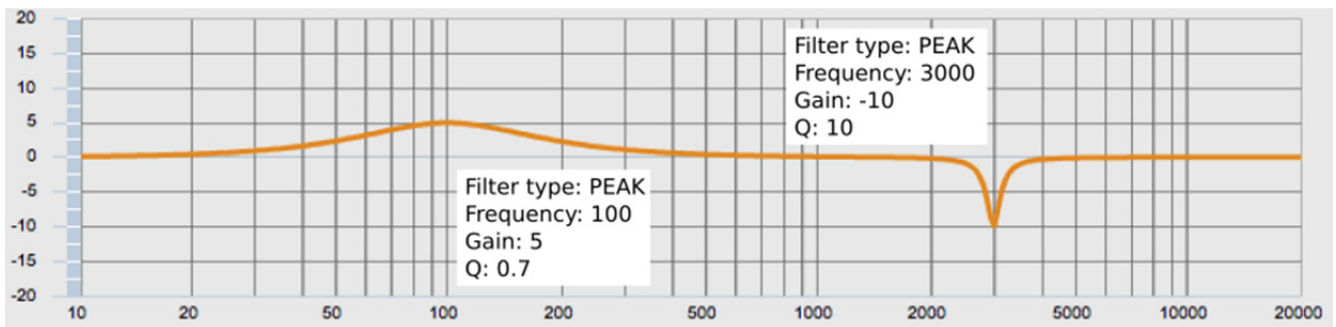
9.2 TYPES OF FILTER

There are four types of filter. This is selected towards the lower right of the settings window:



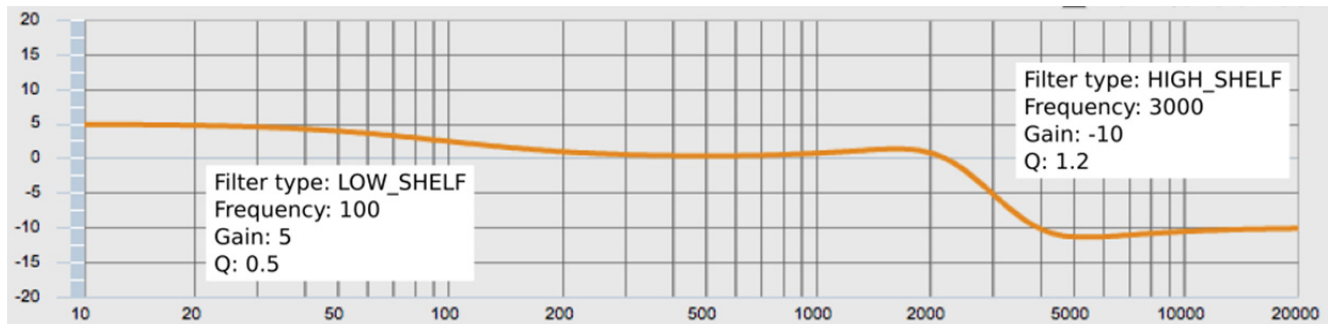
9.2.1 PEAK

This filter creates a dip or a peak in the frequency response. These two examples show how the parameters affect the response:



9.2.2 LOW_SHELF and HIGH_SHELF

This filter reduces or increases part of the frequency spectrum. The Frequency parameter is the frequency where the response has half the value given by Gain (for example, if Gain is 10, then at Frequency Hz the gain is 5 dB). These two examples show how the parameters affect the response:



9.2.3 ALL_PASS

This filter creates a varying phase shift across the frequency band, with the specified frequency being the frequency where half the total phase shift occurs. This filter will have no effect on the displayed graph. This should be considered an advanced option that will not be of use to 99% of users.

10 CONFIGURATION PRESETS

The complete set of data that controls the DSP processor is called a *configuration*. This includes input levels, bass and treble controls, parametric EQ, FIR filtering and the routing matrix. It does **not** include the volume set by the front panel knob.

Four configurations are stored onboard in the HA-DSP. By default, these are all set to “straight-through” processing.

The current preset is indicated in the row near the top of the UI as the darkened one:



When you are in online mode and change any of the processing parameters in the UI, you are changing the settings of the current preset. To switch to a different preset from the preset, just click on the desired



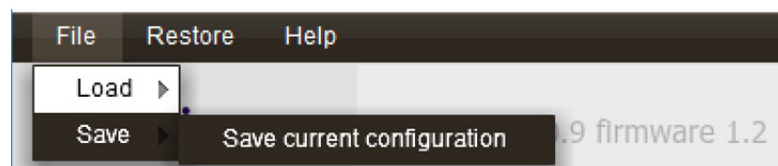
button.

To change the current preset from the front panel:

- Press the button until the LED for the desired preset is lit (1 to 4).
- Press and hold the button for two seconds.
- Release the button.

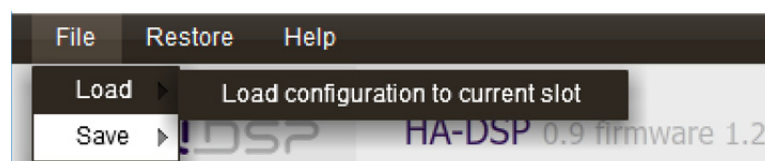
10.1 SAVING AND LOADING PRESETS

To save the current preset to a file, drop down the File menu and select Save then “Save configuration to current slot.” Select the location in the file system and enter the desired file name, then click Save.



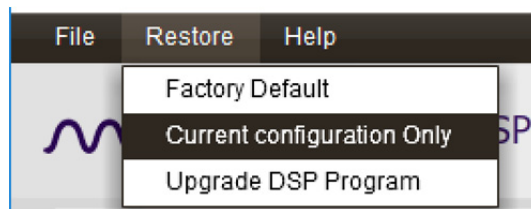
It is recommended that you save your configurations to files to ensure that you do not lose them if you inadvertently reset the HS-DSP to defaults.

To load a preset from a file, first select the preset that you want to load into. Then drop down the File menu and select Load then Load configuration to current slot. Select the configuration file and click Open.

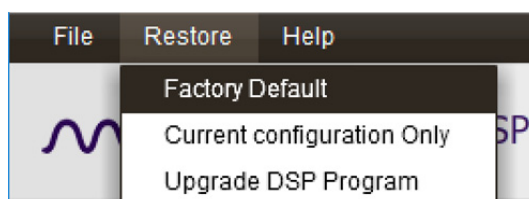


10.2 RESTORING DEFAULTS

To restore the current preset to its default values, drop down the Restore menu and select Current configuration only.



To restore all four presets to their default values, drop down the Restore menu and select Factory Default.



10.3 SYNCHRONIZING OPTIONS

When you click on the Connect button, there are two things that could happen:

- If you haven't changed anything in the UI since you were last connected, the connection will happen immediately and you will be in online mode when the connect button changes to a checkmark.
- If you have changed something in the UI since you were last connected, you will see a dialog asking you if you want to synchronize. Click on the Synchronize Config button to confirm that you want to download the settings from the UI to the HA-DSP.

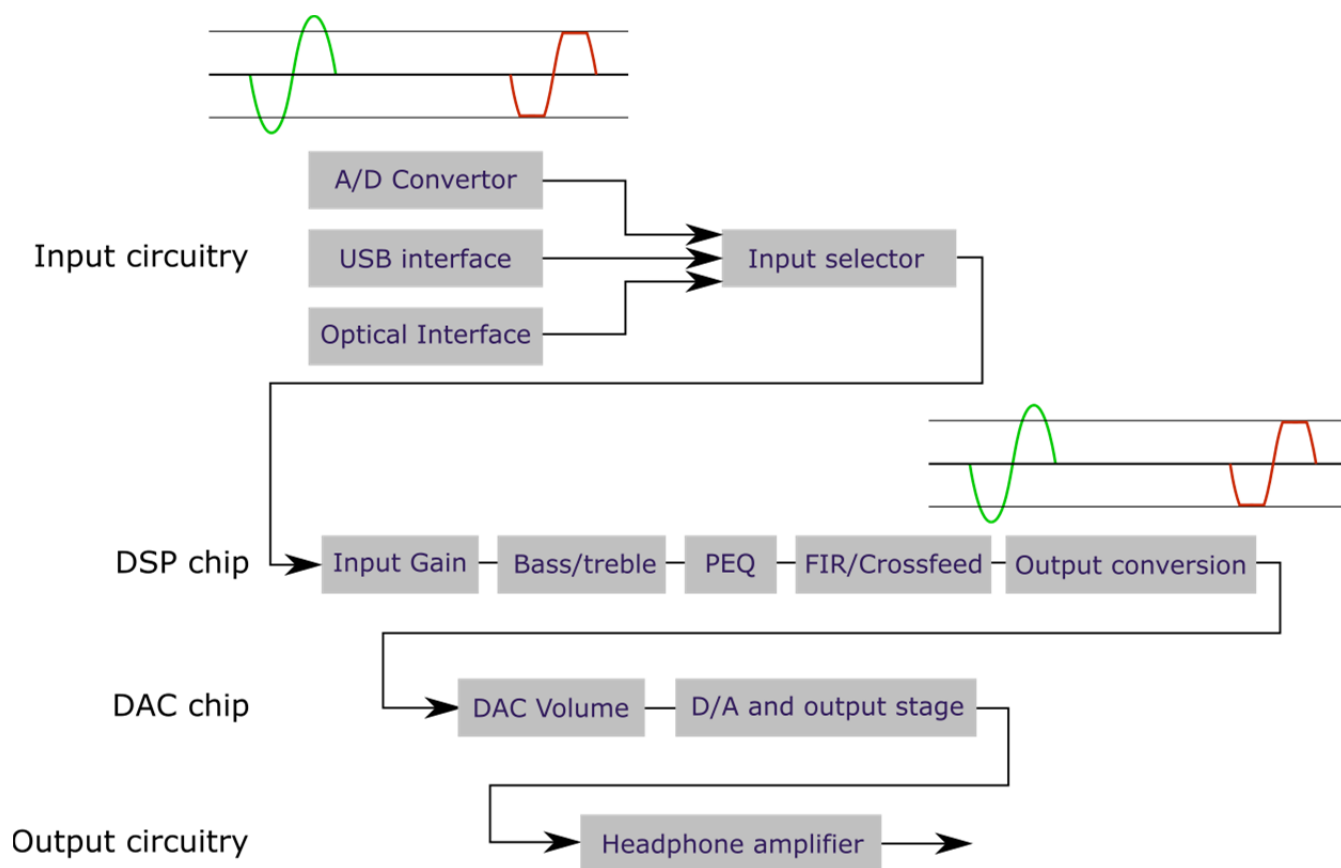


You can instead restore the configuration to its default values (Restore Config button), or cancel the connection request.

Note: you cannot upload the configuration from the HA-DSP back to the UI on the computer. If you have changed settings in the UI but want to revert to a previous version of your configuration, you will need to load it from a file that you saved earlier.

11 AVOIDING CLIPPING

When using digital EQ and DSP in general, it's important to understand the various factors that affect signal level through the processing chain, to avoid clipping (severe distortion) and maintain resolution. Here is the signal processing chain of the HA-DSP:



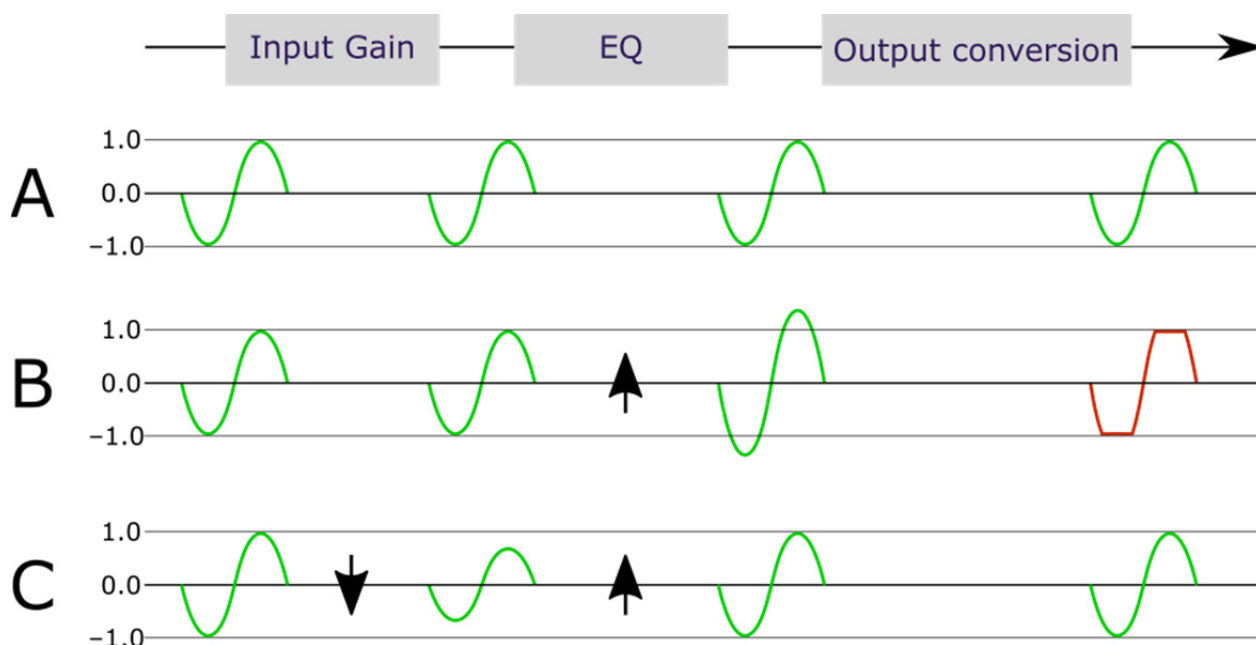
(Note: this applies for normal PCM playback mode. All processing on the line labeled “DSP Chip” is disabled in DSD playback mode.)

There are two places where clipping can occur:

1. In the A/D convertor. The HA-DSP has a maximum input signal level of 1.0 Vrms. If this level is exceeded, clipping of the A/D convertor will result. This can be avoided by reducing the signal level at the analog source.
2. In the Output Conversion stage of the DSP chip. Internally, the DSP uses floating-point numbers, which can be used to represent an arbitrary range of numbers. To send data to the DAC chip, however, the data must be converted to an integer format. If there is internal gain in the DSP processing (because of EQ, for example), then the signal that is sent to the DAC chip may be clipped.

11.1 A SIMPLE EXAMPLE

We'll use a simplified example of the diagram on the previous page. Only the processing in the DSP chip is shown, and the bass and treble controls, parametric EQ and FIR filtering are all lumped into the block labeled "EQ." For simplicity, the signal is just a sine wave.



Case A

The (digital) signal at the input is full scale – that is, it swings between the -1.0 and $+1.0$ limits. At the output conversion, the signal is still full scale, and the DAC chip will output the maximum specified voltage.

Case B

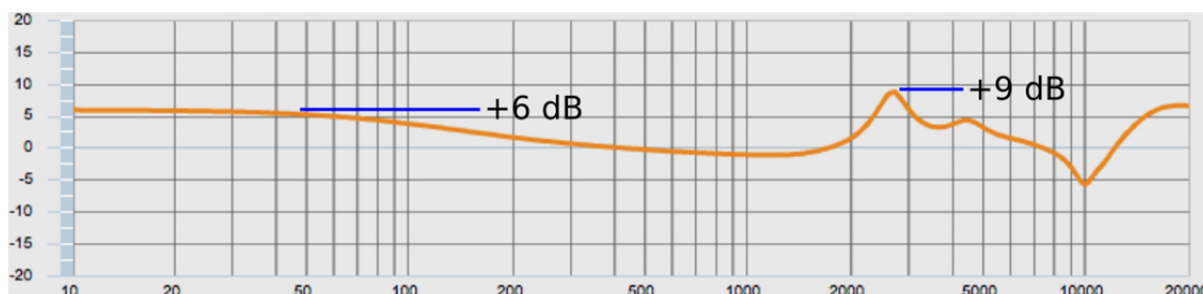
The EQ stage boosts the full-scale signal, so it exceeds the range -1.0 and $+1.0$. This is fine, as internally the DSP has a lot of "headroom" and can represent much larger numbers. At the output stage, however, the signal must be limited to the range -1.0 and $+1.0$. Therefore, it is clipped.

Case C

The signal is scaled down in the Input Gain stage, so that when the EQ stage boosts it, it remains in the range -1.0 to $+1.0$. Again, there is no clipping at the output stage.

11.2 A MORE REALISTIC EXAMPLE

In practice, the HA-DSP is processing a signal with a spread of frequencies, all changing rapidly in level. Consider the example EQ shown on page 23.



With the default unity gain through all DSP processing blocks, this filter will result in clipping on music with a lot of bass content. Assuming this is the only processing (i.e. no bass or treble boost and no FIR filters), we could take two approaches.

Ultra-conservative

Set either Input Gain or Master Volume to -9 dB. This will ensure that no digital input signal (that isn't already clipped) will clip at the output stage. (This approach will always be required in certain situations such as when running full-scale test signals through the HA-DSP.)

Conservative

Set either Input Gain or Master Volume to -6 dB. Since there is typically a lot of energy in the bass region, this will ensure that low frequencies do not cause clipping. The 9 dB peak at 2800 Hz is unlikely to cause clipping with music, as the rest of the spectrum is generally well below this.

In the case where you are using additional EQ, such as Bass or Treble boost, you will need make sure that you click on the Overall Response checkbox to assess the total amount of gain.

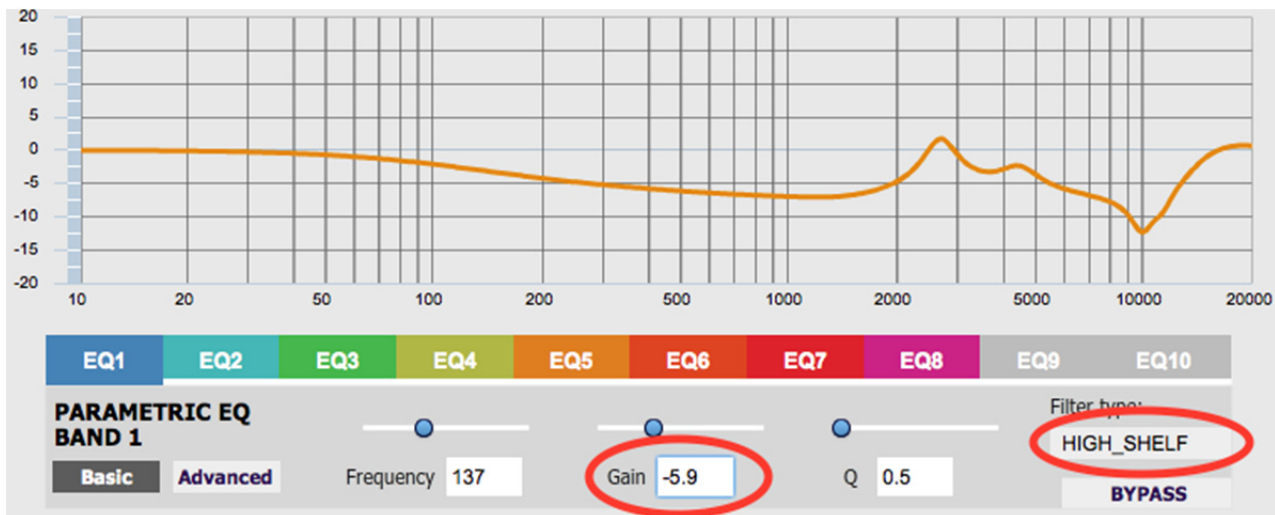
11.3 THE “NO GAIN” APPROACH

An alternative approach to setting input gain is ensuring that your EQ curve is “mostly below zero.”

In one common case where a low-shelf filter is used as a bass boost, the filter can be reversed:

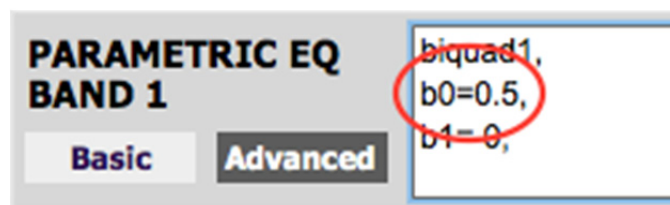
- Change it to a high-shelf filter.
- Set the gain negative instead of positive.

For example, on page 23 the filter shown is a low shelf with gain of 5.9 dB. Here is the same filter reversed as described above:



Note that the whole graph is now shifted down.

A second approach is to use advanced mode of the parametric EQ. Set an unused filter to Advanced mode. To reduce the gain by 6 dB, set coefficient b_0 to 0.5, then press the Process button.



For each halving of b_0 , the gain of the whole PEQ block is reduced by 6 dB. So use the value 0.25 for a 12 dB reduction in level.

12 ADVANCED TOPICS

The functions in this section are for advanced users and headphone EQ developers.

12.1 PEQ ADVANCED MODE

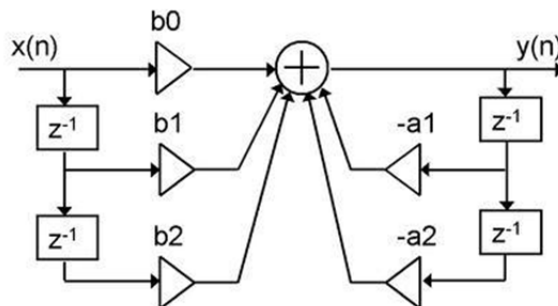
The parametric EQ can use *Advanced* mode, which allows you to allow you to directly provide the low-level parameters aka *biquad coefficients* that control the digital filters of the processor. This allows you to load other types of parametric filters or to import a complete correction file generated by other software.

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

12.1.1 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:

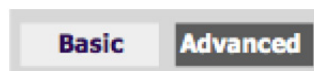
$$H(z) = \frac{b_0 + b_1 z^{-1} + b_2 z^{-2}}{1 + a_1 z^{-1} + a_2 z^{-2}}$$



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: a_1 , a_2 , a_0 , b_1 , and b_2 . These numbers are called the *coefficients*.

12.1.2 Advanced mode for a single filter

To change a single filter, switch it to Advanced mode:



Paste in the desired coefficients, then click on the **Process** button for them to take effect:

EQ1	EQ2	EQ3	EQ4	EQ5	EQ6	EQ7	EQ8	EQ9	EQ10
<div> <div>PARAMETRIC EQ BAND 1</div> <div> <div>Basic</div> <div>Advanced</div> </div> <div> <div>biquad1,</div> <div>b0=0.9997292058283046,</div> <div>b1=-1.9990173425193771,</div> </div> <div> <div>CLEAR</div> <div>IMPORT</div> <div>PROCESS</div> <div>BYPASS</div> </div> </div>									

12.1.3 Parametric EQ file import (REW integration)

Multiple biquads in the parametric EQ block can be set at once by importing a coefficient file. The design program must be set for a **96 kHz** sample rate. 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,  
...  
biquad3,  
...  
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. (Note: 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.

12.1.4 Biquad design software

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

12.1.4.1 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

12.1.4.2 Room EQ Wizard (REW)

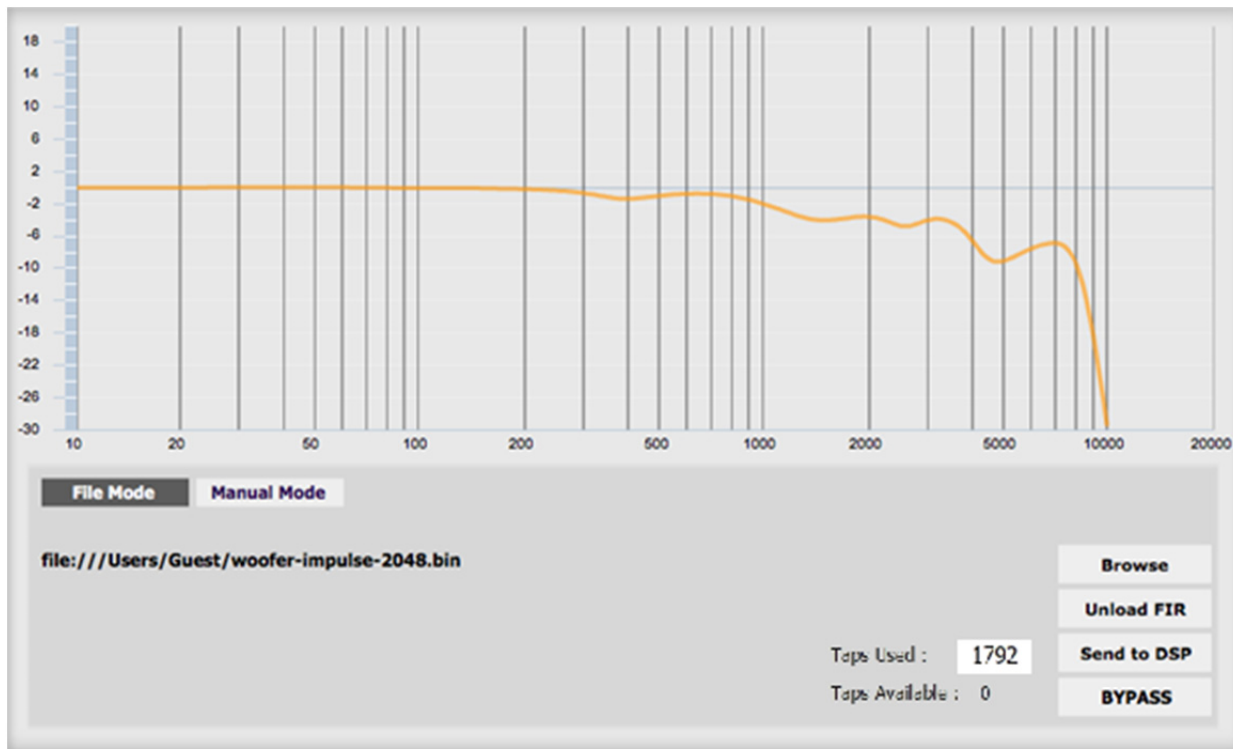
Room EQ Wizard (REW) is a free acoustic measurement and analysis tool, available for Windows, Mac and Linux platforms. It includes the ability to automatically generate a bank of parametric EQ biquads based on a measurement. If you have the ability to measure headphones, you can use REW to generate a set of correction filters. 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](#).

12.2 FIR FILTERING AND ROUTING

FIR filtering is a powerful and advanced feature of the HA-DSP. 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 file browser to select a file containing FIR filter coefficients. (See [Filter file format](#) below).

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

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

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

File Mode / Manual Mode

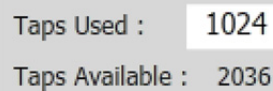
In File Mode, the window displays the **Browse** and **Unload** FIR buttons as shown above. In Manual Mode, the display changes to allow direct text entry of the FIR filter coefficients, as shown below. The coefficients can be pasted into the window from a text editor.



12.2.1 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 HA-DSP can compute a total of 7168 taps. These taps can be distributed as you wish across the four FIR blocks on the output, with the limitation that each output channel must have 6 or more taps and can have no more than 4096 taps. The decision on how many taps to allocate to each channel up to you, and should be determined after working with an FIR filter design program (see below). The number of taps is set in the lower right corner (click on the text entry box and type the desired number of taps, then press **Tab** or **Return**):



Taps Used : 1024
Taps Available : 2036

12.2.2 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.

12.2.3 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. Make sure to select the correct sample rate frequency of the plugin.

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,
```

12.2.4 Loading filter coefficients

In **File Mode**:

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

In **Manual Mode**:

9. Cut and paste the coefficients from the text output of the design program.
10. Press the **Process** button.
11. Confirm that the frequency response graph is as you expect.
12. Press **Send to DSP**. This will write the coefficients into the DSP's memory.
13. 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.

12.2.5 Routing section

The **Routing** matrix mixer is used to direct input channels (Rows) to output channels (Columns). To turn on routing for a crosspoint, click on that crosspoint. The display will change from "Off" to show the mix level (0 dB by default).

The matrix mixer makes the HA-DSP adaptable to many audio processing situations. At each cross-point, the gain of the signal being mixed can be adjusted to a value between -72 and +12 dB. To adjust the gain, right-click on the crosspoint and a gain control will appear. Adjust the gain with the slider, or by typing in the value directly, then click **close**. You can also Invert the polarity at a crosspoint (e.g. L-R). the crosspoint gain value will turn green to indicate an inverted state.



Further information will shortly be added on use case examples for FIR blocks + Routing matrix.

13 SPECIFICATIONS

Computer configuration	Driverless USB 2.0 control interface for Windows and Mac OS X
Chip complement	XMOS xCore-200 processor, ESS Sabre ES9018K2M DAC, Analog Devices ADSP-21489 400 MHz floating-point SHARC DSP, AKM5386 ADC
Audio resolution	24-bit input, sample rate depends on loaded plugin.
USB Audio input	PCM: up to 32-bit input, sample rates from 44.1 to 192 kHz on PC/Mac, PCM up to 96 kHz on select Android devices. DSD: DSD64 and DSD128 (normal and double-rate)*
Digital audio input	Mini-TOSLINK optical connector. 16- or 24-bit, sample rates from 44.1 to 192 kHz.
Analog audio input	Maximum input level: 1.0 Vrms. Signal-to-noise ratio: Up to 100 dB
Analog audio output	Frequency response: 20 Hz to 20 kHz +/- 0.2 dB. Signal-to-noise ratio: 112 dB (32 Ω load, 1 kHz, A-weighted, digital in 0 dB) THD+N: 0.001% (32 Ω , 1 kHz, 65 mW + 65 mW, mid gain) Supported headphone impedance: 16 – 600 Ω Maximum power output: 100 mW + 100 mW (32 Ω , 1 kHz, digital in 0 dB)
Storage/presets	All settings controllable in real time from software user interface. Up to 4 presets stored in local flash memory.
Battery	Lithium Ion 3300 mAh / Up to 10 hours of continuous playback time
Charging	5V 2A (fast charge) USB charger provided
Dimensions (H x W x D)	20 x 69 x 137 mm
Weight	230 g

(*) DSP processor is disabled in DSD playback mode.

13.1 FIRMWARE UPGRADE

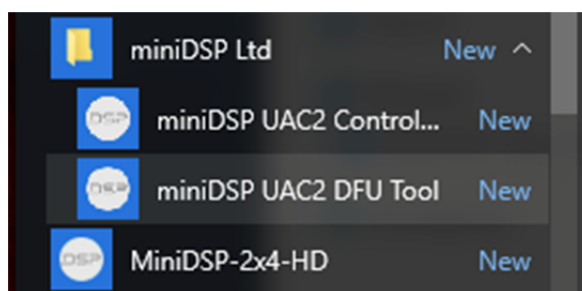
miniDSP may occasionally provide an upgrade to the HA-DSP MCU firmware to enable new features. To upgrade the MCU firmware, first download and install the latest version of the HA-DSP plugin from the **User Downloads** section of the miniDSP website.



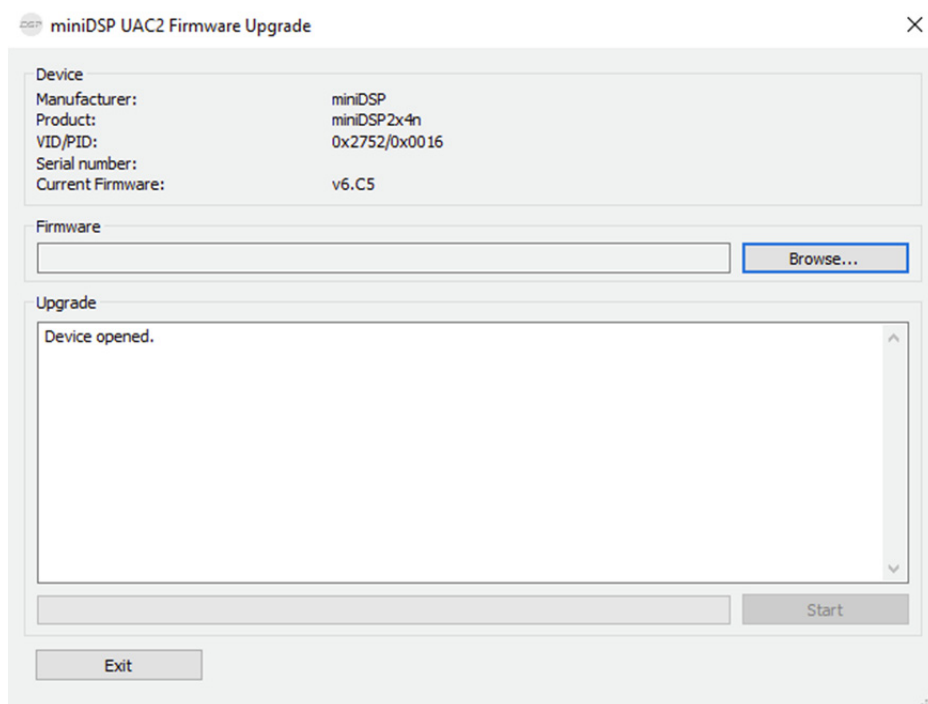
DO NOT DISCONNECT THE USB CABLE OR POWER FROM THE *HA-DSP* WHILE FIRMWARE UPGRADE IS IN PROGRESS. DOING SO MAY “BRICK” YOUR *HA-DSP*

13.1.1 Windows

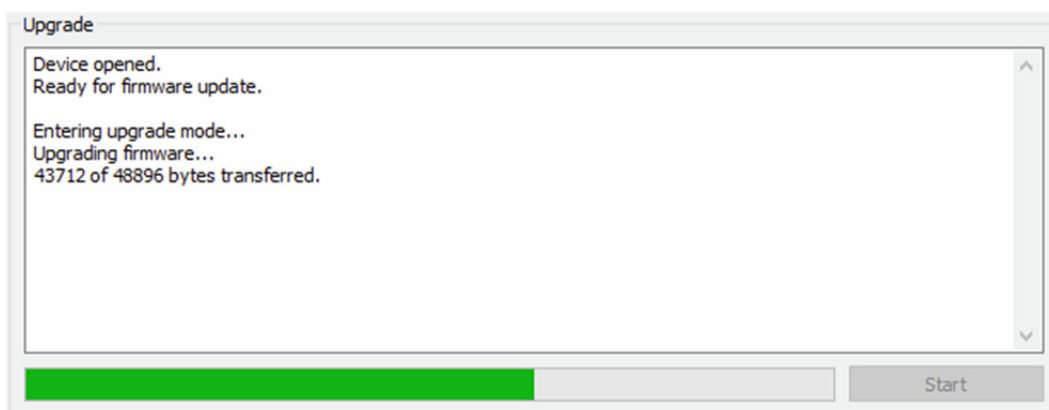
1. Connect the HA-DSP to your computer via USB (if not already connected) and power it on.
2. Start the **miniDSP UAC2 DFU Tool**.



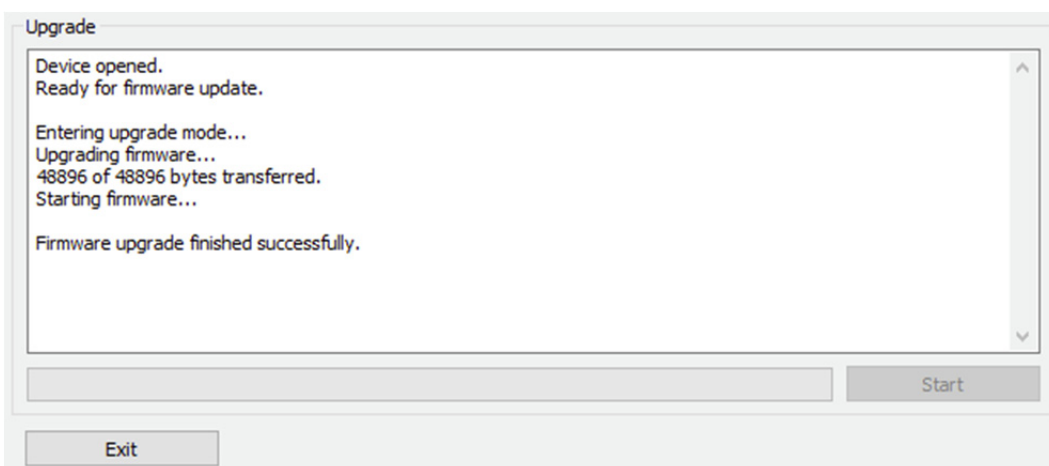
3. The upgrade program will start:



4. Click on the **Browse** button, navigate to the folder **XMOS_Firmware** in the plugin download folder, and select the firmware file. It will have a name like "HA-DSP_vx.xx.bin." (The version number may be different.)
5. Click on the **Start** button.
6. You will get a progress bar as upgrade proceeds:



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



8. Click on **Exit**.
9. That's it! You're done. You can now use your 2x4 HD with the new functionality.

13.1.2 Mac OS X

To load firmware using Mac OS X requires that you use the Terminal program (located in the Applications/Utilities folder). In the examples that follow, black text is the “prompt” printed by Terminal, blue text is text typed in by you, and red text is the program output.



It is important that you type commands as shown including characters like “.” and “/” where noted. Press the Tab key after typing the first two characters of any filename, to activate auto-completion.

NOTE however that the version of the plugin and firmware will change. You therefore need to edit the below commands to the correct version (E.g `HA_DSP_v0_8_drv_4_11` might be `HA_DSP_v0_9_drv_4_11`

Download the latest software for the HA-DSP from the User Downloads area of the minidsp.com website. Double-click on it to unzip it. Assuming that you have placed it into the Downloads folder on your Mac, you will then type:

```
mymac:~ myname$ cd Downloads/HA_DSP_v0_8_drv_4_11/XMOS_Firmware/
mymac:XMOS_Firmware myname $ ls
HA-DSP_v1.0.bin miniDSP_UAC2_DFU_OSX      readme.txt
mycomputer:Firmware myname $
```

You see in the above list the name of the firmware file, **HA-DSP_v1.0.bin** (the version number may change). Now change to the folder holding the actual firmware upgrade tool, and then run it:

```
mymac:XMOS_Firmware myname$ cd miniDSP_UAC2_DFU_OSX/
mymac:miniDSP_UAC2_DFU_OSX myname $ source setup.sh
mymac:miniDSP_UAC2_DFU_OSX myname $ ./xmosdfu --download ../HA-DSP_v1.0.bin
VID = 0x5ac, PID = 0x8007, BCDDDevice: 0x300
VID = 0x5ac, PID = 0x8007, BCDDDevice: 0x300
...
VID = 0x2752, PID = 0x11, BCDDDevice: 0x6e0
XMOS DFU application started - Interface 3 claimed
Detaching device from application mode.
Waiting for device to restart and enter DFU mode...
VID = 0x5ac, PID = 0x8007, BCDDDevice: 0x300
VID = 0x5ac, PID = 0x8007, BCDDDevice: 0x300
...
VID = 0x2752, PID = 0x11, BCDDDevice: 0x6e0
... DFU firmware upgrade device opened
... Downloading image (../miniDSP2x4n_XMOS_v0.13.bin) to device
... Download complete
... Returning device to application mode
mymac:miniDSP_UAC2_DFU_OSX myname $
```

You can now proceed to use the HA-DSP as normal.