Hypex Filter Design manual

Introduction

Hypex Filter Designer is a Windows Software program to control Hypex Electronics B.V. DSP powered products. For example, our FusionAmp plate amplifiers and our MP-DSP Main OEM module. You can set up all needed filters and control all options available. Any knowledge on how to create and design active speakers with digital filters is needed.

This document gives an overview of the options in the Hypex Filter Design software for the Hypex Electronics DSP products.

The Hypex Filter Design software consists of four areas:

- Main screen
- Device settings screen
- Filter design graphical screen
- Filter design advanced screen

The Demo file can give you a better insight of the possibilities in Hypex Filter Designer. This file is only for use as an example.

General Functions

- Volume control
- Channel assignment
- Load/take measurements for speaker (Impulse response)
- Load/apply microphone correction data
- View sum of filters
- Invert filter functions
- Delay
- Graph smoothing
- Wide range of biquad filter setups
- Input gain adapt
- View phase response

Getting started (FusionAmp)

If this is your first Fusion amp, here is a small step by step instruction on how to get your Fusion amp connected to the PC.

What do you need:

- Fusion amp
- Mini USB cable
- PC with Windows OS (Windows 7 or higher)
- C13 power cable

1. Download and install the software from our website.
2. Plug in the USB cable.
3. Plug in the power connector and switch ON the module.
4. All LED's turn on and then off in a sequence, followed by selecting the active preset (By default Preset 1).
5. Start the pc program, Hypex Filter Design.
6. Wait for automatic connection, at “Device info” on the main screen, information about the connected FusionAmp should be given. If there is no response, there could be a problem with the USB connection or your operating system that cannot detect the USB device. For this please refer to the Troubleshooting.

Table of Contents

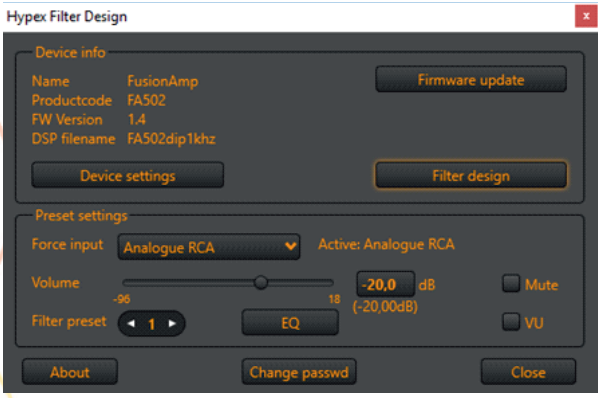
General Functions 02
Getting started (FusionAmp) 02
Main screen 06
  Device info 06
Preset settings 06
  Force input 06
  Auto detect 06
  Volume 06
  Mute 06
  VU 06
  Filter preset 07
  Equalizer 07
  Change password 07
  About 07
  Extra slave option 07
  Volume lock 07
  Source lock 07
Device settings 08
  Preset volume offsets 08
  Input gain configuration 08
  Digital audio channel select 08
  Bridged configuration (CH1/CH2) 09
  Soft-clip limiter 09
Options 09
  Master unit 09
  Soft-clip enabled 09
  Wake on line 09
  Activate signal detection 09
  SUBIN volume is absolute 10
  Shutdown power mode 10
  Volume on start 10
  Standby LED brightness 10
  DAC settings 11
  Defaults 11
Filter Design 11
  Menu bar 11
  File menu 11

Tools menu 12
Clone option 12
Load filter to DSP menu 14
PRESET menu 14
Advanced filter design screen 14
Channel sub menu 14
Biquad sub menu 15
View sum of filters 15
Load measurements for speaker 15
Biquad filters 15
Delay 15
Adding FIR filter(s) 15
Creating FIR Filters 15
Upload filter 16
Designing filters 17
Biquad filter types 18
Graph header 19
Graph left pane 19
Graph right pane (filter graph) 19
Graph bottom pane 19
Filter Objects 19
Adding a filter object 19
Disable or delete filter objects 20
Impulse response graph 20
Step response graph 21
Filter design examples 22
Butterworth 22
Third order low pass Butterworth example (Fc=1000Hz) 22
Fourth order low pass Butterworth example (Fc=1000Hz) 22
Fifth order low pass Butterworth example (Fc=1000Hz) 23
Sixth order low pass Butterworth example (Fc=1000Hz) 23
Linkwitz-Riley 23
For channel 1 (low frequency driver) 24
For channel 2 (high frequency driver) 24
Bessel filter 25
Fourth order low pass Bessel example (Fc=1000Hz) 26
sixth order low pass Bessel example (Fc=1000Hz) 26

Crossover using Bessel 27
All pass 1 28
All pass 2 29
Phase plot 30
Recording 32
Equalizer 33
Copy and write 35
Default EQ 36
Change password 36

Main screen

When a FusionAmp is connected to HFD, the main screen, as seen in picture 1, will show the connected device information and the user can change its volume, source and preset.



Picture 1: Main screen

Device info

When a device is connected via USB, its name, product code and firmware version will be displayed under device info. If a DSP filter settings file has been uploaded, the DSP-file name will be shown as well.

There are a few more parameters with which you can use to personalize your module. To do so, click the "Device settings" button. A new window will pop up. More will be explained in chapter 'Device settings'.

For setting up a filter, delay and gain etc. You can use the filter designer. More will be explained in chapter 'Filter Design'.

Preset settings

Force input

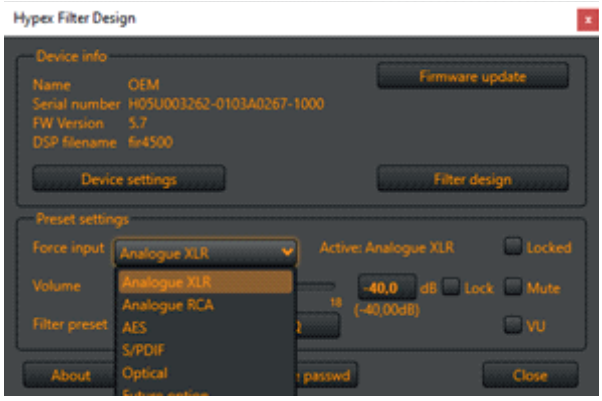
The sources you can select depend on the type of device that is connected to HFD. The sources "Auto detect", "Analogue XLR" and "Analogue RCA" are always available. If your device has high-level capability, the source "SUB" will be added to the list and when your device is equipped with a MP-DSP DIGIN board, the sources "AES", "SPDIF", "Optical" and "Future option" are added. All speak for themselves, but "Auto detect" needs a bit more explanation, see "Auto detect" for more information. "Future option" is an input which is partially available in hardware, but not yet fully implemented.

Auto detect

When "Auto detect" is selected as a source, the FusionAmp will not allow absence of audio for more than 10 seconds. If no audio is present on the current source (you can see what source is selected at the right of "Active:"), the FusionAmp will scan all available inputs for audio in the following priority:

1. AES
2. SPDIF
3. Optical
4. Analogue XLR
5. Analogue RCA

On slave devices the input option "Auto detect" is removed. As slaves are set to follow the same input as the controlling master device, "Auto detect" is not really needed. Master devices still have the "Auto detect" input option available.



Picture 2: Input options

Volume

You can change the master volume in HFD in two ways: with the slider or by entering a numeric value. The volume text beneath the numeric volume input indicates the actual volume. ('Preset volume offset' and 'SUBIN volume is absolute') and added or subtracted to the master volume.

Be careful when changing the volume up! it is very easy to set the volume level to a high level, risking damaging your ears or your hardware. When you are using the slider, it is also possible to change the volume up and down using the arrow keys of your keyboard. This is the safest way to change the volume.

Mute

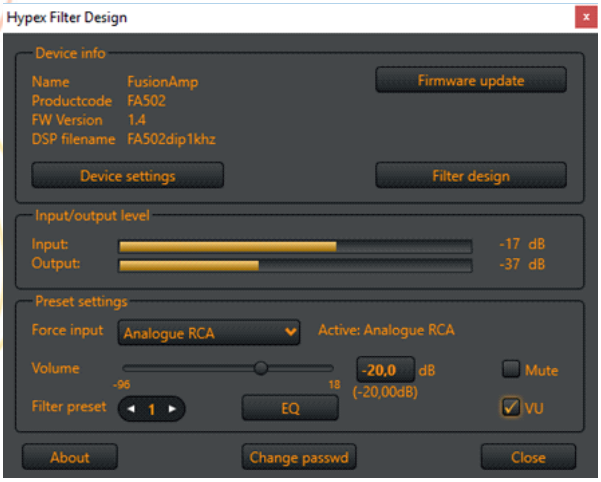
The mute option will mute all channels. The mute is cleared on any volume change or when the mute-checkbox is unchecked.

Picture 3: Device is muted

VU

When a FusionAmp is connected to HFD, it is possible to observe the input and output level

of the DSP. This will give you an indication whether you have the correct input selected and audio is provided to the DSP. This will also indicate if you have set up the DSP with the correct filters and gain. If the DSP has input and no output, it could mean that there are no filter settings uploaded yet, or that the system has low gain or even is muted. Use the VU checkbox to toggle between level indication view and no level indication. By default, this checkbox is unchecked. When checked, the main screen of HFD will expand and look like this:



Picture 3: Main screen with VU indication

The range of the VU indication is -72dB .. +18dB.

Filter preset

It is also possible to select a preset if your device uses multiple presets. Click on the arrows left or right to select the desired preset. Each preset can have his own forced input, filter setting and volume setting. The volume offset for each preset will be explained in chapter 'Preset volume offset'.

Picture 4: Selecting a preset

Equalizer

This button EQ is only available when a FusionAmp is connected to HFD. Please refer to chapter 'Equalizer' for more information about the equalizer.

Change password

To prevent specific functions of the software from being accessed by unauthorized persons you can protect them by setting a password. Please refer to chapter 'Change password' for more information.

About

This window provides more information about the HFD version and the firmware version of the connected device.

Extra slave options

When you have configured the FusionAmp to act as a slave device (see 'Options' in 'Device settings'), two extra settings become available on the main screen:

Picture 5: Extra slave options

The lock functions are especially added for slaves that are connected to a master by means of an S/PDIF cable. The slave receives audio from the master on its S/PDIF source, but also control commands, like ON/OFF command, preset select, master-volume setting or forced source select. Sometimes you want your slave to ignore the source select command or the volume setting command. Both extra options are added to the preset settings, so every preset can have its own source locked and volume locked.

Volume lock

The first extra option of slave units is the volume lock. When checked, the master is not able to change the volume of the slave.

This is useful when your slave's source is connected to an output which has its own level control. This can be a high level (SUB) input for example.

Source lock

A slave can lock its source, so it will not change when the master's source is changed. For example, when the slave is connected to the master with an SPDIF cable, the master source can be switched to Optical, SPDIF or AES, whereas the slave source should always remain SPDIF. An exception is when SPDIF is locked and an analogue source is selected on the master. In that case the slave will follow the master.

The table below (table 1) should clarify which source is active when locking any of the slave sources. To lock the source, check the "Locked" box next to the input selection.

Slave Locked source	Master forced selected source	Slave active source
none	Auto-detect	Auto-detect
none	Analogue XLR	Analogue XLR
none	Analogue RCA	Analogue RCA
none	SUB in	SUB in
none	AES	AES
none	S/PDIF	S/PDIF
none	TOSlink	TOSlink
Auto-detect	Any input	Auto-detect
Analogue XLR	Any input	Analogue XLR
Analogue RCA	Any input	Analogue RCA
SUB in	Any input	SUB in
AES	Any input	AES
TOSlink	Any input	TOSlink
S/PDIF	Analogue XLR	Analogue XLR
S/PDIF	Analogue RCA	Analogue RCA
S/PDIF	SUB in	SUB in
S/PDIF	AES	S/PDIF
S/PDIF	S/PDIF	S/PDIF
S/PDIF	TOSlink	S/PDIF

Table 1: Slave source lock

Device settings

Each module needs to be set up according to the application. On the main screen click the "Device settings" button to open the screen shown below. In this chapter we will explain the parameters, group by group.

Picture 6: Device settings screen

Preset volume offsets

The preset volume is an offset relative to the master volume. This volume can be set for each preset individually. This volume ranges from -24dB to +24dB.

Input gain configuration

The input gain is adjustable by soldering small jumpers on the DSP board. The input gain configuration on screen is just here to visualize that the input gain jumpers are soldered, it will not change the gain itself.

WARNING: This is an advanced modification, soldering experience is required. Please refer to the FusionAmp manual for more information.

Digital audio channel select

For your 2- and 3- way FusionAmp (MP-DSP Digin for OEM users), you must instruct the DSP which audio channel in the digital signal it has to use. By default, it will add left and right for subwoofer applications, but you can change the setting to "Left" or "Right" for stereo applications, so the FusionAmp will output only the left or right signal.

Bridged configuration (CH1/CH2)

On the 2- and 3- way FusionAmps you have the possibility to bridge the first 2 outputs (BTL). If done so, in the DSP only channel 1 is responsible for the output signal. Channel 2 will internally be disconnected and all filter and delay settings you may have done on channel 2 will have no effect. Changing this option will only be in effect when the DSP is restarted. Please use the remote control or power switch to shut down the FusionAmp and switch it back on. This procedure is implemented for safety reasons. After changing the BTL configuration, you also must upload new soft-clip parameters (if you use the soft-clip limiter).

Soft-clip limiter

The DSP in the FusionAmp is equipped with a software limiter which can be used to limit the output power. This limiter will reduce the output power to the set value as soon as the limit is reached. The red LED on the front of the Fusion amp will blink to indicate this.

To set the limiter to suit your needs, provide a maximum output power as well as the impedance of the connected drivers per channel. Power is set in increments of 5Wrms. After setting the proper parameters click the "Update limiter" button to save the settings in the FusionAmp and check 'Soft clip enabled' to enable the limiter.

For BTL configuration the limiter will reduce to 1 channel. Please refer to the "Bridged configuration"section as well.

As per HFD v5.0 it is possible to adjust the effect of the limiter. The limiter on the FusionAmp is solely intended for protection of your drivers. Please note that when the protection is limiting the audio output (that is, when the red led named "Prot" flashes on the plate), it will have a negative impact on the audio quality. Both settings, Attack and Release, are approximate and therefore not very accurate.

The faster Attack is set (0.1msec fastest), the quicker the output will be limited to the maximum power level. But notice that a very fast Attack setting will act like a waveshaper. We found that 0.8msec produced the best overall hearing results. Longer Attack times will initially produce higher output levels (above the maximum power setting), which in turn could damage your driver in case of a long time overload.

With Release you can control the time the protection will stop limiting the output after being active. The longer the release time, the better the audio will sound. Short Release times will probably give some audible distortions. The best performance on the FusionAmp is gained with a Release setting of 99.9msec.

Options

Master unit

The FusionAmp can be used in a multi-device setting, for example a stereo setup. If you connect the two devices (using SPDIF) and want to be able to adjust the master volume on both devices, you must set one FusionAmp as a master and the other as a slave.

The master FusionAmp usually has a remote set attached, adding the ability to control the volume and choose a source or preset with a RC5 remote control. Under "Options" check "Master unit" if the connected device is a master, uncheck it when this device is connected to a master.

Important: when mains is switched for both master and slave, be sure that they are powered at the same time, or power up the slave first and then the master. If the master is started before the slave unit is powered, the slave may not receive commands from the master.

Soft-clip enabled

When checked, the soft-clip limiter is active and will limit the outputs according to the given power and impedance you have set earlier under 'Soft-clip limitter'.

Wake on line

When checked, the FusionAmp will wake up immediately after mains power is applied.

When unchecked, the FusionAmp will go into standby when it is connected to the mains until it is triggered to wake up.

Activate signal detection

The FusionAmp has two power-down modes: "Standby" and "Low power" (See 'Shut down power-mode'). When the device is in any power-down mode, it can wake-up by an analogue signal on the XLR or RCA input. If the device is in "Low-power" power-down mode, it can also wake up by detecting audio on any digital input. If you want to enable the signal detection, set it to "checked".

When signal detection is enabled, it will also automatically enter the selected power-down mode after a certain time of no audio, by default this is 15 minutes.

As of HFD version 4.95 and FusionAmp firmware 1.41, it is possible to set the level of sensitivity of the audio detection. Four levels are available, range 1..4

Level 1 is the default value and the most sensitive one, it will trigger the FusionAmp to switch ON when the audio-level is about -60dBand will trigger to switch off when the audio level has dropped below -70dB.

If your FusionAmp tends to switch on unexpectedly or will not turn off automatically when you have signal detection enabled, you may try to change the sensitivity. Level 4 is the least sensitive one. Each level is about 5dB different from the other. The sensitivity level is only available for the analog inputs to trigger the FusionAmp to switch ON. The digital inputs are fixed at -60dB sensitivity.

It is possible to set a different level for switching ON and for switching OFF, but do not select an OFF level that is less sensitive than the selected ON level, because in that situation, your FusionAmp will shutdown at a certain point in time and could start again immediately due to the fact that it's starting sensitivity is higher than its shutdown sensitivity. HFD will make sure that Level OFF is less or equal to Level ON.

The OFF-sensitivity level also applies to the "auto-source" detect. This implies that it is possible to change the detection level for scanning for audio. When audio drops below the selected level. For the detection of digital audio, a fixed level of -60dB is set.

SUBIN volume is absolute

(1-way FusionAmp or MP-SUBIN only)

Picture 7: Subin volume setting

By default, the volume control knob on the 1-way FusionAmp (MP-DSP Subin for OEM users), can offset the master volume by -12dB to +12dB. If you want to control the volume across the entire range (-96dB to +18dB) using the control knob, select this option. **But use with care**! If checked and the knob is set to +18dB, the FusionAmp may produce maximum power and damage your drivers and/or ears. Note: When enabled, the master volume setting is ignored.

Shutdown power mode

As already mentioned in 'Activate signal detection' the FusionAmp knows two power-down modes: "Standby" and "Low-Power". In "Standby" power mode, the device draws less than 0.5W from the mains power supply. The only wake-up method in this mode is audio on the analogue XLR or RCA (or SUB input for 1-way FusionAmp) inputs when signal detection is enabled (see 'Activate signal detection'), or with a remote control, if your device is equipped with a remote control set. The sensitivity is user selectable in four levels. If audio is detected for at least 0.5s, the unit will start up.

When the device entered "Low-power" mode, the Fusionamp (2- and 3-way only) will also scan the digital inputs. When audio is detected on one of the digital sources (equal or more than -60dB for at least 0.5s), the system will start. In low power mode, the FusionAmp will consume less than 1.5W.

Volume on start

The FusionAmp can apply a default volume after the unit starts. That way, you will always have a known volume level when you switch on the FusionAmp. You can enter a volume between -96.0dB and +18.0dB. Leave this option unchecked if you want your FusionAmp to start with the last selected volume.

If the FusionAmp is a 'master', it will send its start-up volume to any slave connected, during the start-up sequence.

If communication fails during startup, the slave will have a default volume of -96dB.

Standby led brightness

If you have the "front" option connected to the FusionAmp (this is a small pcb with a bicolor led and an IR sensor, which can be purchased separately), the brightness of the red led on this pcb can be set to "OFF" or 25%, 50%, 75% or 100% brightness.

If you have connected the "display" option, instead of the "Standby led brightness", a different setting will appear:

The brightness of the display can be set between 0 and 15. 0 is very dimmed, 15 is maximum brightness. The value under 'max'

represents the brightness the display uses when it is showing a new setting or alert (default 15). After a few seconds (if a new setting was showed), the display will refresh and show a logo. At the same time, the brightness will be the one which is selected under 'dimmed' (default value 6).

DAC filter settings

As of firmware version 1.3, it is possible to control the internal filters of the DAC. Please refer to the datasheet of the AK4454 for more information. You can change its filter type, sound quality (1..3) and FIR settings (0..7).

*Note: *For advanced users only. Please leave these settings in their default position unless you understand what they do.*

Defaults

All settings can be set to their initial values for your convenience. After a confirmation dialog, the default values will be written immediately to the connected FusionAmp. It is wise to do so when volume is set to a low value. Beware that the soft-clip limiter will be disabled!

Filter Design

This part describes the Filter Design tool of HFD. More information about specific modules can be found in the manuals and/or in case of the FusionAmp, take a look at the 'Fusion amp device settings'.

Filter design has two screens: one advanced screen (which represents the internal of the DSP and where biquads can be sorted) and a graphical screen, where filters are created using impulse data and adding biquads).

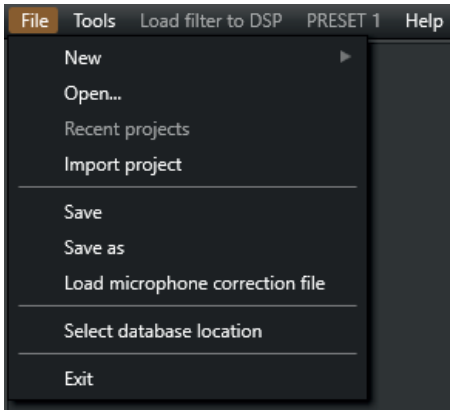
When creating a new project, HFD always opens the graphical interface first. Use the button in the bottom-right corner, labelled "Advanced" to switch to the advanced screen. You can switch back to the graphical screen with a button on the same spot, labelled "Back to filter design".

Menu bar

There are four items (not all always enabled) at the top menu of the screen:

- File
- Tools
- Load filter to DSP
- PRESET

File-menu

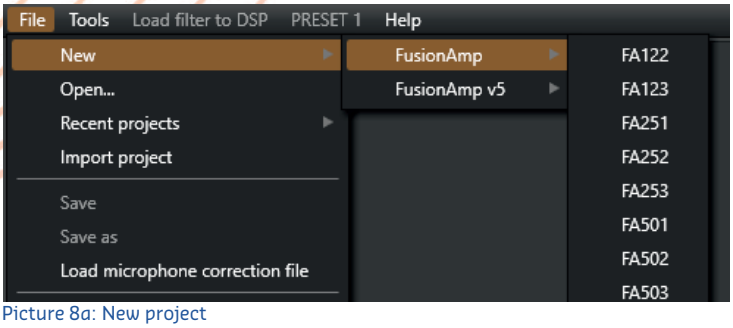


Picture 8: File menu

The file menu is mainly for open/save/create projects and load/unload microphone correction datafile. When a device is connected via USB to your computer, you can only create a new project for that device (in the example DSP2W). Projects are stored in your profiles "Documents" directory under "FilterData". A project consists of one Config.xml file and datafiles containing impulse response data.

Projects created with earlier HFD versions can be imported with the "Import project" button.

When your measurement microphone's manufacturer offers a microphone correction file (*.txt/*.dat) you can load it here. It will be loaded the next time you start HFD automatically. Please keep that in mind and check if a microphone correction file is loaded when starting a new project.



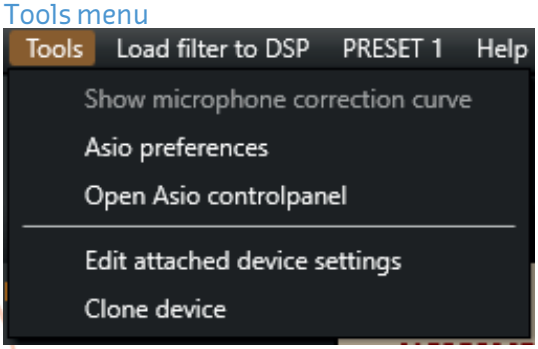
Picture 8a: New project

As per HFD v5.0, the only supported devices are FusionAmps. When no FusionAmp is connected to HFD and a new filter design project is started, two options are available:

- FusionAmp, v1.x
- FusionAmp, v5.x and up

All FusionAmp v5.x and up are FIR capable.

Tools menu



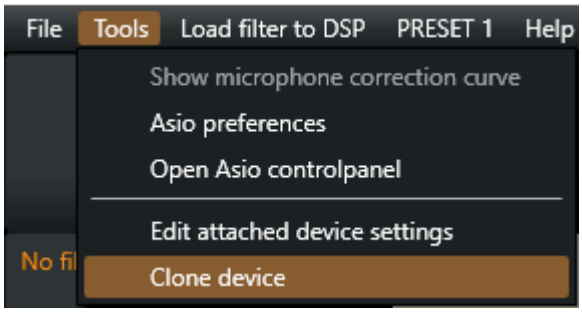
Picture 9: Tools menu

The tools menu is used to select your input/output hardware to generate sweeps and measuring incoming audio to create your own impulse response files. Use ASIO drivers to avoid the Windows latency. If your hardware vendor does not supply ASIO drivers you can use drivers from http://www.asio4all.org/. Also, you can hide or show the loaded microphone correction curve on the graphical filter design screen or edit device settings realtime when a device is connected to your computer.

Asio preferences will open a new dialog in which you can select the audio device, input and output channel of that device to configure your hardware and be able to take measurements in the graphical filter design screen.

Picture 10: Set Asio preferences

Clone option



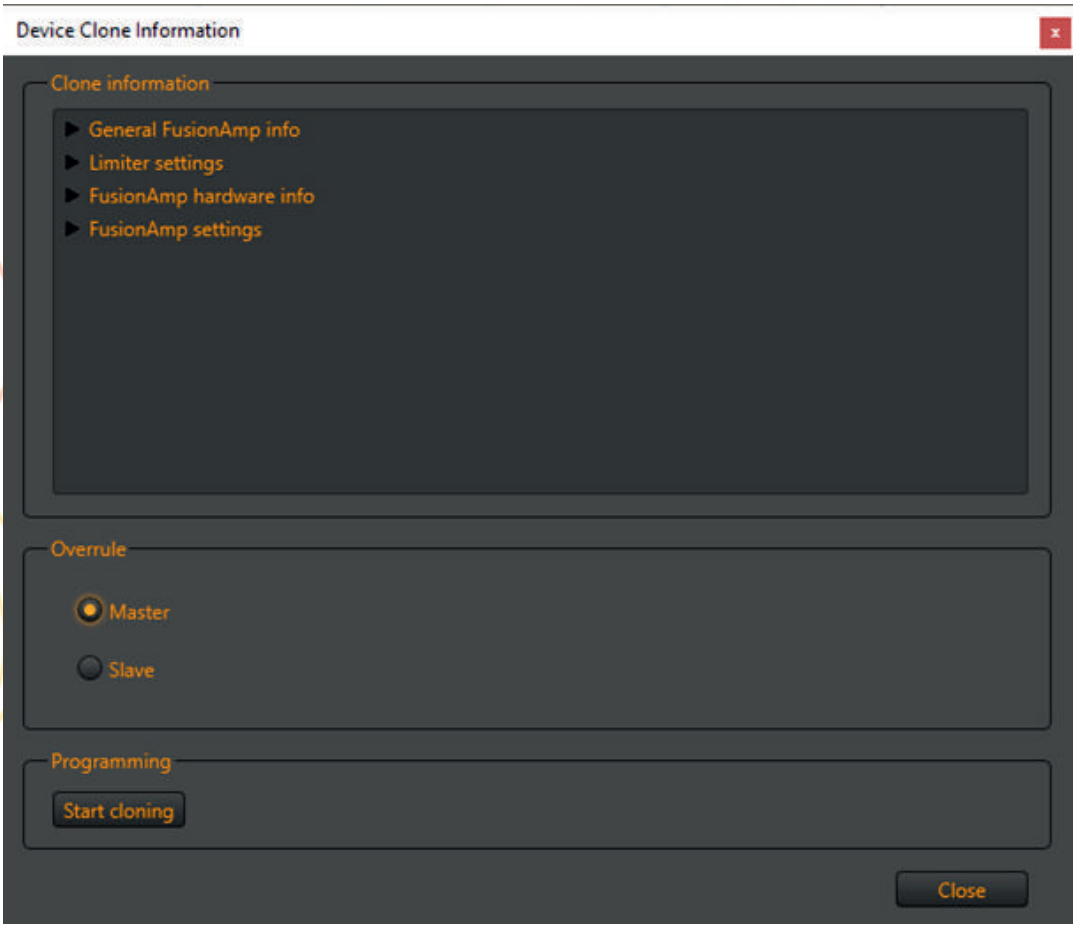
Picture 10a: Clone device

In a stereo setup both the left and right FusionAmps will almost always have the same settings and filters. Or, in case of mass production, one filter design can be transferred to many different FusionAmps. The "Clone device" option can make programming FusionAmps a lot easier as different plates (although they must be of the same module type!) can be programmed with the same project file.
This is accomplished by following these steps:

1. Connect a FusionAmp to HFD which is used as a template.
2. Setup the device settings as needed.
3. Setup the equalizer for all three presets as needed.
4. Open "Filter design" and create a new filter for this plate, or load a project with filter settings.
5. Check if all presets are setup as intended.
6. If needed, upload the presets to the connected FusionAmp.
7. Save the project (this will store all settings, including eq and filters).
8. Disconnect the FusionAmp and connect another of the same type.
9. Under "Tools", the option "Clone device" should again be available.

Once a project is available that is produced the way described above, it is also possible to connect FusionAmps at a later moment. In that case, start HFD, connect a FusionAmp that should receive new settings, open "Filter design", open a project that was created the describe above and the "Clone device" under Tools should be available.

Hit “Clone device” to open the cloning screen:



Picture 10b:Device Clone Information

In this screen, only a few options are available. On top, in the box “Clone information”, all settings that will be sent to the connected FusionAmp can be reviewed. Beneath it, in the “Overrule” box, it is possible to set slave options if one wants to clone a master unit to a slave. The extra slave options are the same as you will find on the main screen when a slave unit is connected:

Picture 10c: Clone Slave Options

This way, you can easily clone a master into a slave, but it is of course also possible to make a clone project of a slave. When loaded, that project would automatically show the above settings.

When all looks fine, hit the Start cloning button to start the cloning process. This will take a little while. After cloning is completed, a message will show the result.

Load filter to DSP menu

Picture 11: Load filter to DSP menu

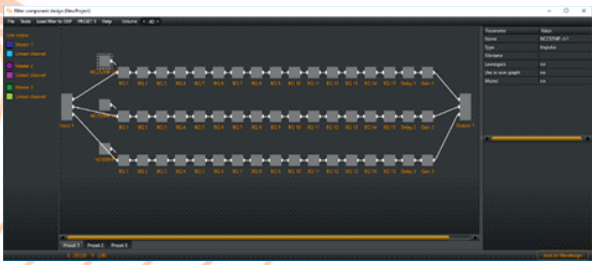
Filters can be uploaded to a device for each preset, or you can upload all presets in one click. **If you choose to upload all presets, please make sure you implemented cross-over filters to protect your drivers!** Select a preset with menu PRESET, edit its content if you want to and upload the preset to the attached device, or upload all presets.

PRESET menu

Picture 12: Preset menu

Select a preset if your device has preset-capability. Also, you can copy the current selected preset to another preset. That action will only overwrite all filter settings of the destination preset. Settings done on the main screen or on the device settings screen are preserved.

Advanced filter design screen



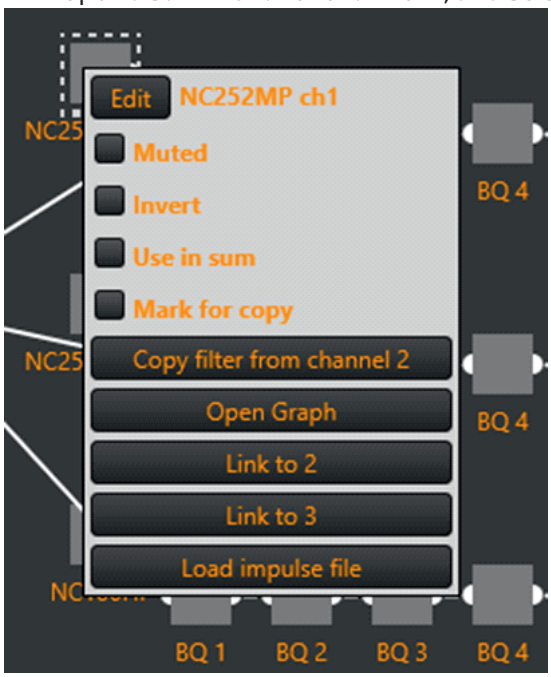
Picture 13: Advanced screen overview

In the advanced screen you can:

- Double click on the box labelled "NC252MP ch1" to open the graphical interface.
- Right click on the box labelled "NC252MP ch1" to open a sub-menu.
- Select a biquad by left-click a biquad filter.
- Select multiple biquads by pressing the Ctrl-key and left click biquads.
- Exchange biquads (sort) by selecting exactly two biquads on the same channel and right-click on of them to open a sub-menu.
- View biquad filter parameters (at the right top).
- View biquad response (at the right bottom).
- Link channels (three slaves maximum)
- Select preset to work on.
- Upload filters to connected device.
- Copy a complete channel to another.

Channel submenu

After right-clicking the box labelled "NC252MP ch1", you open a sub-menu with options for this channel. Of course, when you do the same action on the box labelled "NC252MP ch2", it will open a sub-menu for channel 2, and so on.



Picture 14: Right click on the channel box

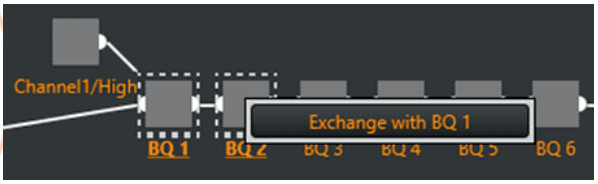
Here you can:

- Alter channel properties open the graphical interface (same as double-click the channel box).
- Link this channel to another channel.
- Load an impulse datafile (measured response).
- Set/reset mute for this channel.
- Set/reset invert.
- Set/reset use channel in sum response.
- Mark a channel if you want to copy. biquads from that channel to another.
- Copy biquads from a selected channel (button "Copy filter from channel X" is only visible when a channel is marked for copy).

- Edit the name of channel in HFD (click "Edit" and change the name in the newly opened dialog).

Biquad sub-menu

When you select exactly two biquads on the same channel, you can exchange their parameters by right clicking on of the two and click on "Exchange with ..."



Picture 15: Right click on one of two selected biquads

This feature can come in handy when you want to optimize the DSP calculations for your filter. Build your filter in the graphical section and afterwards, sort biquads with this option.

View sum of filters.

By selecting the "show sum" the user is able to see the full filtered channels. This function is only available for channels with a response loaded.

click the most left channel-box and click "Load impulse file".

Load measurements for speaker

You can load the impulse file on either the Graphic filter design screen, this can be done by selecting the channel you want to import and click "import", or on the advanced screen, right click the most left channel-box and click "Load impulse file". The filter designer expects the impulse response measurement as a text file with one sample per line. There is no restriction on the absolute gain of the impulse response data. The only thing that matters is that the absolute gain be the same for all three measurements. The filter designer computes a gain offset based on all loaded responses to centre them collectively on the vertical scale.

Biquad filters

To set the filtering you ought to know about the available biquads. There is a full list of the available filters in the datasheets.

Delay

The delay per channel can be selected under common settings. The reason why the delay takes bigger steps, for the AS2.100 approx. 21us, is because of the sample rate. We can only delay per whole sample, so the step count depends on the sample rate.

Adding FIR filter(s)

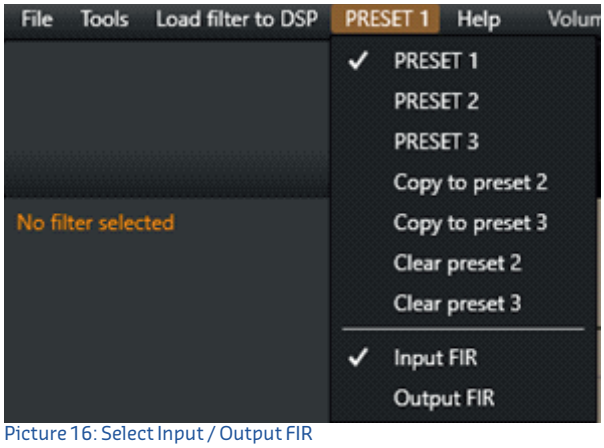
FusionAmp firmware v5.7+ has DSP options for implementation of FIR filters. There are two places where the FIR filters can be implemented:

1. Right after the input multiplexer, just before the 15 biquads.
2. After the 15 biquads.

In case of 1: One 4500 tap FIR filter is inserted at the beginning of the audio processing path.

In case of 2: Each channel has its own 1500 tap FIR filter.

To choose between 1 or 2, select either "Input FIR" or "Output FIR" in HFD under the "PRESET" menu:



Picture 16: Select Input / Output FIR

The selection is situated under "PRESET", because each preset can hold its own FIR filter position. It is not possible to mix input and output FIR filters. By default, presets have input FIR filters selected.

Creating FIR filters

FIR filters need to be created with an external application. Please use 93750Hz as the samplerate. That application should export coefficients formatted as floating point numbers, one on each line in a text-file. Such a file can then be selected by right-clicking on the FIR component in the routing overview:

Picture 17: Routing overview

In the above picture, right-click on

and select a valid text file with exactly 4500 lines which contain only 1 floating number per line. A few lines of a file can be like this:

```
0
0
0.000000000000000000191444222693412
-0.000000000000000000747345703533892
0.000000000000000000167927999145463
-0.000000000000000000303050172238464
0.000000000000000000483429188881767
-0.000000000000000000710401375140 36
0.000000000000000000935918088324 2
```

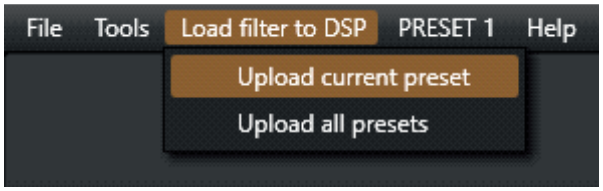
Picture 18: Text file content example

HFD tries to convert any number to a floating point number, as long as it only contains one decimal point or comma, but not both! There are a few FIR examples in the installation folder under "Sample files".

NOTE: FIR filters are an addition to the filter project, so before uploading a project, please load an existing project and add a FIR filter, or add biquads and/or delay to your discretion before uploading the preset, because it will overwrite the existing preset in the attached device.

Upload filter

After selecting a FIR file, the project must be uploaded to the DSP. To do so, select "Upload current preset" from the menu:



Picture 19: Upload project to DSP

After uploading, by default, the FIR filter is bypassed, so to enable it, just hit

with your left mouse button to toggle the state between 'enabled' (green) and 'bypassed' (red).

FIR filters at the input always have 4500 elements, adding a constant delay of 24msec to your audio path. When using separate FIR filters on each channel (1500 taps), the delay per channel will be 8msec.

Picture 20: Routing overview

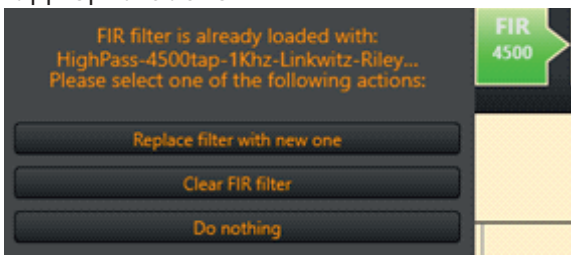
TIP: Hovering the mouse over the "FIR" part, will show a tooltip with the filename.

TIP: Presets uploaded to the FusionAmp will also store the filename of the FIR filter, so give your FIR filter file a name representing the contents, so you can remember what kind of filters you implemented in the FIR.

NOTE: To create FIR filters, please use 93750Hz as the samplerate, as this is the rate at which the DSP processes the audio.

Also note that for input FIR filters you MUST create a filter with exactly 4500 taps and for output FIR filters exactly 1500 taps.

When a FIR filter may be removed, or changed, just right-click the FIR option and select the appropriate action:



Picture 21: Select action for existing FIR filter

Designing Filters

Use the graphical design screen to build your filters.

The first step is equalising the magnitude responses of the drivers flat over their entire useable frequency range.

To do this you can use shelving filters, and boost/cut sections. A sharp peak followed by an equally sharp dip can be corrected using a second-order shelving filter with a high Q.

Exercise care when deciding what to correct. When correcting for diffraction errors, do not exceed a Q of 3 because this could make it worse. Errors that are caused inside the driver, or internal cabinet resonances that emanate through the same diaphragm, may be corrected ruthlessly - provided the measurement has sufficient resolution to pin them down.

As a rule of thumb, sharp dips are diffraction artefacts while sharp peaks are caused by the drivers themselves. Exceptions are room resonances (if the response is not correctly truncated) and diffractions on repetitive patterns.

The second step is designing the actual crossover filters. All the usual strategies work. Delaying higher frequency drivers with respect to lower-frequency ones is a powerful alternative to using asymmetric slopes and yields substantially improved coherence through the crossover region.

Picture 22: Graphical filter design screen

Biquad filter types

Biquad function	Parameters	Use
Unity	-	Section is not used
Lowpass 1	Cut-off frequency (always -3dB)	First order lowpass
Lowpass 2	Cut-off frequency (asymptotically) Q	Second order lowpass.
Highpass 1	Cut-off frequency (always -3dB)	First order highpass
Highpass 2	Cut-off frequency (asymptotically) Q	Second order highpass
Shelf1	Centre Frequency (halfway point) Gain Direction	First order shelf. Useful for baffle-step correction
Shelf2	Centre Frequency (halfway point) Gain / Q /Direction	Second order shelf.
Asymmetric Shelf	Pole frequency and Q / Zero frequency and Q	Equalising the bottom end of closed - box woofers with large magnets
Boost/Cut	Centre frequency / Q /gain	Dip/peak filter.
Allpass1	Centre frequency	Phase correction
Allpass2	Centre frequency / Q	Phase correction

Table 2: Biquad filter types

The graphical filter design screen has four main parts (see picture 17):

- Header (1)
- Left pane with filter object parameters, channel parameters, general settings and recording options (2)
- Right pane with graphical representation of the selected channel(3)
- Bottom pane with tabs for Magnitude, impulse, Step and Phase graph, impulse load/clear/save buttons and cursor position (4)

Picture 23: Graphical filter design screen main parts

Graph header

The header shows what channel is currently selected (highlighted bar). To select another channel, just click on the channel name. Also, it displays the gain and delay of all channels. Each channel can be muted/unmuted by clicking the "Mute" box in the header. In the same way, channels can be added to the sum-response by clicking the "Sum" box. Red means, the channel is excluded from the sum-response, a green "Sum" box indicates, the channel is part of the sum-response.

Graph left pane

When you select a filter in the graph, its parameters will show up in this pane at the top. Below the filter object parameters, the channel parameters are presented. Here you can edit the channel gain, channel delay and channel invert option. Also, the channel name and output name can be changed, although these changes will be added to your project, they are not uploaded to a dsp, so the names are more for your convenience. In the channel parameter box, the number of remaining filter objects is displayed and when you have recording hardware installed on your computer, a recording button will be visible.

At the bottom of the left pane, general settings are presented like smoothing and measurement sampling rate.

Graph right pane (filter graph)

In this graph, you can view and/or change the response of the selected channel. Add/Disable/Delete filter objects, show/hide impulse responses, show/hide sum response.

Graph bottom pane

Select the magnitude/impulse or step graph, import/clear/save channel impulse response, view cursor position in the magnitude graph or switch to the advanced screen by clicking "advanced" button.

Filter objects

Picture 24: Filter graph

In the Magnitude graph, the thick orange line represents the result of the filter sections on the measured response. The red line represents the measured response (impulse response). You can show or hide the measured speaker response by checking or unchecking the checkbox "Show" at the bottom of the screen, or by checking/unchecking the box labelled with the channel number at the top of the graph (in the example "1"). Lowpass filter objects are always coloured blue, high pass filter objects red. A selected filter object is green. When more filters are placed on top of each other, the colour will be dark red. All other filters are orange coloured. It is possible to temporary disable a filter, just to hear the effect after uploading the filter-parameters to the attached DSP. These filter objects are drawn Grey.

When a filter object is selected (by left-clicking its circle), the response of that particular filter is drawn as an aqua coloured line. When no objects are selected (by left clicking somewhere in the graph, not on a filter object) the effect of all enabled filter objects is drawn as a single aqua coloured line.

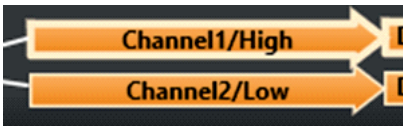
When channels are added to the sum, the sum-line can be displayed by checking "Show sum of all selected channels" at the right top of the graph. The sum is drawn in colour black.

Impulse responses of other channels can also be added to your graph. The will be drawn in a very light purple colour when enabled by checking the box corresponding to its channel number at the top of the graph (in the example channel 2 would be an option, but it has no impulse response loaded, so checkbox "2" is disabled).

You can scroll the Y-axis by left clicking and hold somewhere in the graph (NOT on a filter object) and dragging the graph up or down.

Adding a filter object

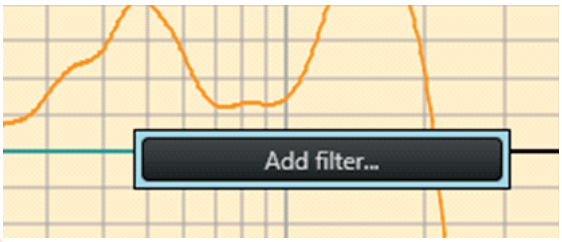
Select a channel by clicking on the channel name in the channel header section (top of the screen):



Picture 25: Select a channel in the graphic screen

In this case "Channel1/High" is selected (its surrounded by a thick bright line). To add filters to the selected channel, right-click in the

graph at the frequency-position you want the biquad. A popup menu will appear:

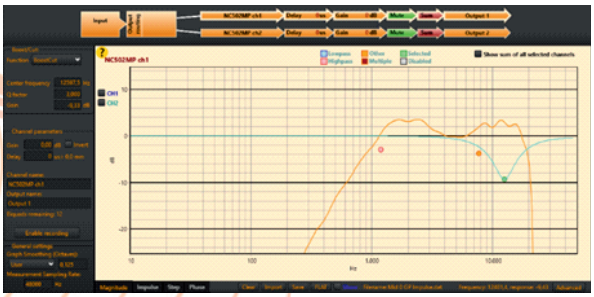


Picture 26: Add a filter object

Click "Add filter..." to open at the left side of the screen the filter properties settings:

Picture 27: Select the filter object function

Click on the down arrow of the Function box to select the desired filter function. This must be done before any other action, otherwise, the adding will be cancelled. Select for example a "BoostCut":



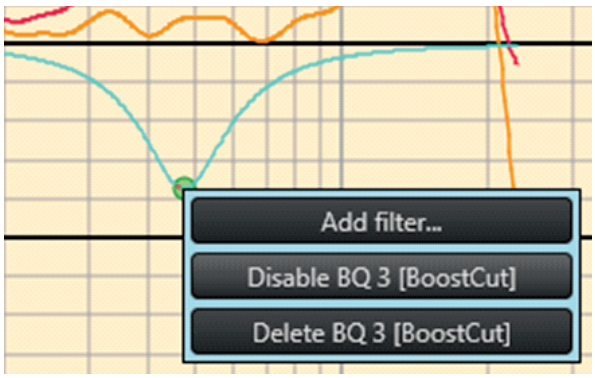
Picture 28: Filter object curve

After designating its function, you can change the parameters by hand (that is, entering a frequency, Q-factor and gain), or with your mouse. Left click the green circle and drag it to where you want it to be. Use your mouse wheel to change the Q-factor. The mouse-wheel is also handy when hovering over an edit box in this screen. When your mouse is over the "Center frequency" box and you rotate the mouse-wheel, the frequency will change. The same will happen if you do this when hovering over the Q-factor, gains and delay.

While dragging the filter, you can change its frequency and gain. If you rotate the mouse-wheel while dragging, the Q-factor will also change.

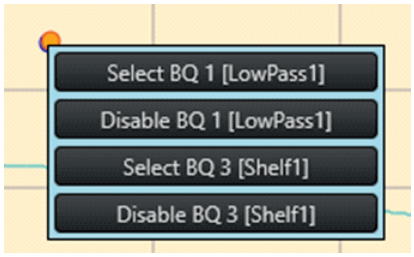
Disable or delete filter objects

To disable or delete a filter object, select the corresponding circle, right-click to open a sub-menu:



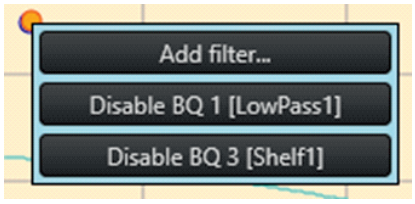
Picture 29: Disable or delete a filter object

When more than one filter are placed over each other, the menu will contain buttons for the other objects too. Left mouse-click:



Picture 30: Left mouse click

Right mouse-click:



Picture 31: Right mouse click

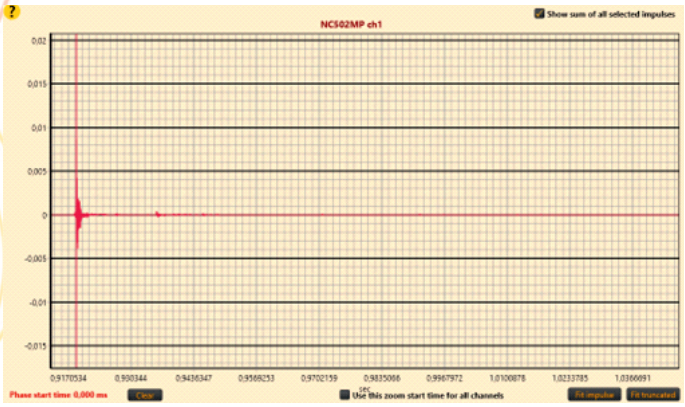
NOTE: a disabled filter object will not be uploaded to the DSP. You can use the disable function to temporarily remove the filter object from your channel to observe its effect.

Impulse response graph

Select the impulse response graph, by clicking tab "Impulse" at the bottom pane. It represents the loaded impulse data. Scroll through the X and Y axis by left click and hold somewhere in the graph and start dragging. Also, you can zoom in on a part of the curve.

Use the middle mouse button (or press the mouse-wheel) and draw a rectangular box around the part you want to show (starting from the left side to the right). When you release the middle mouse button, the graph is redrawn. With this zoom-option it is possible to select a part of the impulse and truncate it (click button “Truncate” on the bottom pane). Echoes can be eliminated in this way.

To zoom-out, just press the middle-mouse (or wheel) button and draw a rectangular box from right to left. The size does not matter. You can zoom-in/zoom-out multiple times.

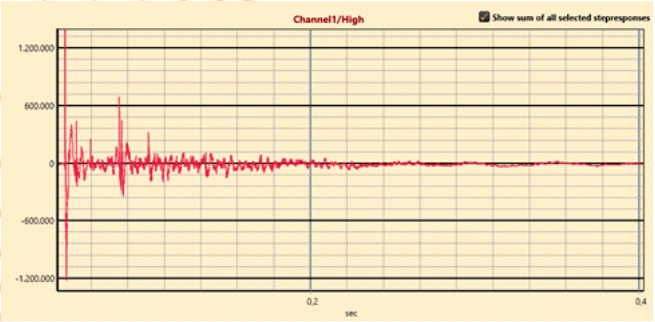


Picture 32: Impulse response graph

After truncating the impulse response, you can set its start value to be used in all channels. That way, your SUM response will always use de right delays which were inserted during the measurement of all drivers. But only when your measurement setup did not change during the actual measurements.

Step response graph

Click the tab “Step” at the bottom pane to select the step-graph. Like the Impulse graph, you can drag the graph by left clicking somewhere in the graph, hold the click and drag. Zooming also works in the same way.



Picture 33: Step response graph

Filter design examples

After flattening your impulse response, you can pass or block the signal for a particular driver. First order filters have a slope of 6dB/Octave, second order filters a slope of 12dB/Octave, and so on. To help build multiple order filters, some examples are provided: Butterworth, Linkwitz-Riley and Bessel.

This paragraph also shows a phase plot of an Allpass1 and an Allpass2 filter.

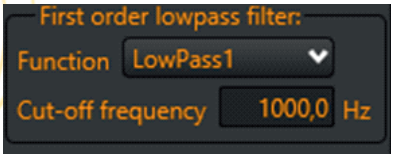
Butterworth

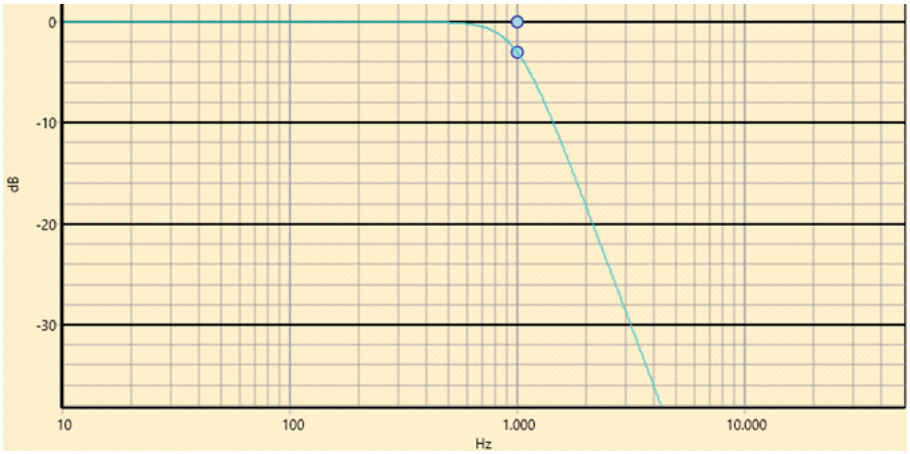
To build a lowpass Butterworth filter, you can use biquad elements in cascade having the same cut-off frequency with specific Q-factors. The table shows the various Q-factors for Butterworth filters from 2nd to 6th order:

Orde	Q 1stbiquad	Q 2ndbiquad	Q 3rdbiqua
2	0.71		
3	1st order filte no Q	1.0	
4	0.54	1.31	
5	1st order filte no Q	0.62	1.62
6	0.52	0.71	1.93

Table 3: Butterworth coefficients

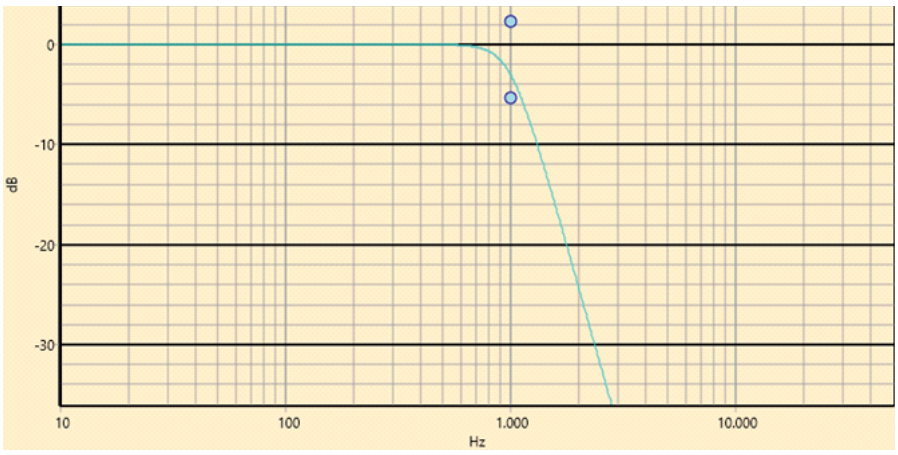
Third order low pass Butterworth example (Fc=1000Hz)





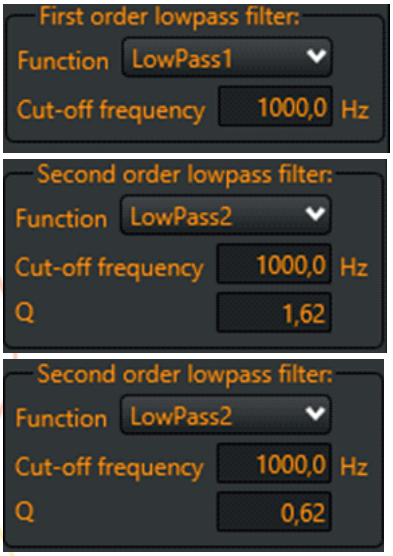
Picture 34: Third order low pass Butterworth example

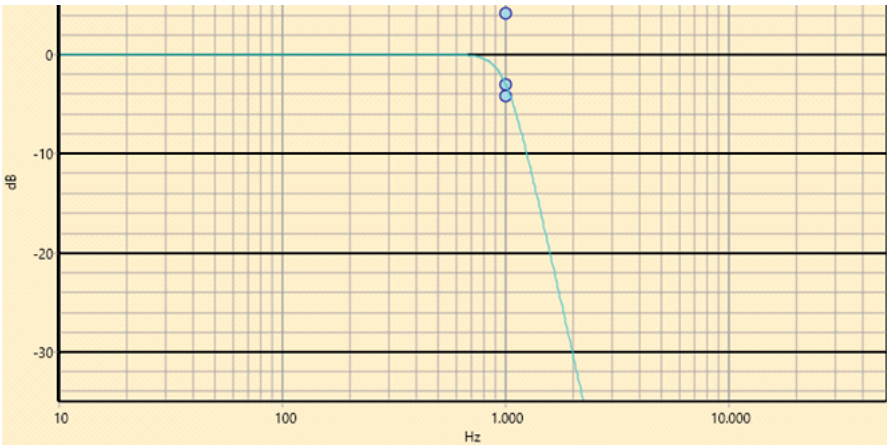
Fourth order low pass Butterworth example (Fc=1000Hz)



Picture 35: Fourh order low pass Butterworth example

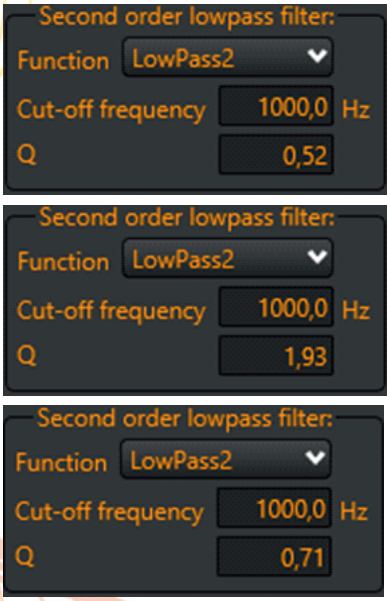
Fifth order low pass Butterworth example (Fc=1000Hz)

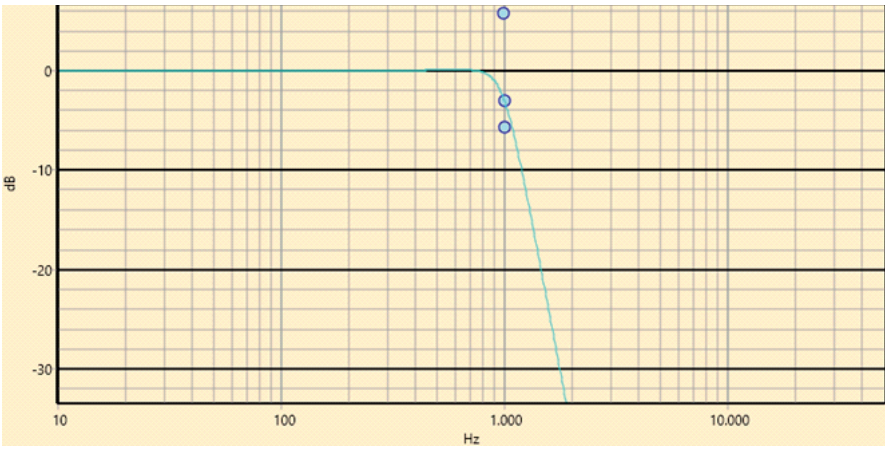




Picture 36: Fifth order low pass Butterworth example

Sixth order low pass Butterworth example (Fc=1000Hz)





Picture 37: Sixth order low pass Butterworth example

Linkwitz-Riley

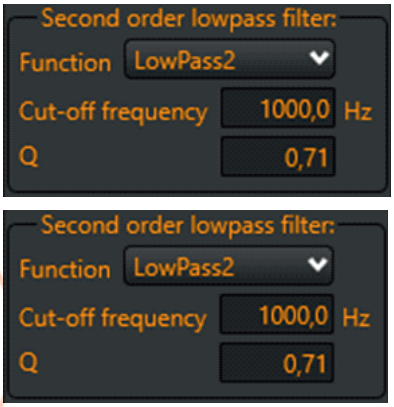
For best crossover filter results, Linkwitz-Riley filters are a good choice. They come only in even-order, so like 2nd, 4thand 8th etc. The 2nd order filter will give you a phase shift of 180 degrees. Best is to use the 4th order Linkwitz-Riley, which will result in 360 degrees phase shift. That way, the low and high output appear in phase (although the low output will have a one-period delay). A 2nd order low-pass or high-pass LR (short for Linkwitz-Riley) filter has a Q of 0.5 on its biquad. A 4th order LR filter (LR4) is a cascade of two 2nd order Butterworth filters (see Butterworth).

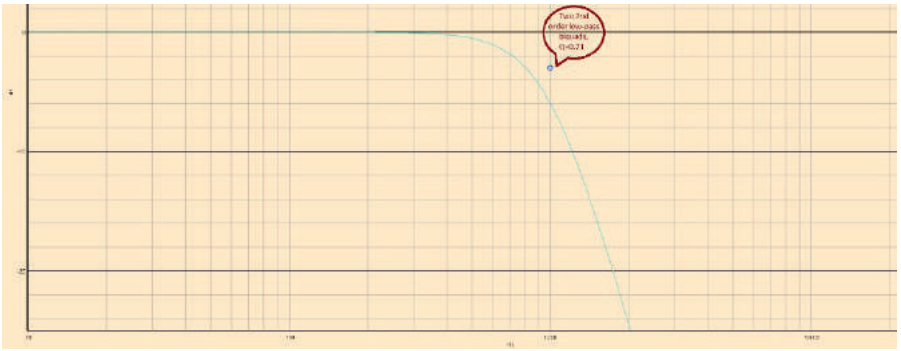
An 8th order LR filter (LR8) is a cascade of two 4th order Butterworth filters, etc. The main advantage of LR filters is, when using them as low-pass and high-pass crossover filters (with the same cut-off frequency), your total filter response will be flat.

Example: Crossover filter, LR4, cut-off frequency 1kHz.

Create with two low-pass biquads on channel 1 and 2 high-pass biquads on channel 2. The two low-pass and two high-pass biquads have a Q-factor of 0.71 as in cascaded Butterworth filters:

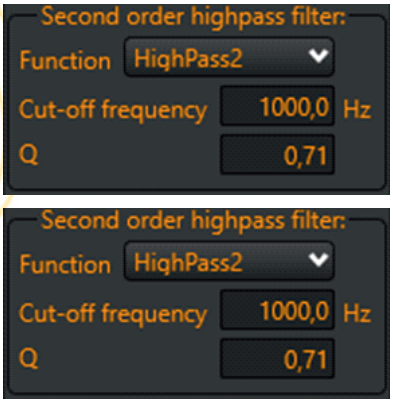
For channel 1 (low frequency driver)

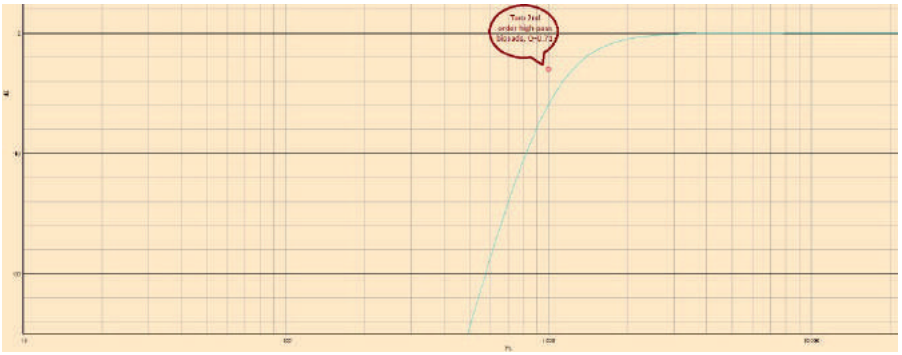




Picture 38: Ch1 biquad settings

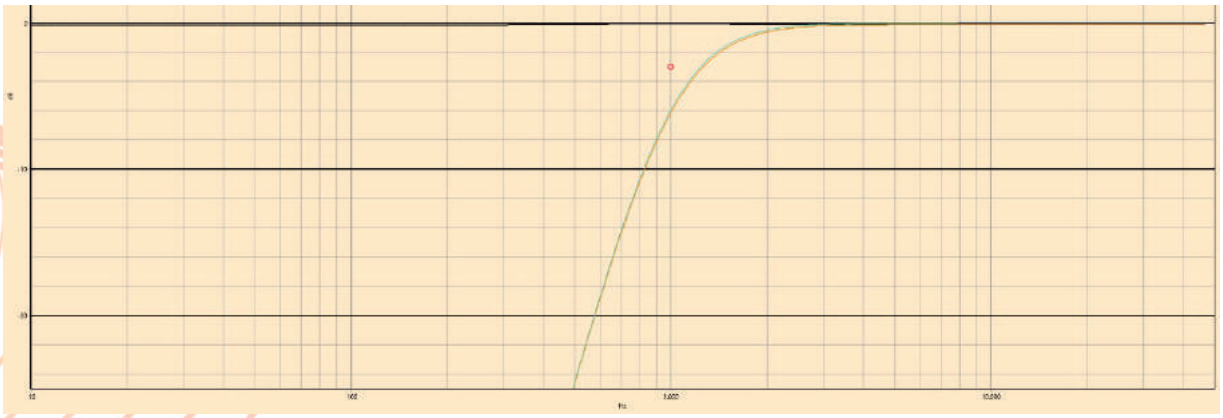
For channel 2 (high frequency driver)





Picture 39: Ch2 biquad settings

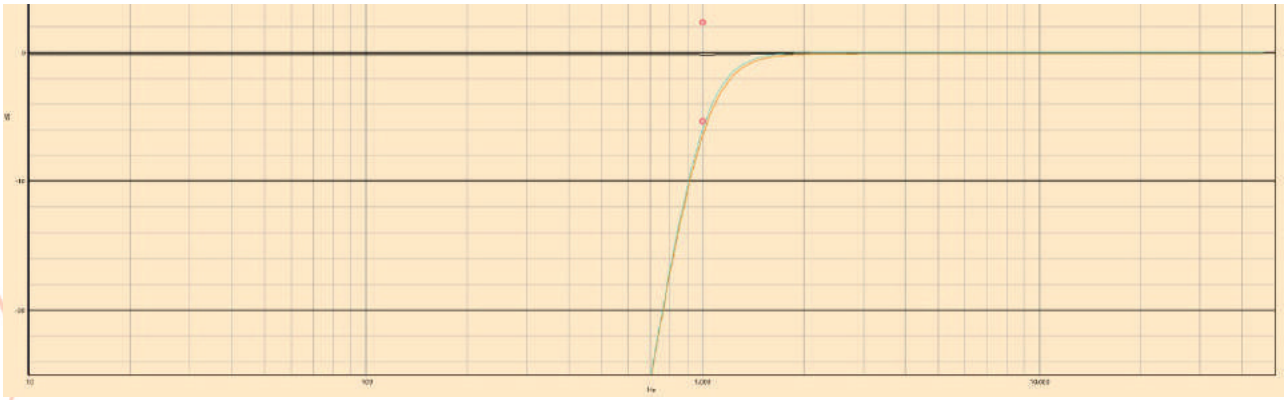
Looking at the sum-result of both channels (you can simulate this in HFD by loading a flat impulse file on both channels), you can observe that the result of the Linkwitz-Riley filter is flat over the total frequency range (black line just below 0dB):



Picture 40: sum result of both channels

LR Order	Q 1st biquad	Q 2nd biquad	Q 3rd biquad	Q 4th biquad	Remarks
2	0.5				
4	**0.71**	**0.71**			**Preferred crossover filter**
8	0.54	1.31	0.54	1.31	

Table 4: Linkwitz-Riley coefficients



Picture 41: Plot of High pass LR8

It's a bit hard to see, but just below the 0dB you can observe the SUM of channel one and two, which is very constant over the whole frequency range.

Bessel filter

Bessel filters have a more linear phase response than Butterworth filters, but they have a worse frequency response than Butterworth filters. Bessel filters are build similar to Butterworth, except for the cut-off frequencies of the used biquad objects. Table 3 shows the various cut-off frequency multiply factors and Q-factors for Bessel filters from 2nd to 6th order:

Order	Q first biquad	First frequency factor	Q second biquad	Second frequency factor	Q third biquad	Third frequency factor
2	0.58	1.27x				
3	1st order filter, no Q	1.32x	0.69	1.45x		
4	0.81	1.60x	0.52	1.43x		
5	1st order filter, no Q	1.50x	0.92	1.76x	0.56	1.56x
6	1.02	1.90x	0.61	1.69x	0.51	1.60x

Table 5: Bessel filter coefficients

Filter object cut-off frequency = Fc x frequency factor

So, for example, if you need to create a fourth order lowpass Bessel filter with a cut-off frequency (Fc) of 1000Hz, you need two biquads:

Biquad	Cut-off frequency	Q-factor
1	1.6x1000=1600Hz	0.81
2	1.43x1000=1430Hz	0.52

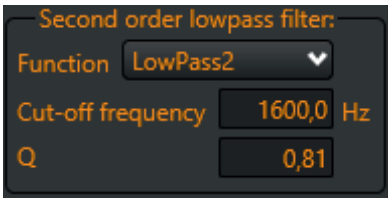
Table 6: Example 4th order Bessel filter Fc=1000Hz

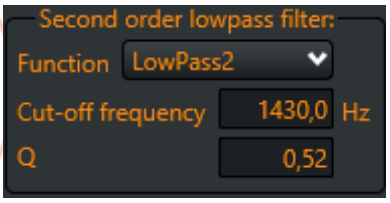
In case of a 6th order lowpass Bessel filter with a cut-off frequency of 1000Hz, you need three biquads:

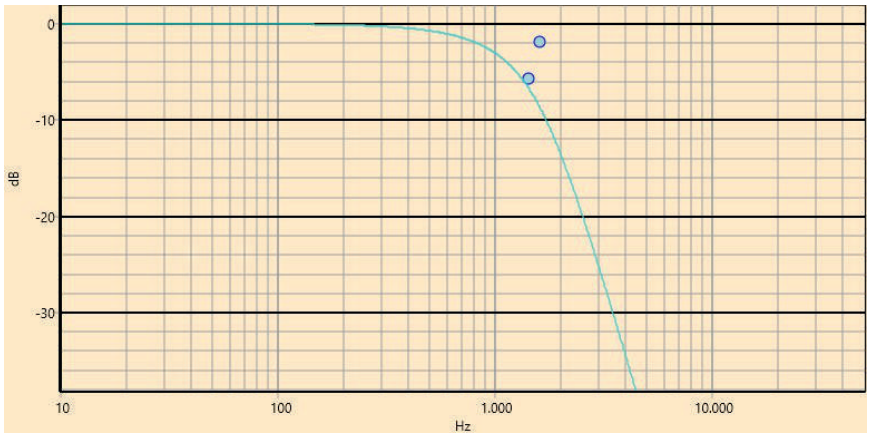
Biquad	Cut-off frequency	Q-factor
1	1.9x1000=1900Hz	1.02
2	1.69x1000=1690Hz	0.61
3	1.6x1000=1600Hz	0.51

Table 7: Example 6th order Bessel filter Fc=1000Hz

Fourth order low pass Bessel example (Fc=1000Hz)

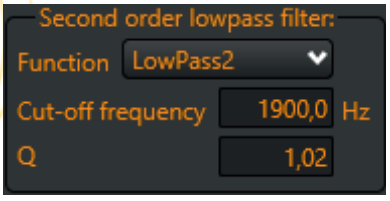


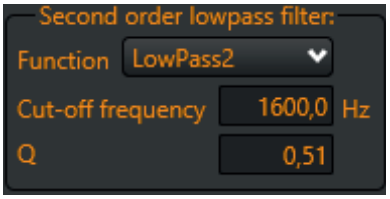


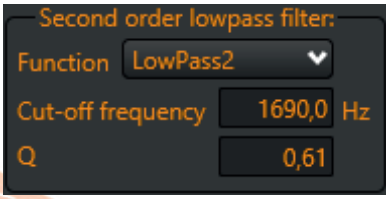


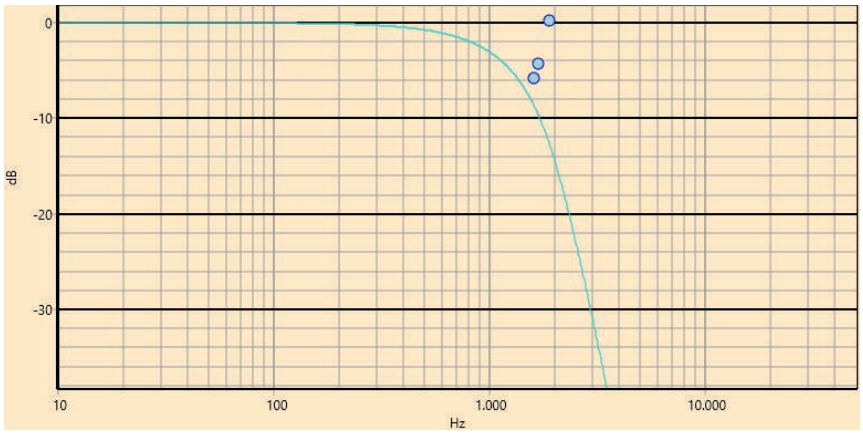
Picture 42: Fourth order low pass Bessel example

Sixth order low pass Bessel example (Fc=1000Hz)







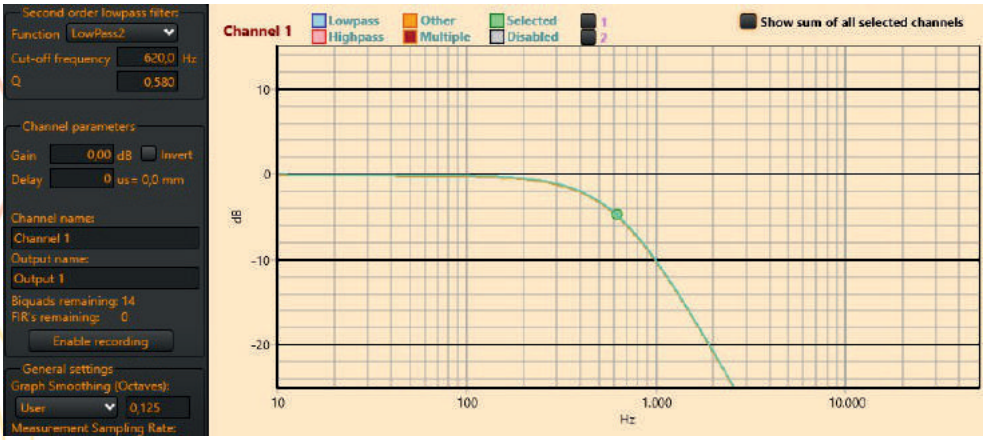


Picture 43: Sixth order low pass Bessel example

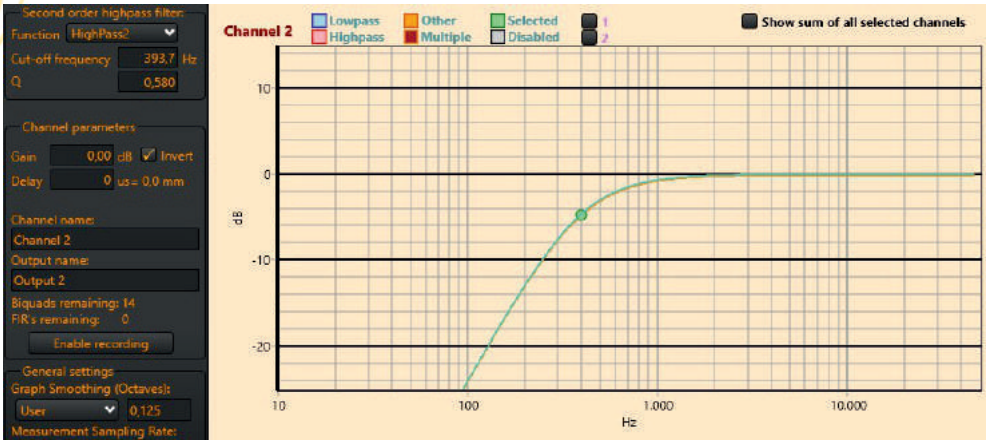
Crossover using Bessel

When creating a crossover filter, use 1/freqFact for the high-pass filter elements. For example, if you implemented a second order low-pass filter with a frequency factor 1.27, use a high-pass filter with a frequency factor if 1/1.27=0.787 This way, the biquads will match their responses at -3dB if you invert one channel.

For example, a 500Hz second order crossover filter, built with one low-pass filter on channel 1 and one high-pass filter on channel 2 and having this channel inverted.



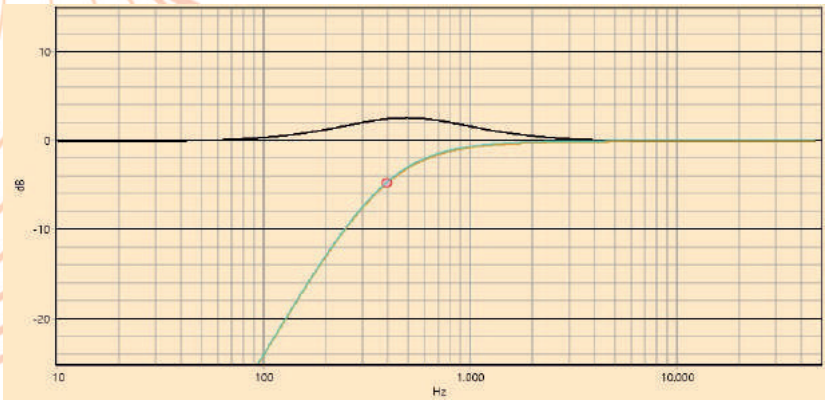
Picture 44: Ch1 low pass filter



Picture 45: Ch2 high pass filter

Notice the inverted channel 2

Will give the following total response:

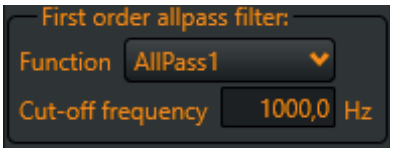


Picture 46: total response

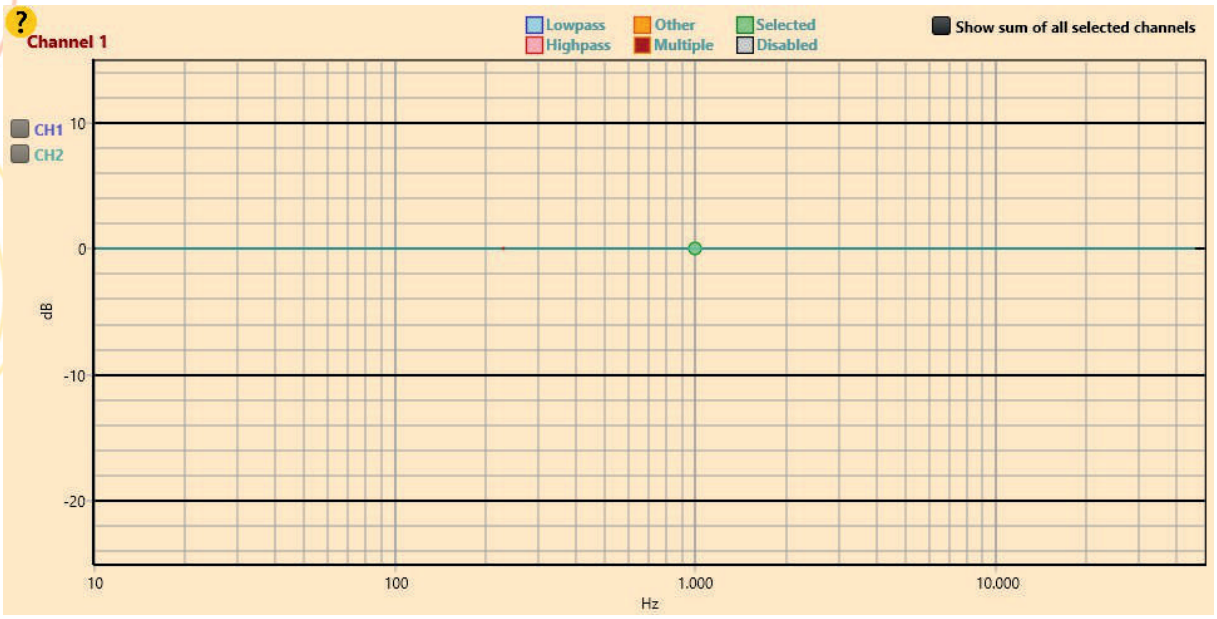
At the designed crossover frequency (500Hz) is a slight boost visible.

All pass 1

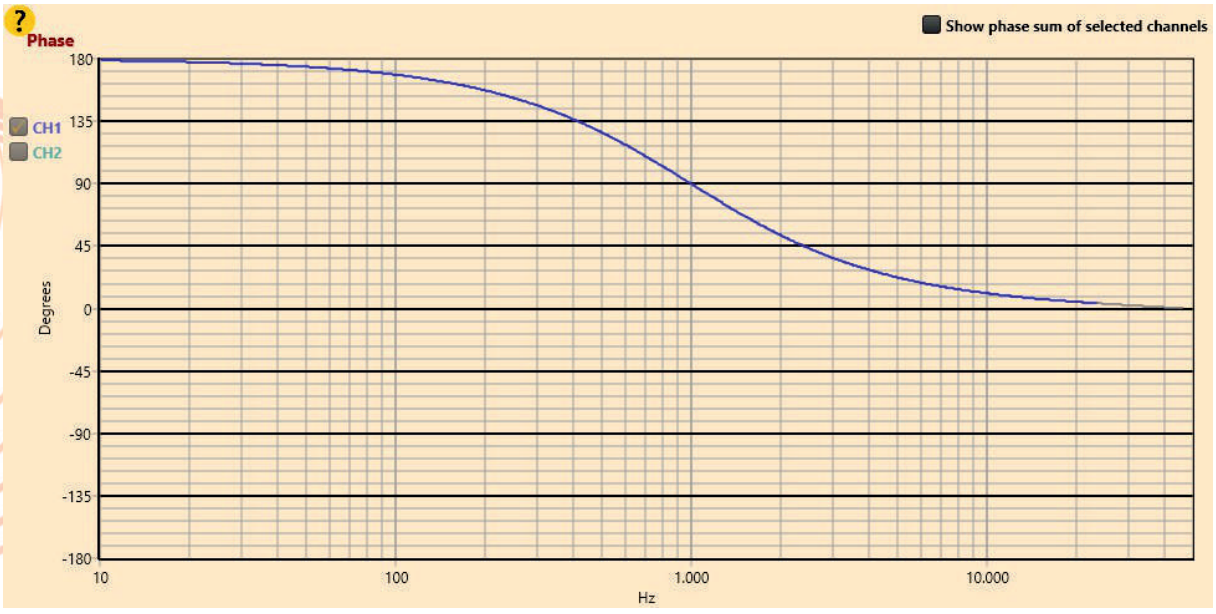
All pass filters are implemented as 1st and 2nd order. An All pass 1 filter will have a gain of 0dB, but will drop the phase over increasing frequency from 180 to 0 degrees.



Picture 47: All pass 1 filter



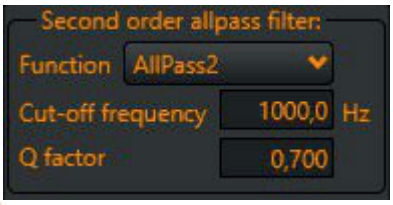
Picture 48: gain of 0dB over the whole frequency scale



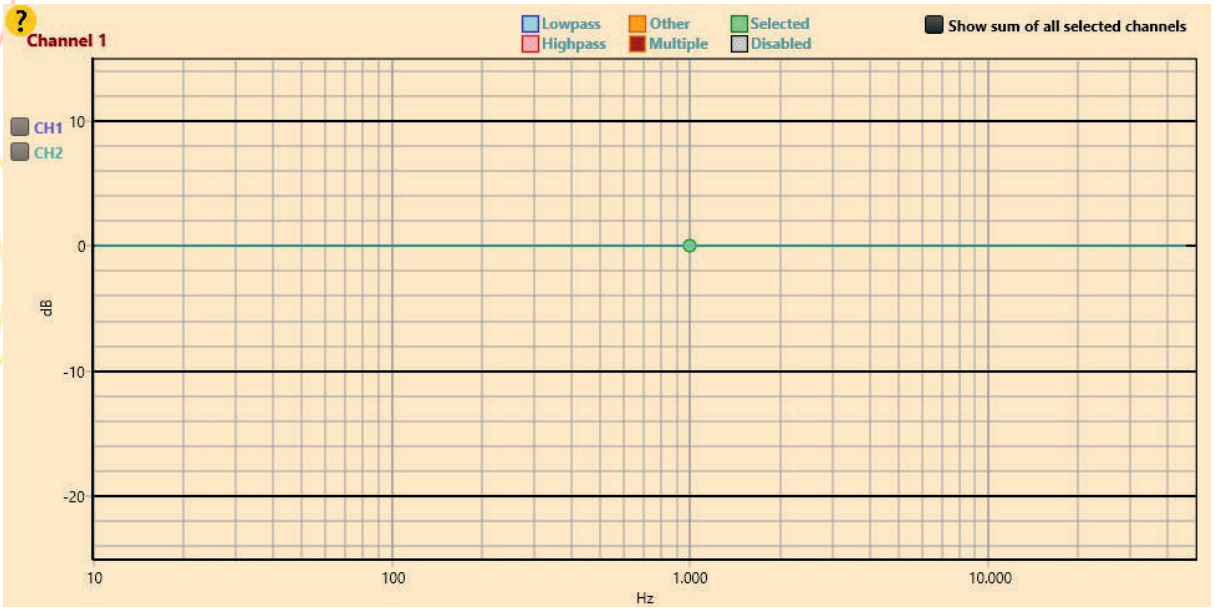
Picture 49: At 1kHz this filter will have a phase of 90 degrees

All pass 2

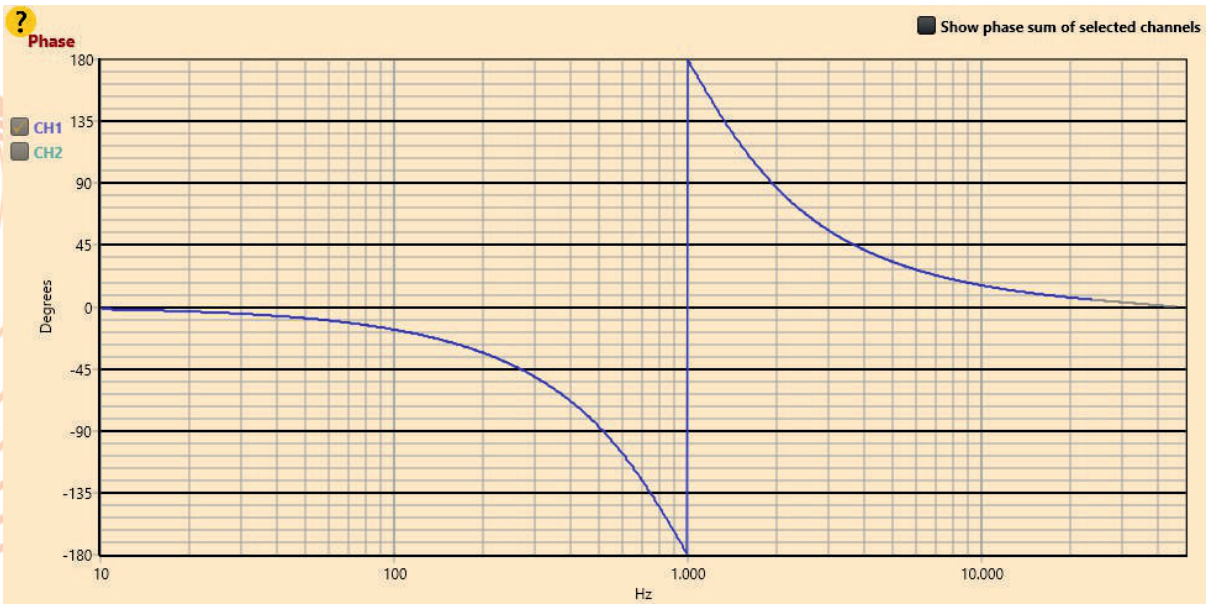
As with the 1st order All pass filter, the All pass 2 also will add no gain, but does add a phase shift of 360degrees in total:



Picture 50: All pass 2 filter



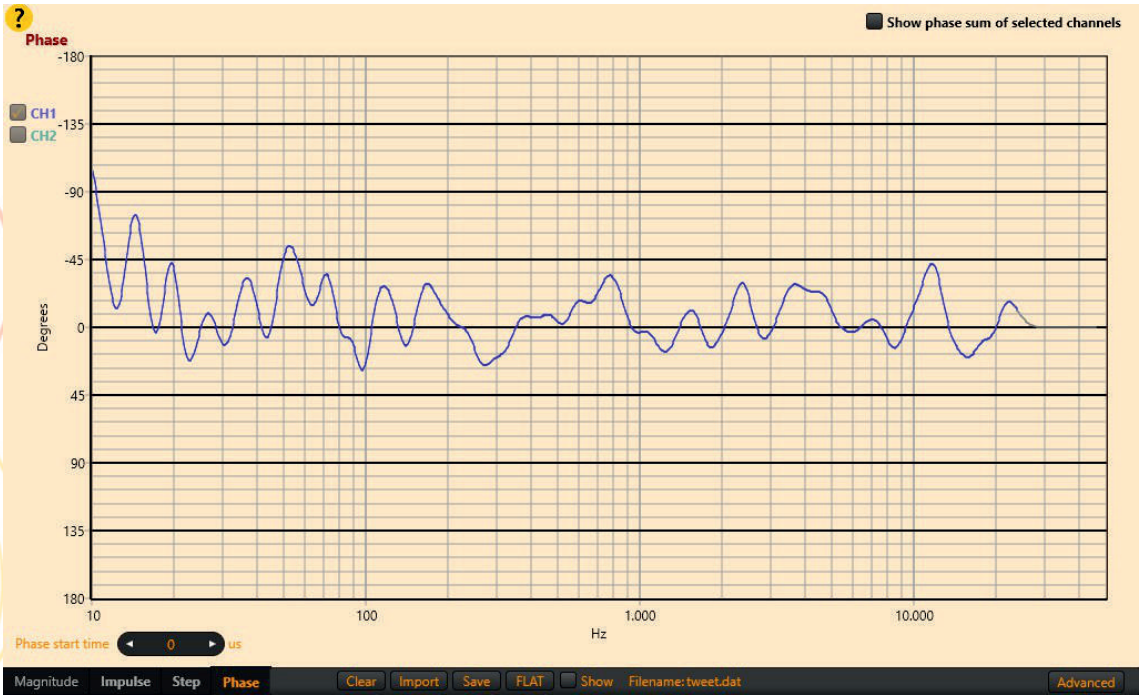
Picture 51: gain of 0dB over the whole frequency scale



Picture 52: phase response starts at 0degrees and will drop to -180degrees at its centre frequency

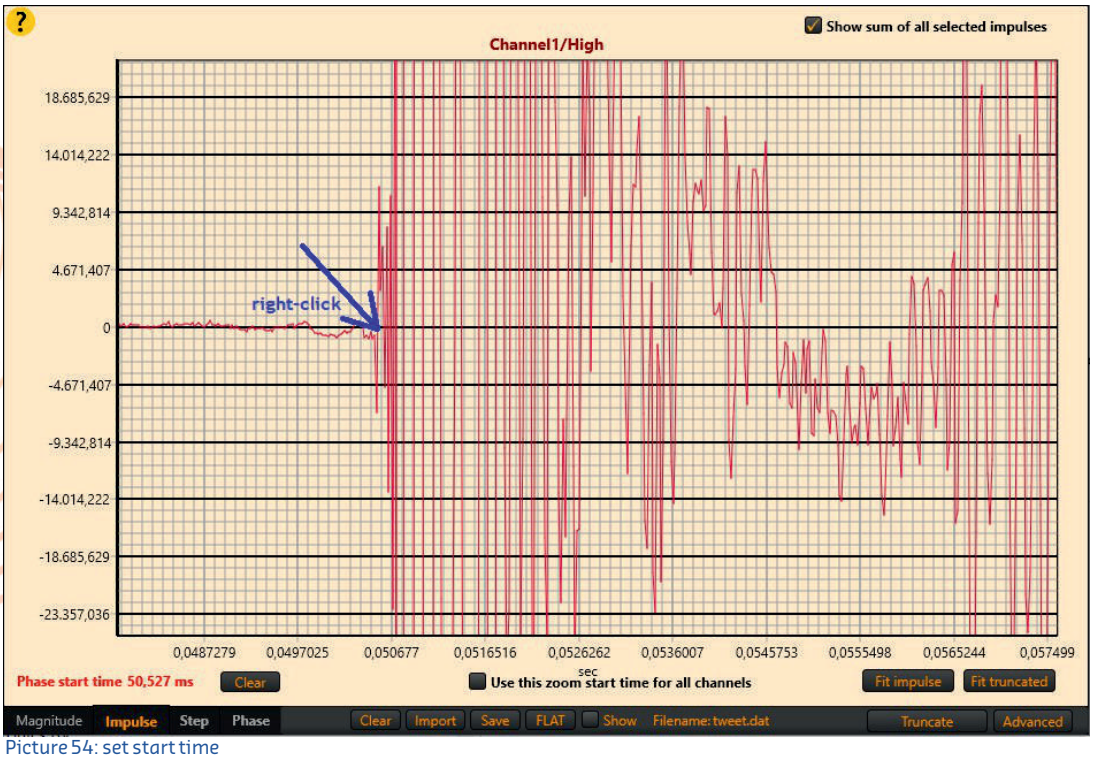
Phase plot

HFD has the option to view the phase of the channel response. It can be observed by selecting the 'Phase' tab:



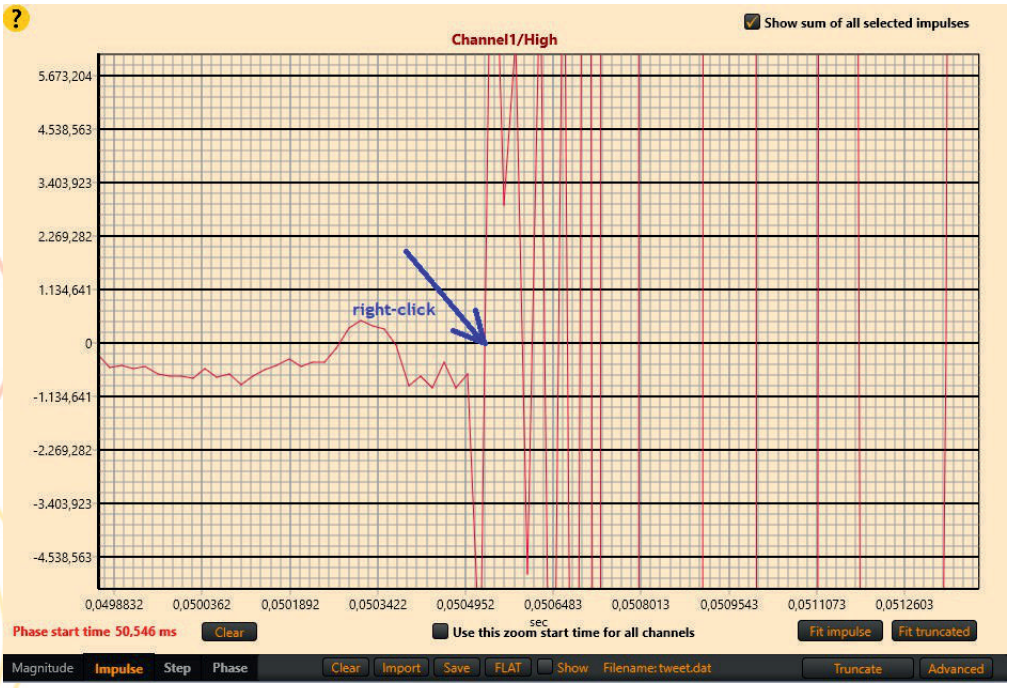
Picture 53: phase plot

Due to delays in the recording, the phase will be wrapped around a lot. To skip recording delay, it is possible to adjust the start time. This can be done by manually clicking on the '<' or '>' arrow of the start-time edit box, or hovering with your mouse over it and rotate the mouse wheel. Every click will stand for 1 sample. It is also possible to enter a value manually. Check the impulse tab to get an approximation of where the calculations should start. If you **right-click** on the graph at the spot pointed out by the blue arrow in Picture 48, it will set the start time:



Picture 54: set start time

Or zoom in further, to get a more accurate spot:



Picture 55: zoom in on phase plot

After entering 55000us as the start time in this example, or right clicking near the blue arrow, you can adjust the start time. On the Phase plot tab, you can use the mouse wheel (while hovering the mouse cursor over the start time edit box), to adjust the time a few samples forward or backwards. After tweaking the start time manually to 50759us, the phase plot could look like this:

Picture 56: phase plot after start time adjustment

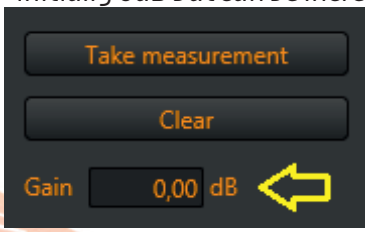
NOTE: If smoothing is set, the phase-plot will look more gracious, but may not represent the real response. So in doubt, set channel smoothing to None or select a low level smoothing.

Recording

When your computer has recording capabilities, the recording button will be enabled in the left pane. To take measurements, click on “Enable recording” button. It will open new features in the left pane:

Picture 57: Enable recording button

A “Measure info” box will be visible at this time. It shows which audio driver is selected and which inputs and outputs. These settings can be changed in the “Tools-Asio preferences” menu-option. Before taking a measurement, HFD calculates a sweep to output with a certain gain. This gain is initially 0dB but can be increased or decreased at your discretion. Change the output gain here:



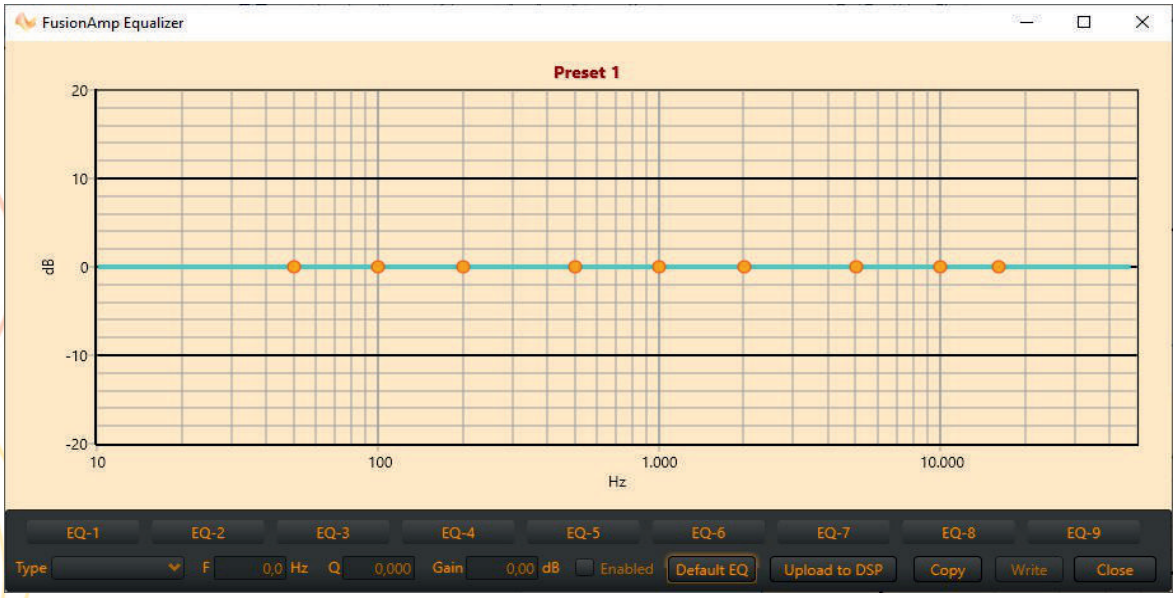
Picture 58: Change sweep output gain

Hit the “Take measurement” button to start a measurement. HFD will output a sweep signal and record incoming audio. Afterwards, the impulse and step response are shown in the graphs. From here you can zoom in on the impulse response and truncate to remove echo's and start editing a proper filter for the selected channel.

To remove a recording, click “Clear”. To repeat a recording (and deleting the last one), just hit the “Take measurement” button again.

Equalizer

A parametric equalizer is added to the FusionAmp as of firmware version v1.4 The equalizer is placed right after the input selection, before the signal is provided to the three channels.



Picture 59: Equalizer screen, no band selected

By default, all biquads are of type “BoostCut” with a gain of 0dB, making the equalizer to pass audio without any alterations. Up to 9 biquads are available to prepare the signal, before it is fed to the output router. Therefore, all three channels use the same equalizer. The equalizer settings are stored in the preset settings, making that every preset has its own equalizer.

There are several ways to enable a biquad in the EQ. For example, push button “EQ-4”. This will select the default biquad with a center frequency of 500Hz. This can be changed in the frequency edit control. You can also drag the biquad over the screen, adjusting the frequency and gain at the same time. Rotating at that moment the mouse-wheel, it will change the Q-factor.

Picture 60: Equalizer screen, preset 1, band 4 selected: BoostCut

After selecting the biquad of band 4, you may change its function. For example, we want to make it a low shelf filter. Choose “ShelfLo” at the bottom-left.

Picture 61: Equalizer screen, band 4 selected and filter type changed to ShelfLo

Now you may change gain and Q properties by hand, or click the biquad in the graph (the little green circle, indicating a selected biquad) and drag it around in the graph. When a biquad is selected, you can change its Q by rotating the mouse wheel. This is also possible at the input controls for frequency, Q and gain: hoover your mouse cursor over the input and rotate the mouse wheel to change the value in the box under the cursor.

Picture 62: Equalizer screen, band 4 selected and filter type changed to ShelfLo

Up to 9 biquads can be used to create an equalizer which suits your demands. In the next picture, the biquad of band 9 is selected and given a new frequency of 12kHz. This is done by changing the frequency in the corresponding edit box. You can change the value by typing 12000 into the box, or hover with the mouse cursor over the edit control and rotate the mouse wheel. The frequency will change with 1000Hz at this level (It will change with 100Hz if it is between 1000Hz and 9999Hz, and with 10Hz if between 10 and 1000Hz). Or you can drag the biquad to 12kHz.

Picture 63: Equalizer screen, band 9 selected and frequency changed to 12kHz

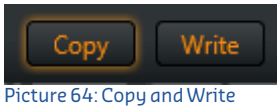
Updates to the FusionAmp are not done automatically. Therefore, after changes are made, these must be saved to the FusionAmp by uploading them, using the "Upload to DSP" button. Once uploaded, the equalizer will have immediate effect.

As you can see in the picture above, one biquad is coloured orange (not selected) and one is green (selected). You can also select the left biquad (of band 4) by clicking on it with your mouse.

Equalizer settings are stored per preset. So, when selecting another preset while not saving the changes you made, the settings of the newly selected preset will replace the ones on the screen. Therefore its best practise to only change the preset when you are done editing and uploading the equalizer.

TIP: The equalizer screen opens besides the main screen. Therefore, it is possible to use both the equalizer screen and the main screen at the same time, making it possible to upload equalizer settings and changing the volume to listen if the settings have the expected effect. Or to select another preset and edit the equalizer settings for that preset too.

Copy and Write



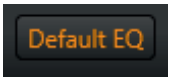
Picture 64: Copy and Write

After the equalizer is uploaded to the DSP, it is possible to make a copy of all three equalizers (one per preset) of the connected FusionAmp. Hit the "Copy" button to copy the EQ parameters from the FusionAmp. HFD will remember them for you, even if the FusionAmp is disconnected. The parameters are NOT remembered after closing HFD.

When another FusionAmp is connected and you have stored the EQ parameters, it is possible to recall these parameters and writing them to the connected FusionAmp. Hitting "Write" will send the stored EQ parameters to the connected FusionAmp. All three EQ presets will be overwritten.

This feature comes in handy, when you want to copy the Equalizer settings from one FusionAmp to another (For example, master to slave).

Default EQ



Picture 65: Default EQ

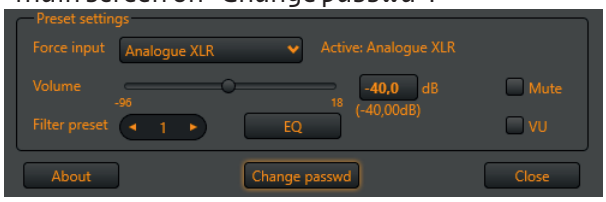
When you want to start all over with the current selected equalizer (that is, the eq for the current preset), you can set all nine biquads to their default values by hitting the "Default EQ" button. It will set all biquads to type "BoostCut", with a Q of 0.707, a gain of 0dB and a frequency of 50Hz(band 1), 100Hz(band 2), 200Hz(band 3), 500Hz(band 4), 1kHz(band 5), 2kHz(band 6), 5kHz(band 7), 10kHz(band 8) and 16kHz(band 9). This button will not affect the other presets! After loading the default values, the equalizer must be uploaded to the DSP (hit "Upload to DSP" button).

If you upgraded the firmware from v1.3x to a higher version, like v1.4x or v1.5x, it is mandatory that you upload a default EQ first, as firmware versions previous to v1.4 did not have the equalizer option.

Change password

To prevent customers being able to change the settings above, the FusionAmp should be protected with a password.

Close the device settings screen and click in the main screen on "Change passwd".



Picture 66: Change password

This will open a new password screen:

Picture 67: Password login dialog

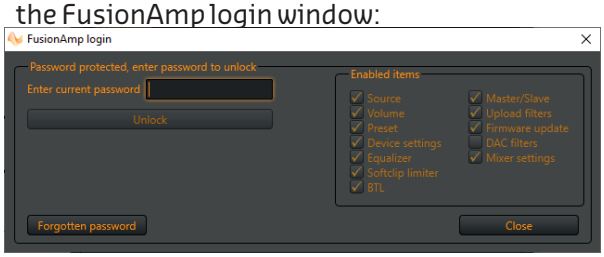
On the left side, the standard password protection entry fields are visible, on the right (grouped as "Enabled items") you will find the options than can be protected with a password. When an item is checked, it is NOT protected and the user can change this option on a connected FusionAmp without entering a password.

For example, if you want a user being unable to change any device settings, uncheck "Device settings" and protect the FusionAmp with a password. Besides a password, a valid e-mail address is necessary. This e-mail address will be programmed into the FusionAmp also, in case of a forgotten password.

Picture 68: Disable Device settings

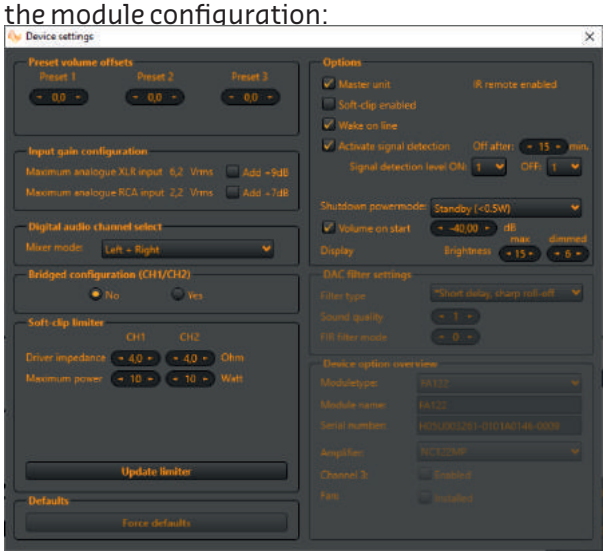
Every time a password protected FusionAmp is connected to HFD, it should be unlocked first, to be able to change the "Enabled items". Once unlocked, clicking "Enabled items" will have immediate effect (no save is necessary). Needless to say that an unlocked FusionAmp has all its protected items accessible. So please keep your password a secret!

When a password protected FusionAmp is connected to HFD, clicking "Unlock" will open the FusionAmp login window:



Picture 69: Enter password

n this example, only the DAC filter settings are protected. The user is able to change all device settings, except the DAC settings, and of course, the module configuration:



Picture 70: Disabled DAC settings

The items that are not accessible are greyed out ("DAC filter settings" and "Device option overview").

Document Revision	Description	Date
R0.0	Initial Draft	15-06-2020
R0.1	Corrected	23-06-2020
R1.0	First release	15-12-2020
R1.1	Minor screenshot and textual changes	16-09-2022
R2.0	HFD v5.0 implementation	15-05-2023