

The left side of the page features decorative wavy lines in shades of orange and yellow, creating a modern, flowing design element.

# Hypex Filter Design manual

# Introduction

Hypex Filter Design is a Windows Software program to control Hypex Electronics B.V. DSP powered products. For example, our FusionAmp plate amplifiers and our DSP OEM modules. 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 Design. 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 (DSP/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:


- DSP3-213/DSP3-224/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.

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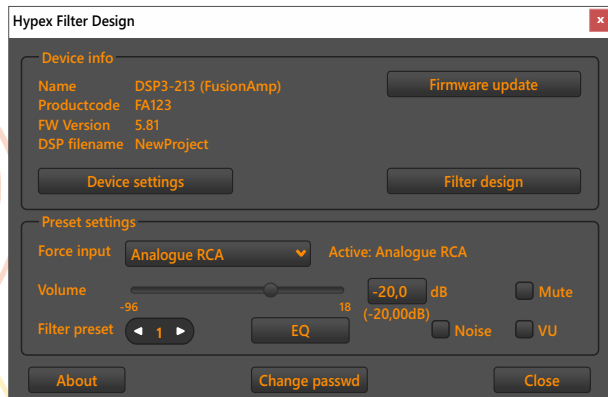


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## 1. Main screen

When a DSP 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 filename 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'.

## 2. 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 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.

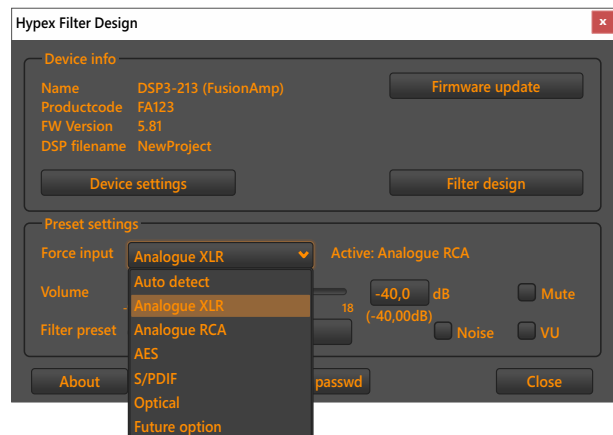
### Auto detect

When "Auto detect" is selected as a source, the DSP 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 DSP 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.

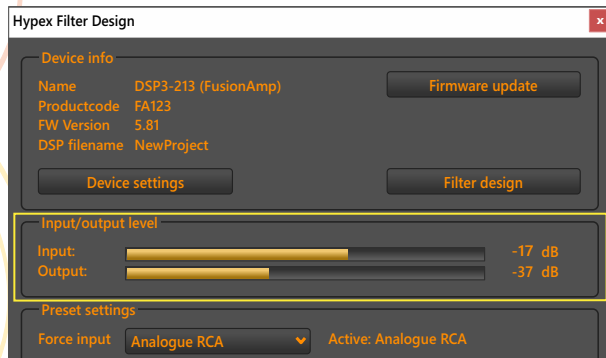


Picture 3: Device is muted

### VU

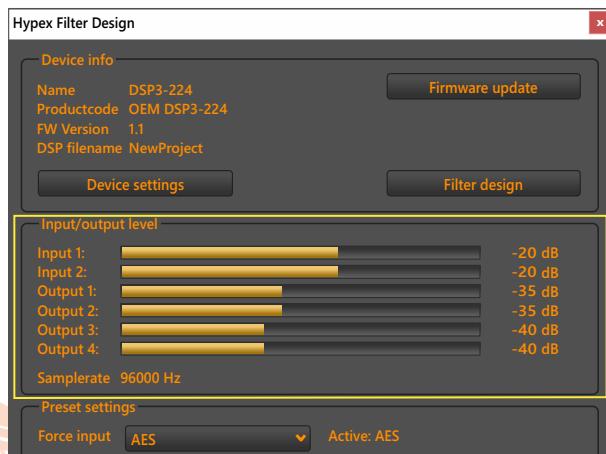
When a DSP is connected to HFD, it is possible to observe the input and output level of the device. This will give you an indication

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 for DSP3-213 devices:



Picture 4: Main screen with VU indication (DSP3-213)

And it will look like this for DSP3-224 devices:



Picture 5: Main screen with VU indication (DSP3-224)

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

### Noise

When checked, the connected device will produce a noise signal with level -55dB rotating on the three DSP3-213 outputs (only outputs that are configured with an amplifier), or on the four DSP3-224 outputs (also only on outputs that have an amplifier configured). Unchecking this box will resume normal audio output.

### 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 6: Selecting a preset

### Equalizer

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

### Change password

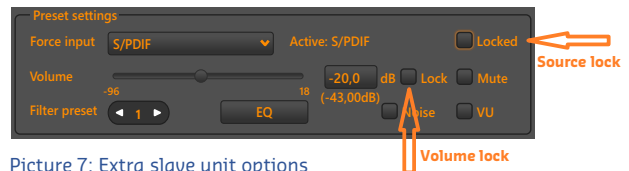
To prevent specific functions of the software from being accessed by unauthorized persons you can protect them by setting a password. Please refer to chapter 'Change password' for more information.

### About

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

### Extra slave unit options

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



Picture 7: Extra slave unit options

The lock functions are especially added for slave units that are connected to a master unit by means of an S/PDIF cable. The slave unit receives audio from the master unit 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 unit 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 unit is not able to change the volume of the slave unit. This is useful when the source of the slave unit 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 unit can lock its source, so it will not change when the source of the master unit is changed. For example, when the slave unit is connected to the master unit with an SPDIF cable, the master unit source can be switched to Optical, SPDIF or AES, whereas the slave unit source should always remain SPDIF.

An exception is when SPDIF is locked and an analogue source is selected on the master unit. In that case the slave unit will follow the master unit.

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

Slave unit Locked source	Master unit Forced selected source	Slave unit 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 unit source lock

## 3. Device settings DSP3-213

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 8: 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 devices, 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 DSP will output only the left or right signal.



### Bridged configuration (CH1/CH2)

On the 2- and 3- way devices 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 device 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 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 DSP and check **Soft-clip enabled** in the 'Options' section 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 DSP3-213 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 DSP is gained with a Release setting of 99.9msec.

### Options

#### **Master unit**

The DSP 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 DSP as a master unit and the other as a slave unit.

The master DSP 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 unit, uncheck it when this device is connected to a master unit.

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

#### **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 limiter'.

#### **Wake on line**

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

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

#### **Activate signal detection**

The DSP has two power-down modes: "Standby" and "Low power" (See 'Shut down power-mode'). When the DSP is in any power-down mode, it can wake-up by an analogue signal on the XLR, RCA or SUB input. If the DSP 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 DSP3-213 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 device to switch ON when the audio-level is about -60dB and will trigger to switch off when the audio level has dropped below -70dB.

If your device tends to switch on un-expectedly 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 device 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 device 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 device or SUBin3-110 only)



Picture 9: SUBIN volume setting

By default, the volume control knob on the SUBin3-110 (1-way device), 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 amplifier may produce maximum power and damage your

drivers and/or ears. Note: When enabled, the master volume setting is ignored.

#### Shutdown powermode:

As already mentioned in 'Activate signal detection' the DSP 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 device) 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 DSP (2- and 3-way devices 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 device will consume less than 1.5W.

#### ✓ Volume on start

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

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

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

#### Standby led brightness

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

#### Display brightness

If you have connected the "OLED Display" option, a different setting will appear:



Picture 10: Display brightness setting

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).

Note: OLED displays are very sensitive to burn-in symptoms. To avoid burn-in symptoms, **we recommend that you set the 'dimmed' level to 6 or lower.**

### 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 DSP. It is wise to do so when volume is set to a low value. Beware that the soft-clip limiter will be disabled!

## 4. Device settings DSP3-224

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.



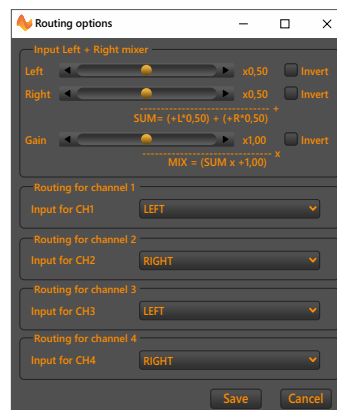
Picture 11: Device settings screen

It is slightly different than the DSP3-213 settings screen. This is because the DSP3-224 has stereo inputs and 4 output channels (2x2).

### Preset volume offsets and input routing

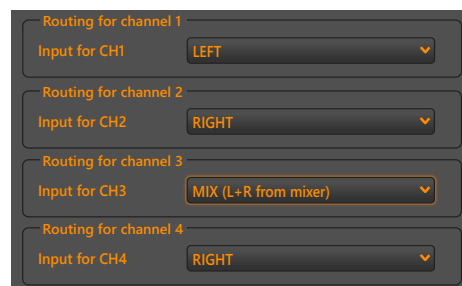
The preset volume is an offset relative to the master volume. This volume can be set for each preset individually. This volume ranges from -24dB to +24dB.

Every output can be fed with audio from the LEFT input, RIGHT input or a combination of LEFT+RIGHT. This is done through the **Routing** button and is valid per preset. Hitting this button will open a new window:



Picture 12: Routing options

By default, LEFT input will be routed to outputs 1 and 3 and the RIGHT input will be routed to outputs 2 and 4. This can be changed to suit your needs. It is also possible to mix LEFT and RIGHT and add some gain to the result and route the (mono) result to an output. For example, when output 1 is driving your LEFT speaker, output 2 your RIGHT speaker and a subwoofer is connected to output 3, the configuration would look like this:



Picture 13: Channel 3 mixed

### Bridged configuration

As dual output amplifiers only can be connected through Hypex HBOX connectors, outputs 1 and 2 can be joined in a bridge setup and outputs 3 and 4 can be joined in a bridge setup. When BTL is activated for outputs 1 and 2, only channel 1 is needed for filtering, as output 2 will inverse the output of channel 1. When BTL is activated for channels 3 and 4, only filtering is needed on channel 3, as output 4 will be the inverse of channel 3 output.

### Soft-clip limiter

The implementation of the limiter slightly differs from the one in the DSP3-213. Each output has its own limiter, which can be enabled individually. Also, each limiter has its own attack and decay parameters, but by default, they all use the same settings as the limiter for channel 1 (this is set by checkbox ☒ All channels use the Attack and Decay values of CH1).

Enter for each driver its impedance and the maximum power you want to output into that driver. When finished, hit [Update limiter](#) and the settings will be programmed into the DSP3-224.

### Options

The DSP3-224 has the same option settings as the DSP3-213, please take a look at the "Options" paragraph at page 9.

The DSP3-224 can be a master for a DSP3-213 in slave mode, connected through the S/PDIF on the digital board.

### Device option overview

As mentioned before, the DSP3-224 has 4 outputs. Each output can be assigned to a single amplifier, but mostly, as the DSP3-224 is a stereo dsp, outputs will be connected to a dual amplifier module, like a NC122MP, NC252MP, NC502MP, NCAS500, NCAS1000 or NCx102EXT (which as no power supply of its own). This implies that when you configure channel 1 to be connected to a NC252MP, channel 2 will also be configured for the same amplifier.

Valid configurations are for example:

Amplifier ch1:	NC252MP
Amplifier ch2:	NC252MP
Amplifier ch3:	NC100HF
Amplifier ch4:	NO Amp

Or

Amplifier ch1:	NC252MP
Amplifier ch2:	NC252MP
Amplifier ch3:	NCx102EXT
Amplifier ch4:	NCx102EXT

Or

Amplifier ch1:	NC500MP
Amplifier ch2:	NC100HF
Amplifier ch3:	NC500MP
Amplifier ch4:	NC100HF

Etc.

When an output is configured as "NO Amp", it only means that the DSP3-224 will not check the temperature of that channel. Also, there will be no gain adjustment, making 0dB appear as a 3.1Vrms signal.

## 5. 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).

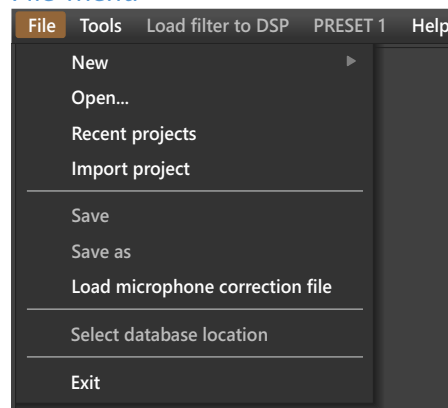
When creating a new project, HFD always opens the graphical interface first. Use the [Advanced](#) button in the bottom-right corner, to switch to the advanced screen. You can switch back to the graphical screen with the [Back to filter design](#) button at the same spot.

### 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

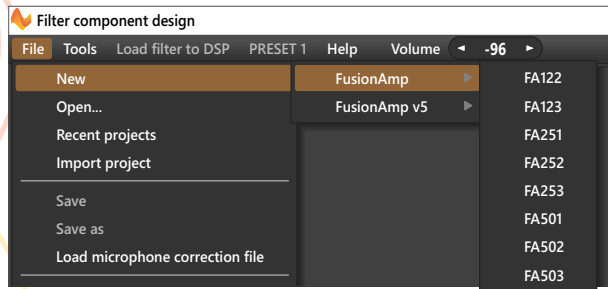


Picture 14: File menu

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 menu item “Import project”.

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.



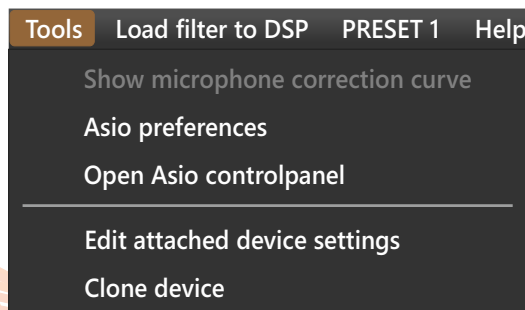
Picture 15: New project

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 16: 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 <https://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.

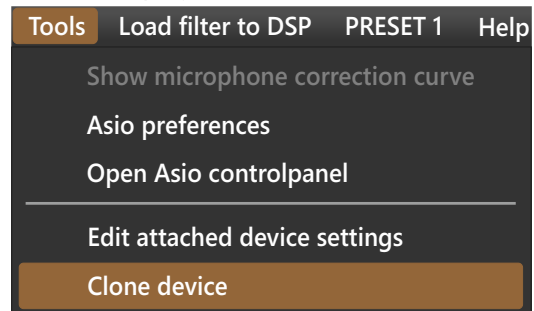
Asio preferences will open a new dialog in which you can select the audio device, input and output channel of that device to configure your hardware and be able to take measurements in the graphical filter design screen.



Picture 17: Set Asio preferences

#### Clone device

In a stereo setup both the left and right devices will almost always have the same settings and filters. Or, in case of mass production, one filter design can be transferred to many different devices. The “Clone device” option can make programming multiple devices a lot easier as different plates (although they must be of the same module type!) can be programmed with the same project file.



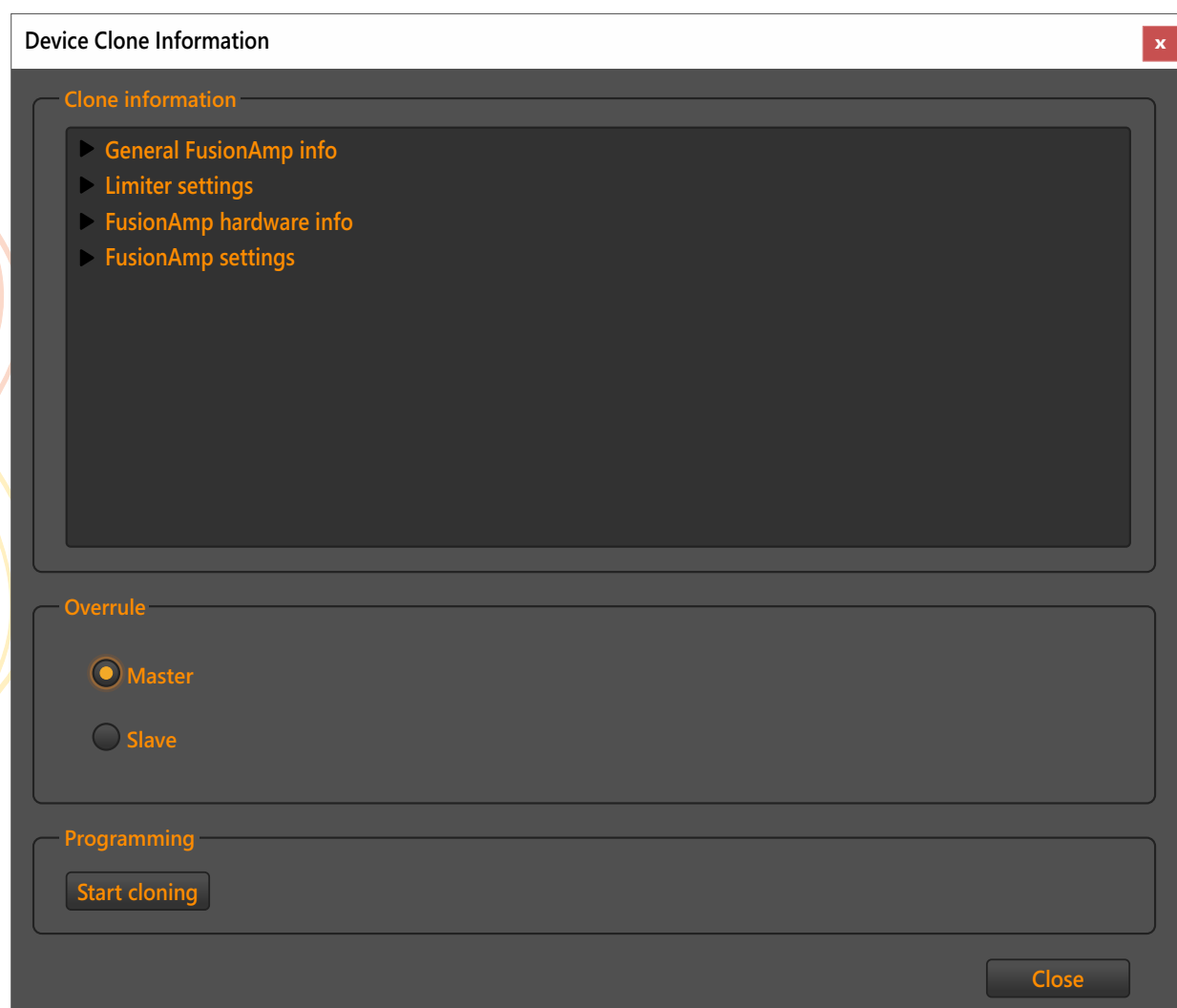
Picture 18: Clone device

This is accomplished by following these steps:

1. Connect a DSP 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 DSP.
7. Save the project (this will store all settings, including eq and filters).
8. Disconnect the current DSP and connect another of the same type.
9. Under “Tools”, the option “Clone device” should again be available.



Hit “Clone device” to open the cloning screen:



Picture 19: 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 unit. The extra slave unit options are the same as you will find on the main screen when a slave unit is connected:

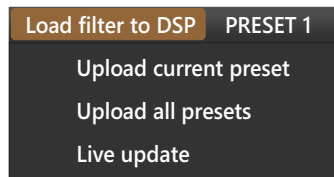


Picture 20: Clone Slave unit options

This way, you can easily clone a master unit into a slave unit, but it is of course also possible to make a clone project of a slave unit. 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.

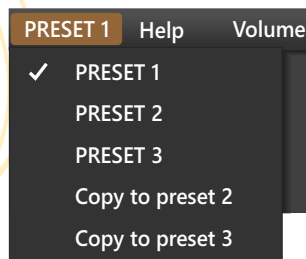
## Load filter to DSP menu



Picture 21: 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 22: 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

Hit **Advanced** button in the right bottom corner.



Picture 23: Advanced screen overview

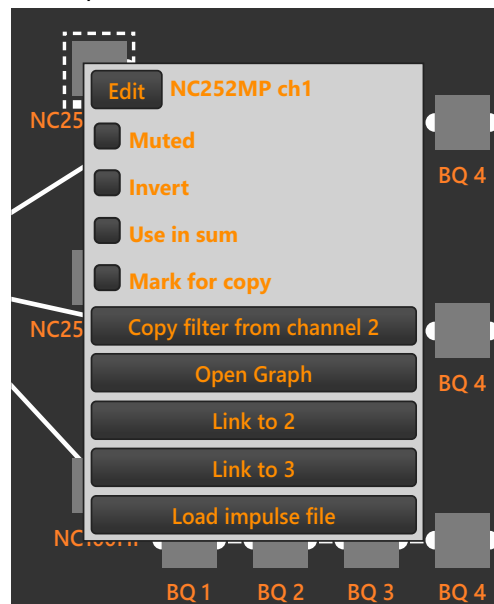
In the advanced screen you can:

- 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 slave units max.)
- 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 24: Right click on the channel box

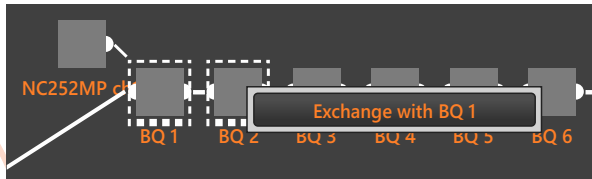
Here you can:

- 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 submenu

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



Picture 25: 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. 21 $\mu$ s, is because of the sample rate. We can only delay per whole sample, so the step count depends on the sample rate.

## 6. Adding FIR filter(s)

DSP3-213 (firmware v5.7+) and DSP3-224 have both DSP options for implementation of FIR filters. There are two places where the FIR filters can be implemented:

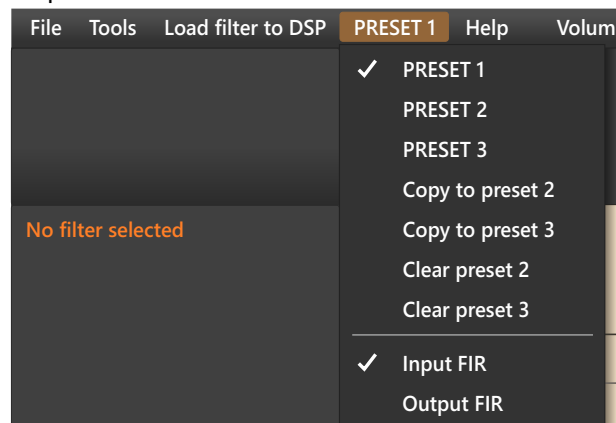
1. Right after the input multiplexer, just before the 15 biquads.
2. After the 15 biquads.

In case of 1: One 4500 (DSP3-213) or one 2048 (DSP3-224) tap FIR filter is inserted at the beginning of the audio processing path.

In case of 2: Each channel has its own 1500 (DSP3-213) or 1024 (DSP3-224) tap FIR filter.

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

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.

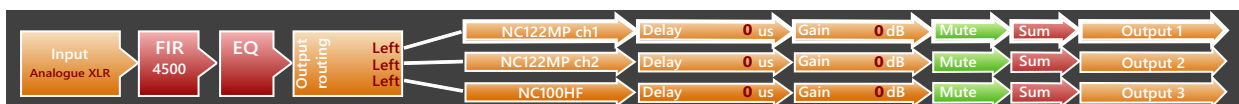


Picture 26: Select Input / Output FIR

### Creating FIR filters

From hereon, all examples are shown for the DSP3-213, but are also applicable for the DSP3-224, but with different number of taps for the FIR filter.

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 in picture 27:



Picture 27: Routing overview

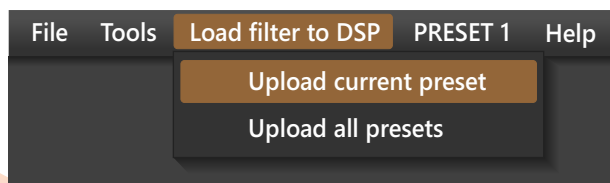
**FIR**  
**4500**

```
0  
0  
0.000000000000000000019144422693412  
-0.00000000000000000074734570353892  
0.000000000000000000167927999145463  
-0.000000000000000000303050172238464  
0.000000000000000000048342918881767  
-0.00000000000000000071040137514036  
0.00000000000000000009359180883242
```


Picture 28: Text file content example

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.

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



Picture 29: Upload project to DSP

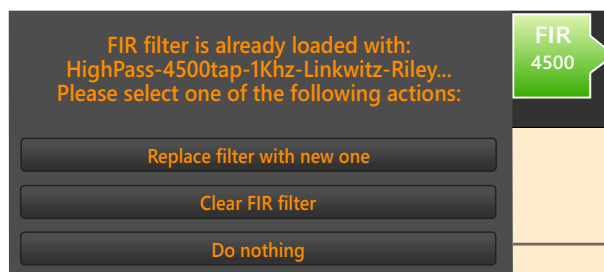
bypassed, so to enable it, just hit  with your left mouse button to toggle the state between 'enabled' (green) and 'bypassed' (red).

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.

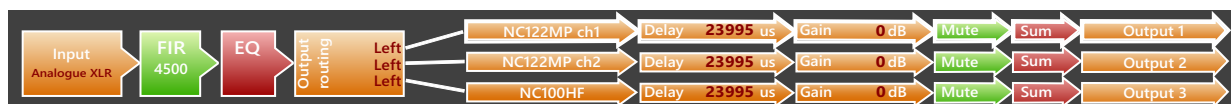
**TIP:** Presets uploaded to the FusionAmp will also store the filename of the FIR filter, so give your FIR filter file a name representing the contents, so you can remember what kind of filters you implemented in the FIR.

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 31: Select action for existing FIR filter



### Picture 30: Routing overview

## 7. Designing Filters

Use the graphical design screen to build your filters.

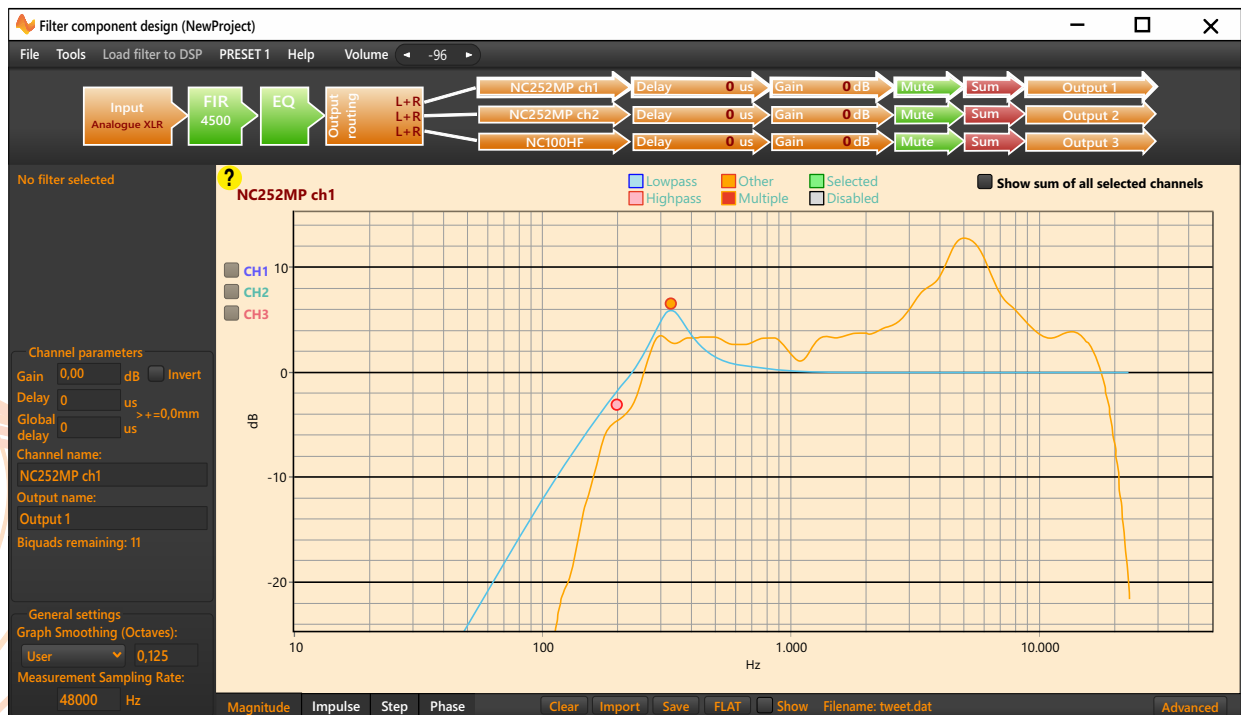
The first step is equalising the magnitude responses of the drivers flat over their entire useable frequency range.

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 32: Graphical filter design screen

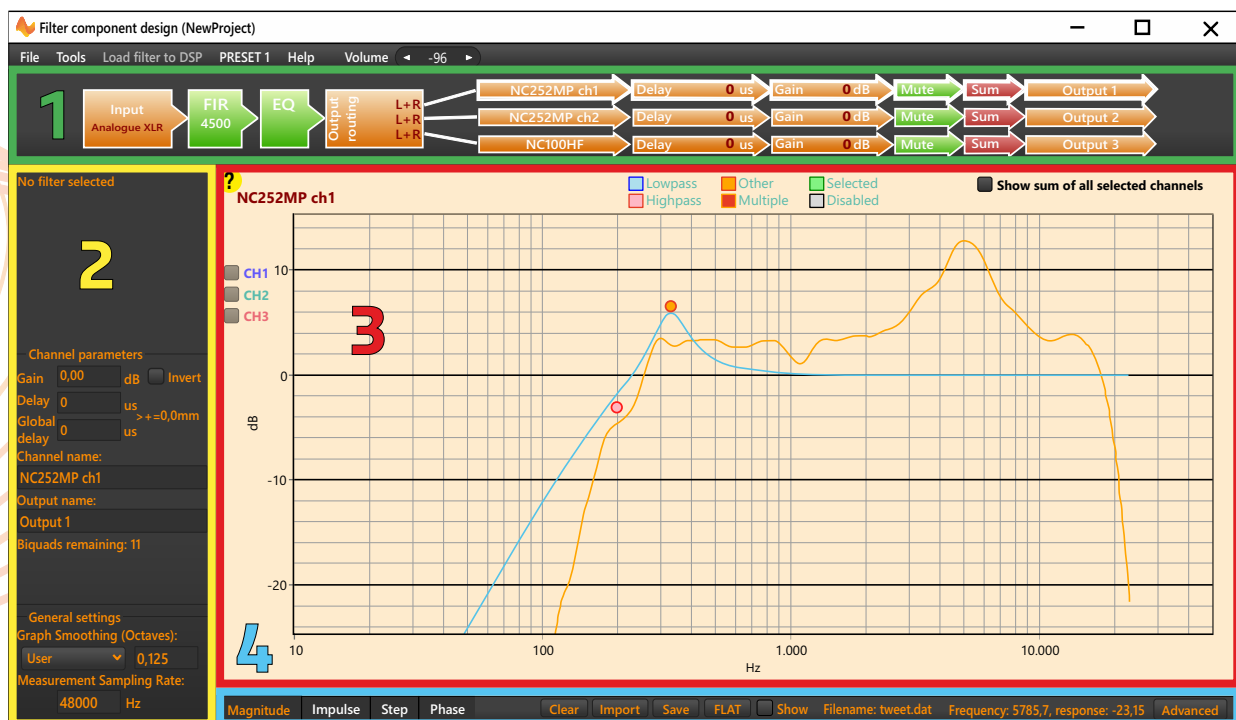
## Biquad filter types

Biquad function	Parameters	Use
Unity	-	Selection 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.
Shelf 1	Centre Frequency (halfway point) Gain / Direction	First order shelf. Usefull for baffle-step correction.
Shelf 2	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 woovers with large magnets.
Boost/Cut	Centre Frequency / Q / Gain	Dip/peak filter.
Allpass 1	Centre Frequency	Phase correction.
Allpass 2	Centre Frequency / Q	Phase correction.

Table 2: Biquad filter types

The graphical filter design screen has four main parts (see picture 27):

- **Header (1)** with channel options
- **Left pane (2)** with filter object parameters, channel parameters, general settings and recording options
- **Right pane (3)** with graphical representation of the selected channel
- **Bottom pane (4)** with tabs for Magnitude, Impulse, Step and Phase graph, Clear/Import/Save/Flat and Advanced buttons and cursor position



Picture 33: 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.

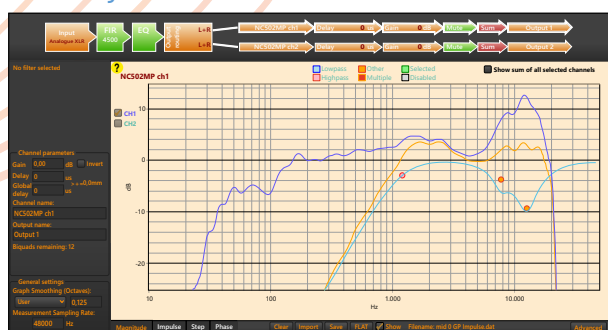
### Graph right pane (filter graph)

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 34: 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 temporarily 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. They 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):

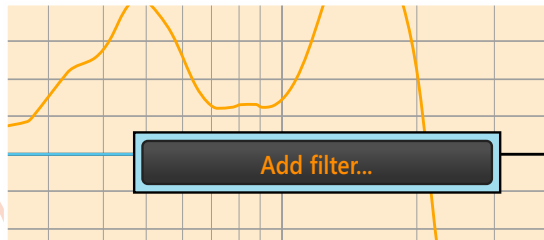


Picture 35: Select a channel in the graphic screen

In this case "Channel1/High" is selected (its surrounded by a thick bright line).

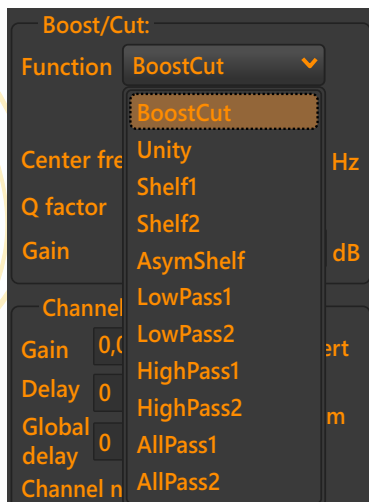


To add filters to the selected channel, right-click in the graph at the frequency-position you want the biquad. A popup menu will appear:



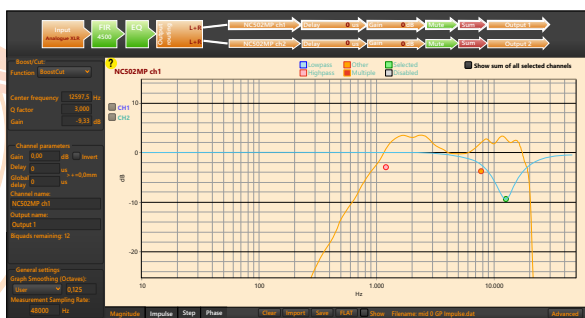
Picture 36: Add a filter object

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



Picture 37: 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":



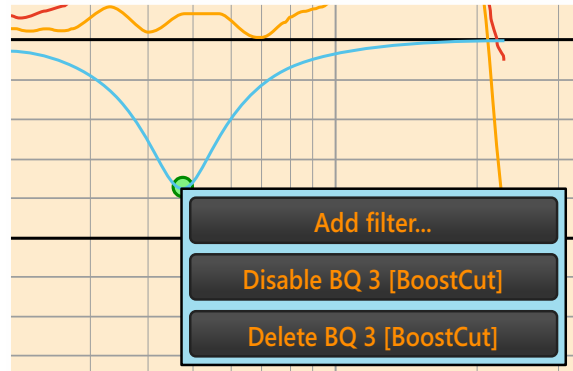
Picture 38: Filter object curve

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.

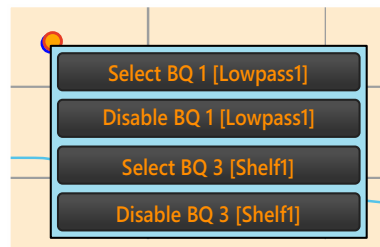
### Disable or delete filter objects

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



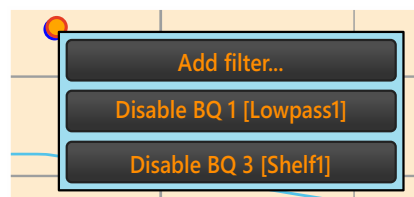
Picture 39: Disable or delete a filter object

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



Picture 40: Left mouse click

Right mouse-click:



Picture 41: 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.

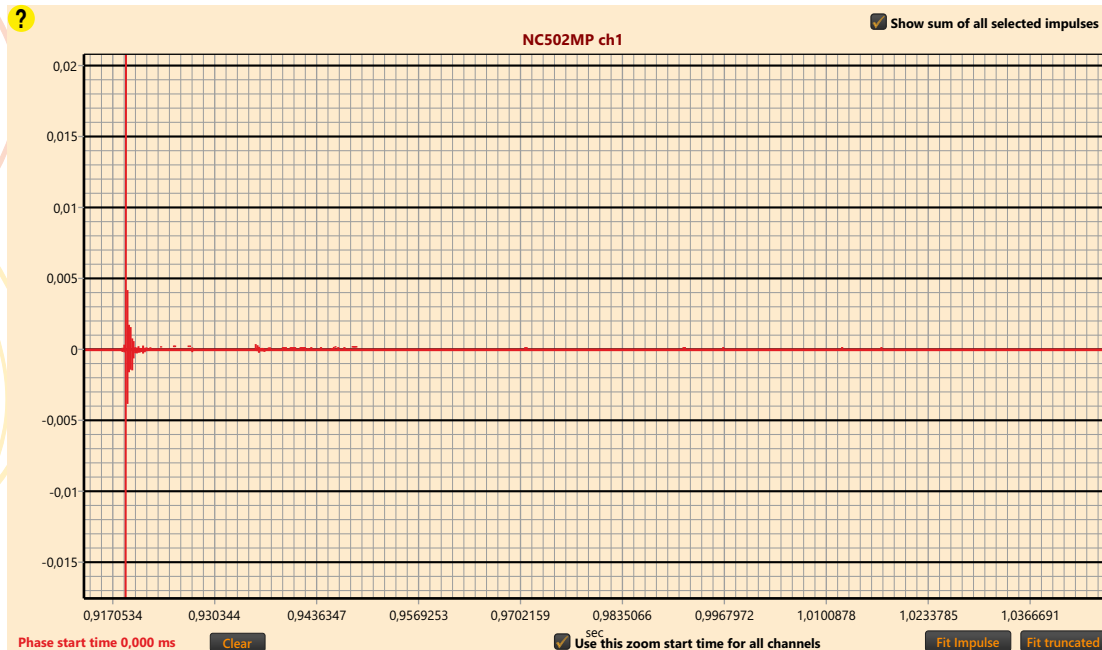
### Impulse response graph

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.

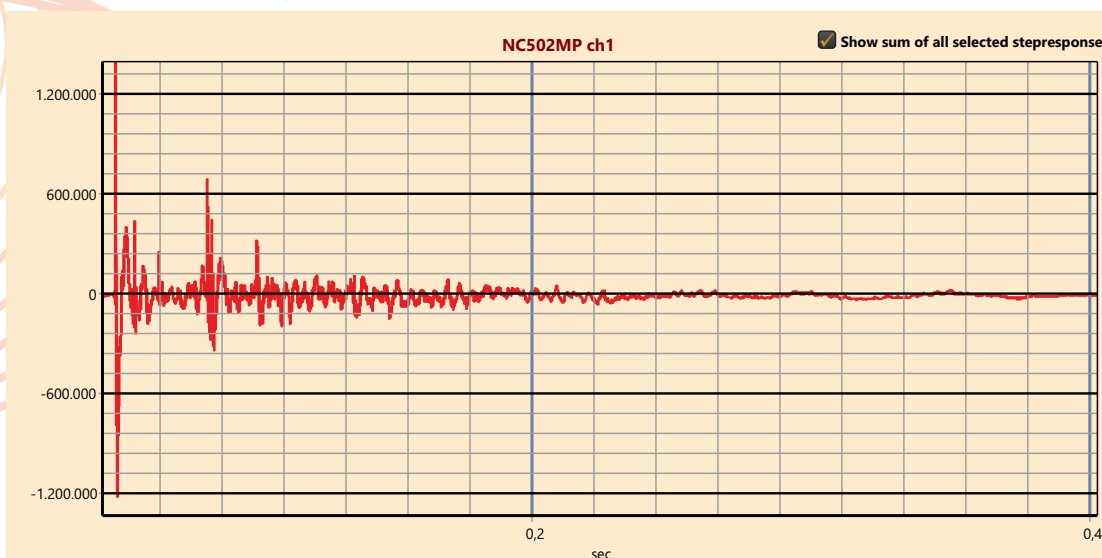


Picture 42: Impulse response graph

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 the 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 43: Step response graph



## 8. 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:

Order	Q-factor 1 <sup>st</sup> biquad	Q-factor 2 <sup>nd</sup> biquad	Q-factor 3 <sup>rd</sup> biquad
2	0.71	-	-
3	First order filter, no Q	1.00	-
4	0.54	1.31	-
5	First order filter, 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)

First order lowpass filter:

Function **LowPass1** ▼

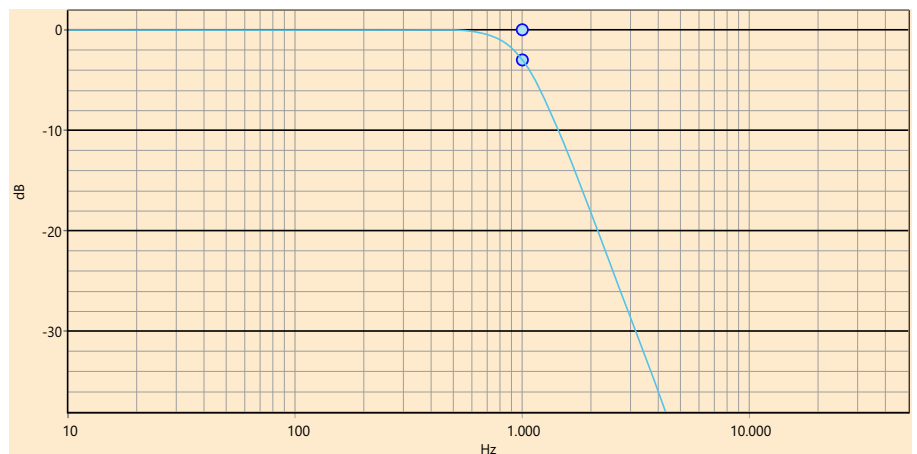
Cut-off frequency **1000,0** Hz

Second order lowpass filter:

Function **LowPass2** ▼

Cut-off frequency **1000,0** Hz

Q factor **1,0**



Picture 44: Third order low pass Butterworth example

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

Second order lowpass filter:

Function **LowPass2** ▼

Cut-off frequency **1000,0** Hz

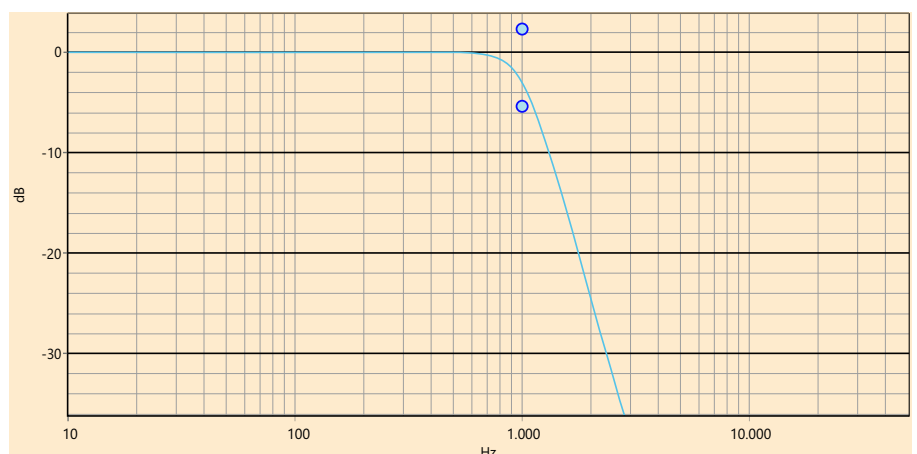
Q factor **0,54**

Second order lowpass filter:

Function **LowPass2** ▼

Cut-off frequency **1000,0** Hz

Q factor **1,31**



Picture 45: Fourth order low pass Butterworth example

### Fifth order low pass Butterworth example ( $F_c=1000\text{Hz}$ )

First order lowpass filter:

Function LowPass1

Cut-off frequency 1000,0 Hz

Second order lowpass filter:

Function LowPass2

Cut-off frequency 1000,0 Hz

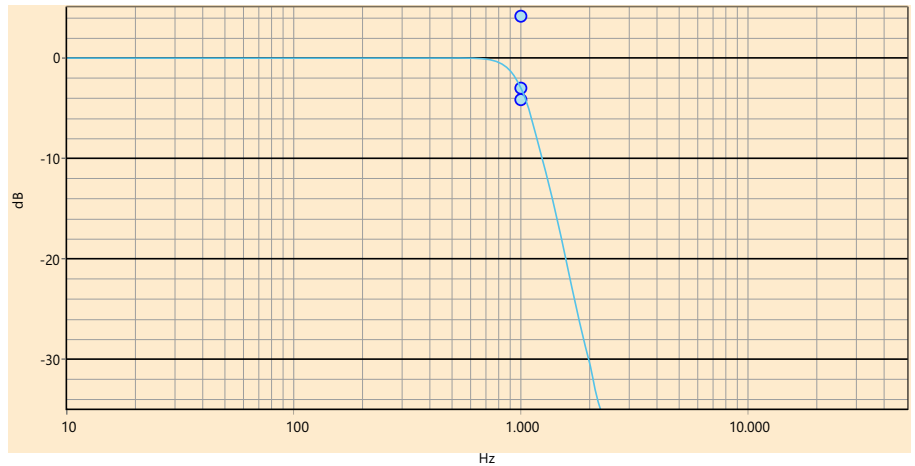
Q factor 1,62

Second order lowpass filter:

Function LowPass2

Cut-off frequency 1000,0 Hz

Q factor 0,62



Picture 46: Fifth order low pass Butterworth example

### Sixth order low pass Butterworth example ( $F_c=1000\text{Hz}$ )

Second order lowpass filter:

Function LowPass2

Cut-off frequency 1000,0 Hz

Q factor 0,52

Second order lowpass filter:

Function LowPass2

Cut-off frequency 1000,0 Hz

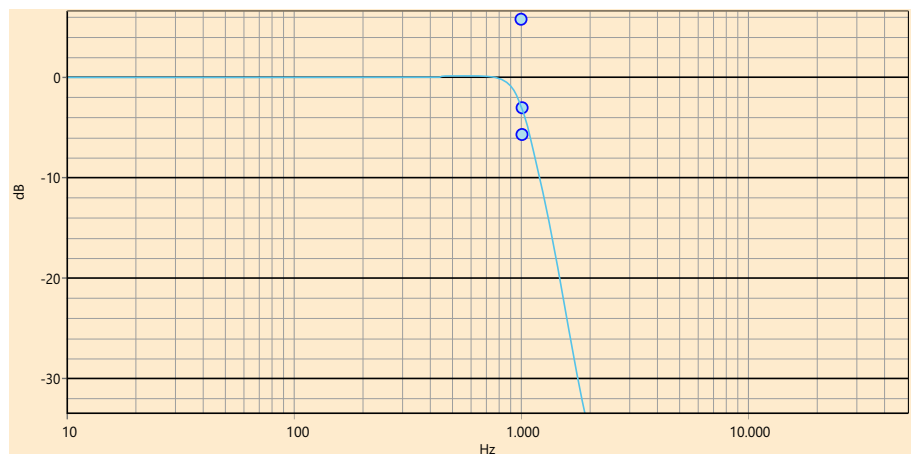
Q factor 1,93

Second order lowpass filter:

Function LowPass2

Cut-off frequency 1000,0 Hz

Q factor 0,71



Picture 47: 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, 4th and 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 0.50 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 1 kHz.

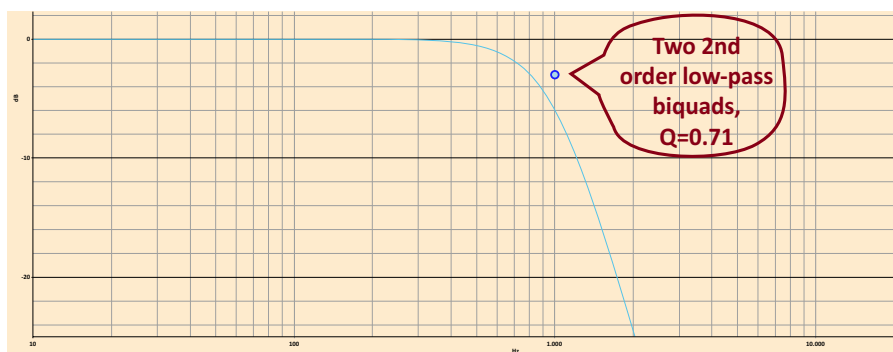
Create with two low-pass biquads on channel 1 and 2 high-pass biquads on channel 2. The two low-pass and two high-pass biquads have a Q-factor of 0.71 as in cascaded Butterworth filters:

For channel 1 (low frequency driver)

Second order lowpass filter:  
 Function LowPass2  
 Cut-off frequency 1000,0 Hz  
 Q factor 0,71

Second order lowpass filter:  
 Function LowPass2  
 Cut-off frequency 1000,0 Hz  
 Q factor 0,71

Picture 48: Ch1 biquad settings

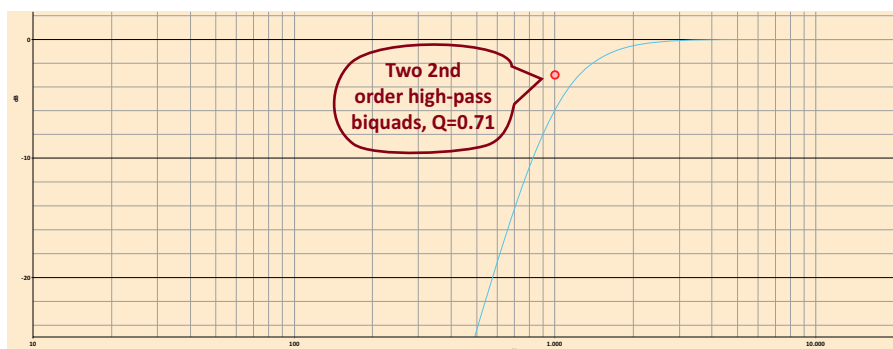


For channel 2 (high frequency driver)

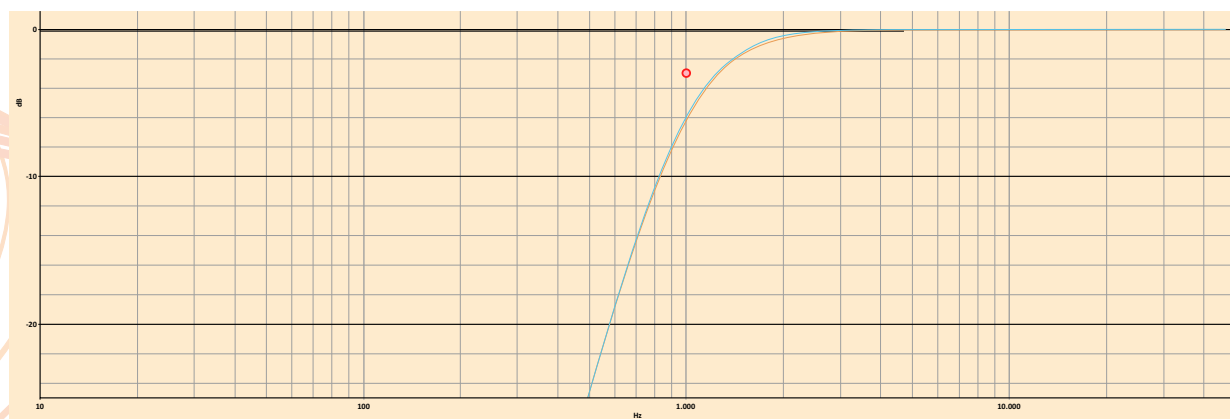
Second order lowpass filter:  
 Function HighPass2  
 Cut-off frequency 1000,0 Hz  
 Q factor 0,71

Second order lowpass filter:  
 Function HighPass2  
 Cut-off frequency 1000,0 Hz  
 Q factor 0,71

Picture 49: Ch2 biquad settings



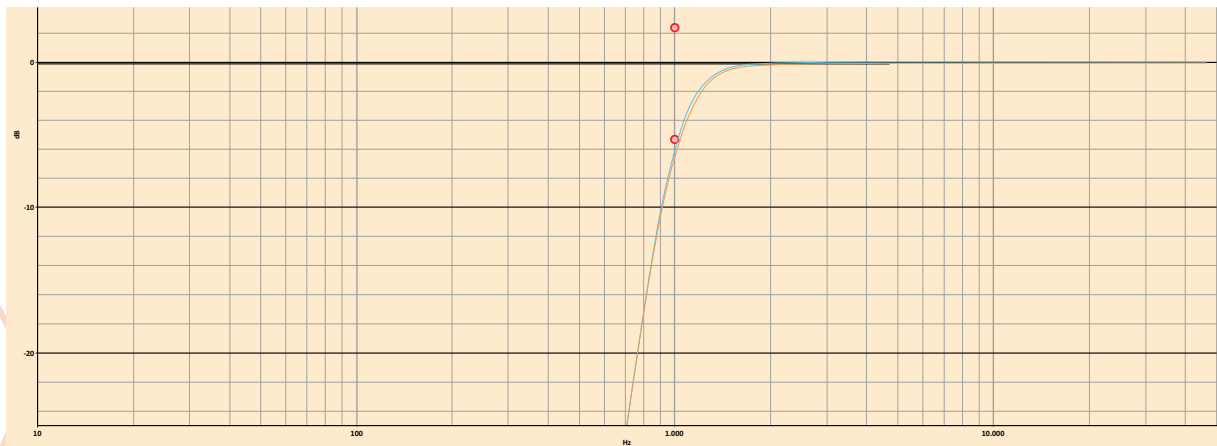
Looking at the sum-result of both channels (you can simulate this in HFD by loading a flat impulse file on both channels), you can observe that the result of the Linkwitz-Riley filter is flat over the total frequency range (black line just below 0dB):



Picture 50: Sum result of both channels

LR Order	Q-factor 1 <sup>st</sup> biquad	Q-factor 2 <sup>nd</sup> biquad	Q-factor 3 <sup>rd</sup> biquad	Q-factor 4 <sup>th</sup> biquad	Remarks
2	0.50	-	-	-	
<b>4</b>	<b>0.71</b>	<b>0.71</b>	-	-	<b>Preferred crossover filter</b>
8	0.54	1.31	0.54	1.31	

Table 4: Linkwitz-Riley coefficients



Picture 51: 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 5 shows the various cut-off frequency multiply factors and Q-factors for Bessel filters from 2nd to 6th order:

Order	Q-factor 1 <sup>st</sup> biquad	1 <sup>st</sup> frequency factor	Q-factor 2 <sup>nd</sup> biquad	2 <sup>nd</sup> frequency factor	Q-factor 3 <sup>rd</sup> biquad	3 <sup>rd</sup> frequency factor
2	0.58	1.27	-	-	-	-
3	1 <sup>st</sup> order filter, no Q	1.32	0.69	1.45	-	-
4	0.81	1.60	0.52	1.43	-	-
5	1 <sup>st</sup> order filter, no Q	1.50	0.92	1.76	0.56	1.56
6	1.02	1.90	0.61	1.69	0.51	1.60

Table 5: Bessel filter coefficients

Filter object cut-off frequency =  $F_c \times \text{frequency factor}$

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

Biquad	Cut-off frequency	Q-factor
1	$1.60 \cdot 1000 = 1600\text{Hz}$	0.81
2	$1.43 \cdot 1000 = 1430\text{Hz}$	0.52

Table 6: Example 4th order Bessel filter  $F_c=1000\text{Hz}$

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	$1.90 \cdot 1000 = 1900\text{Hz}$	1.02
2	$1.69 \cdot 1000 = 1690\text{Hz}$	0.61
3	$1.60 \cdot 1000 = 1600\text{Hz}$	0.51

Table 7: Example 6th order Bessel filter  $F_c=1000\text{Hz}$

#### Fourth order low pass Bessel example ( $F_c=1000\text{Hz}$ )

Second order lowpass filter:

Function

Cut-off frequency  Hz

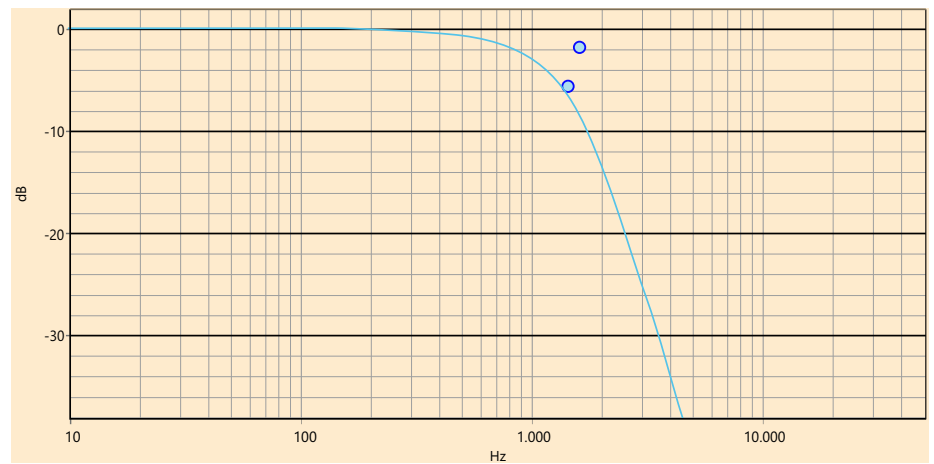
Q factor

Second order lowpass filter:

Function

Cut-off frequency  Hz

Q factor



Picture 52: Fourth order low pass Bessel example

#### Sixth order low pass Bessel example ( $F_c=1000\text{Hz}$ )

Second order lowpass filter:

Function

Cut-off frequency  Hz

Q factor

Second order lowpass filter:

Function

Cut-off frequency  Hz

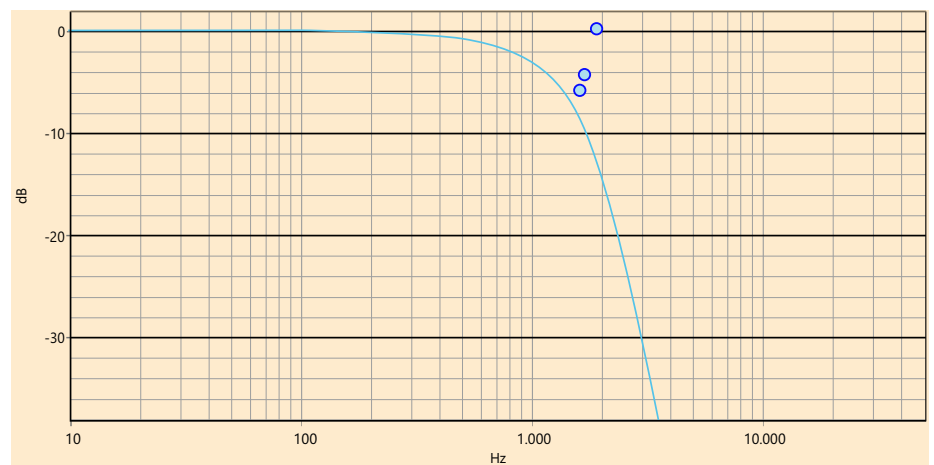
Q factor

Second order lowpass filter:

Function

Cut-off frequency  Hz

Q factor

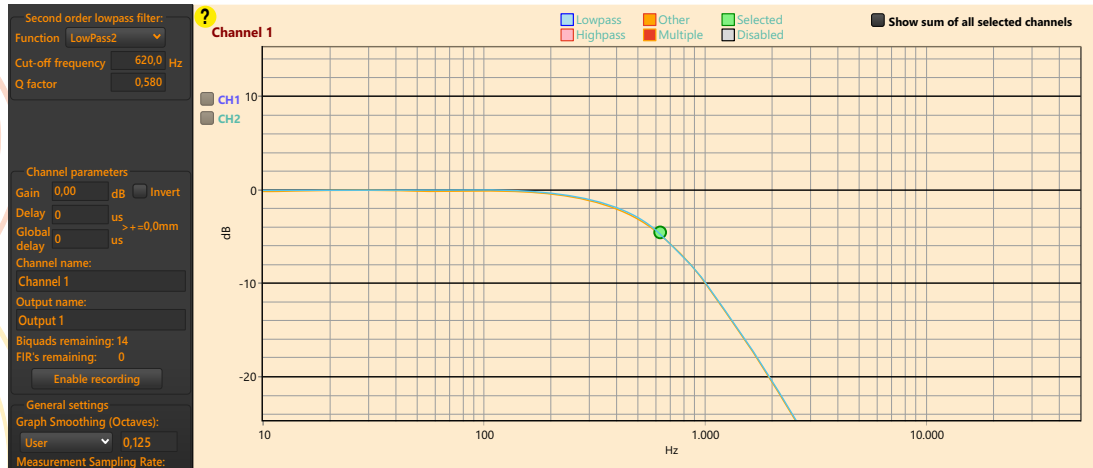


Picture 53: Sixth order low pass Bessel example

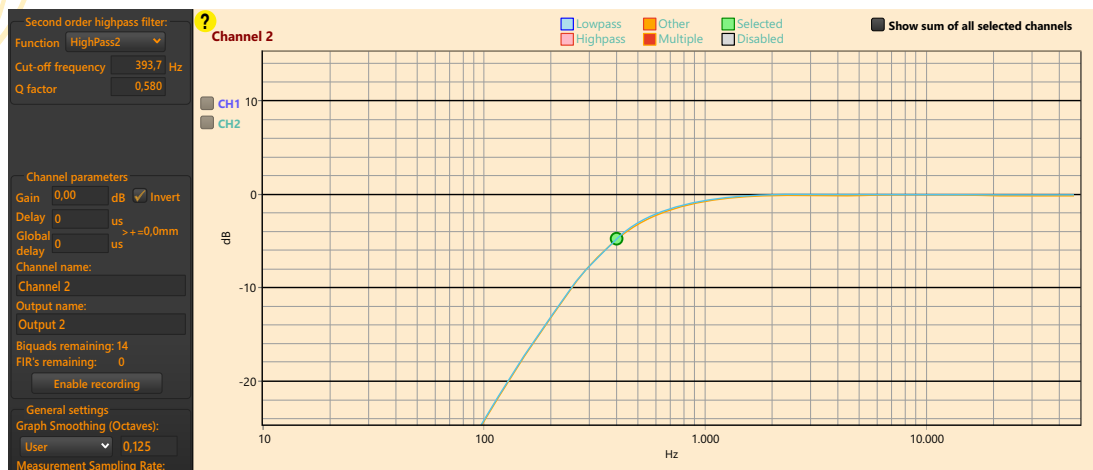
## Crossover using Bessel

When creating a crossover filter, use  $1/\text{freqFact}$  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.



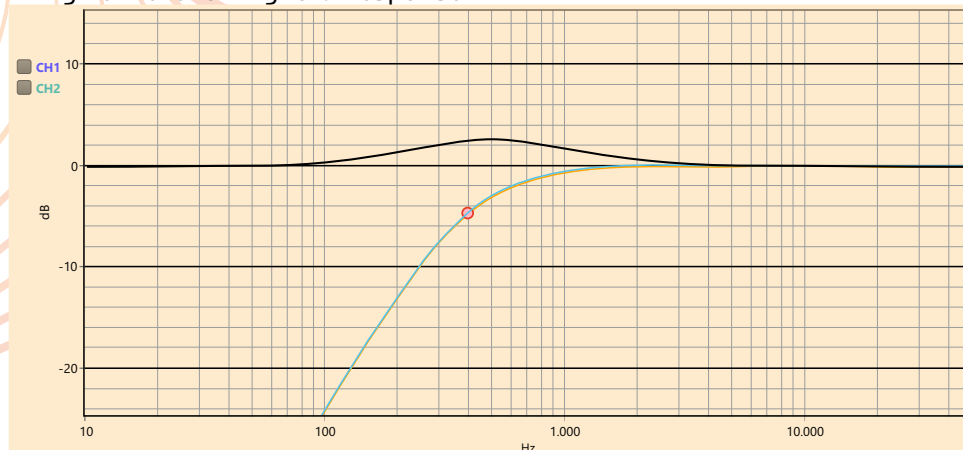
Picture 54: Ch1 low pass filter



Picture 55: Ch2 high pass filter

Notice the inverted channel 2

Will give the following total response:

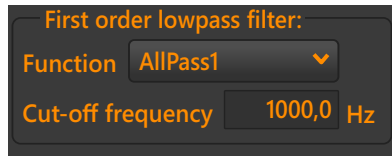


Picture 56: Total response

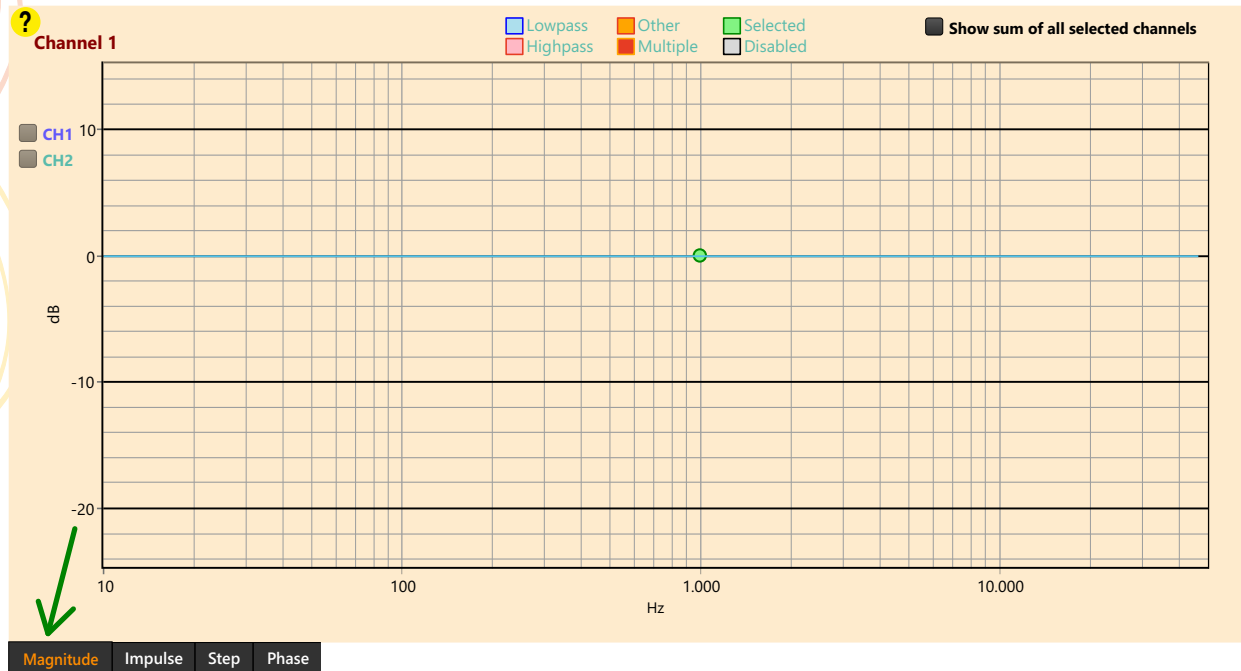
At the designed crossover frequency (500Hz) is a slight boost visible.

## All pass 1

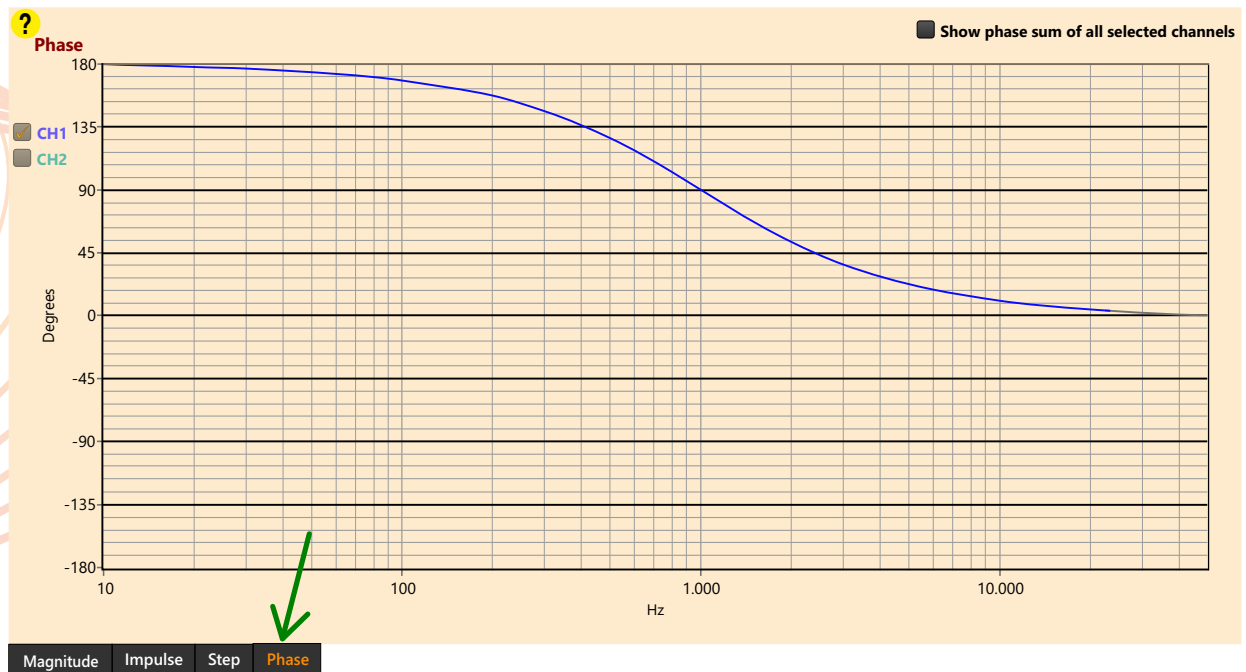
All pass filters are implemented as 1<sup>st</sup> and 2<sup>nd</sup> order. An 'All pass 1' filter will have a gain of 0dB, but will drop the phase over increasing frequency from 180° to 0°.



Picture 57: All pass 1 filter



Picture 58: Gain of 0dB over the whole frequency scale

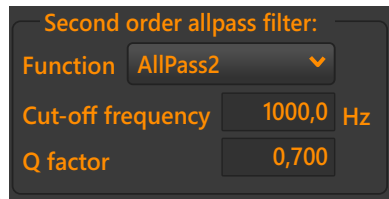


Picture 59: At 1kHz this filter will have a phase of 90°

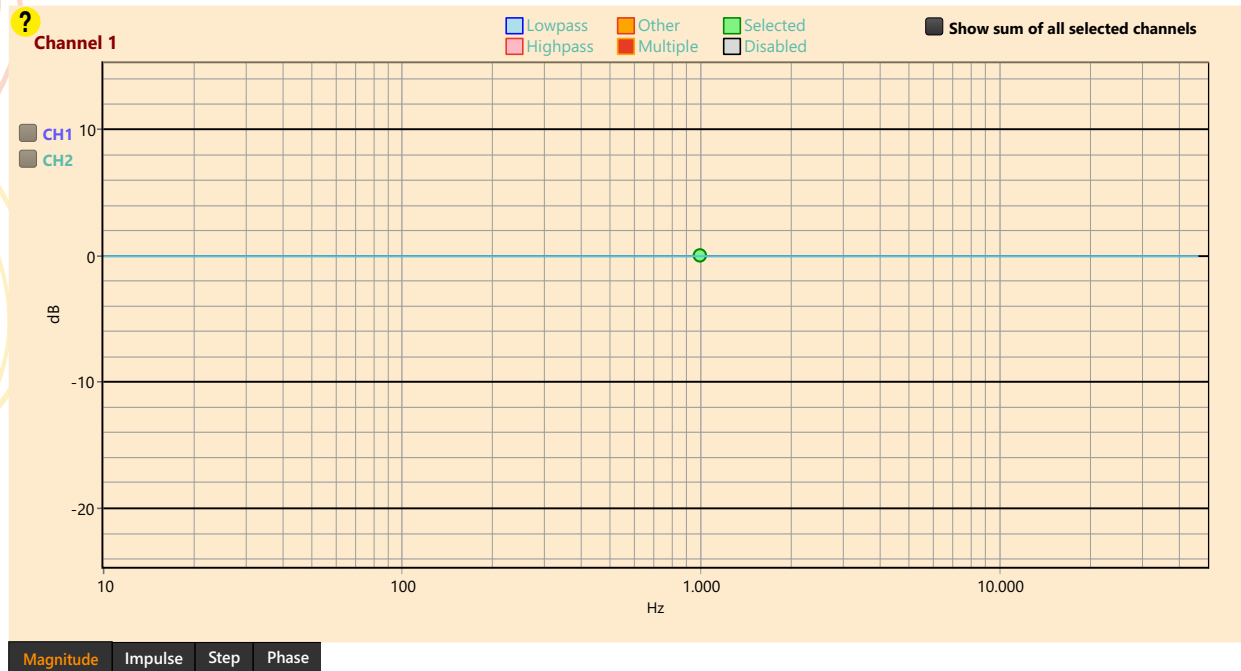


## All pass 2

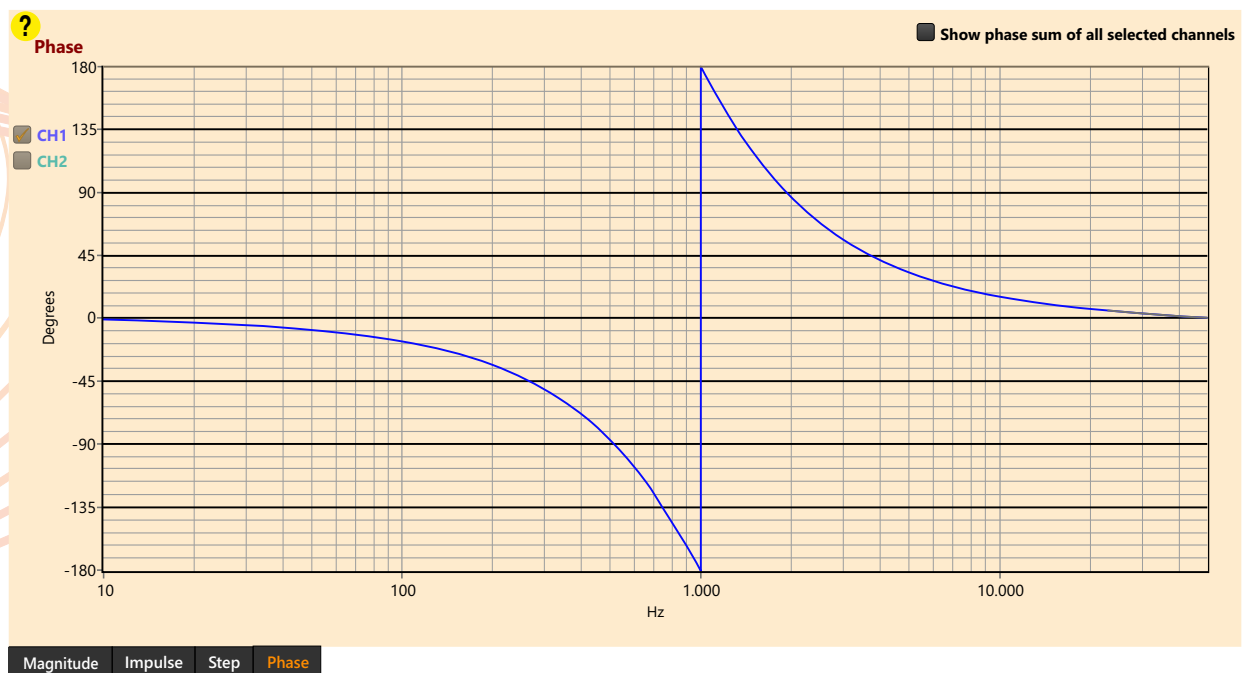
As with the 1<sup>st</sup> order All pass filter, the 'All pass 2' also will add no gain, but does add a phase shift of 360° in total:



Picture 60: All pass 2 filter



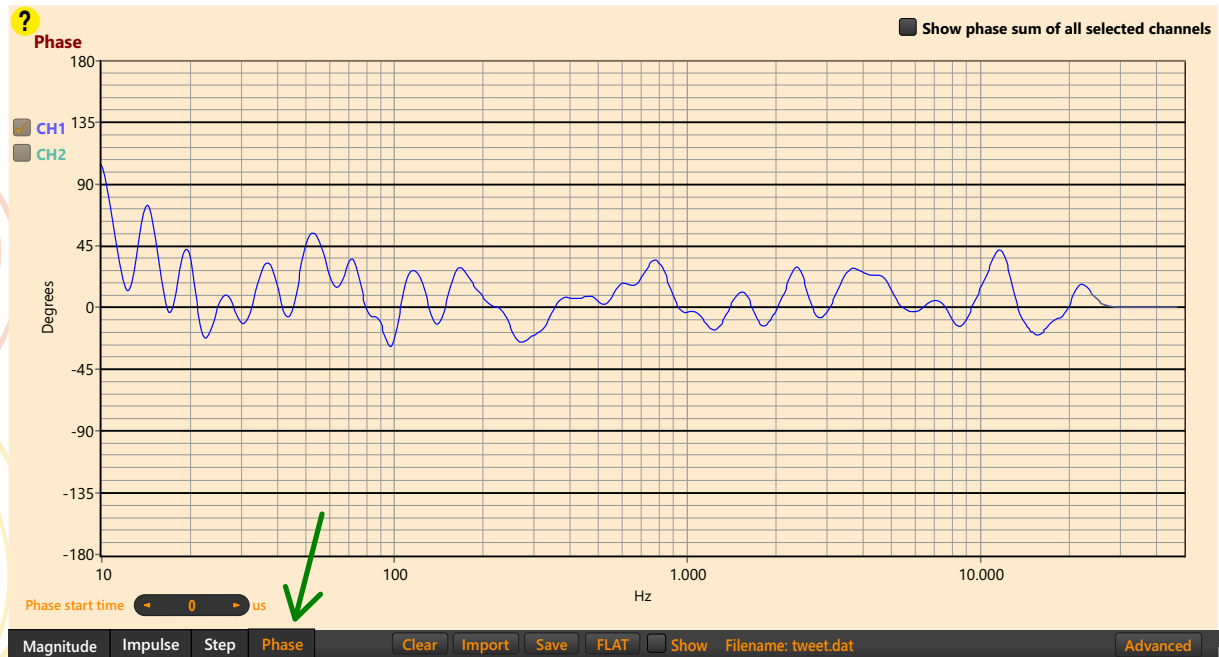
Picture 61: Gain of 0dB over the whole frequency scale



Picture 62: Phase response starts at 0° and will drop to -180° at its centre frequency

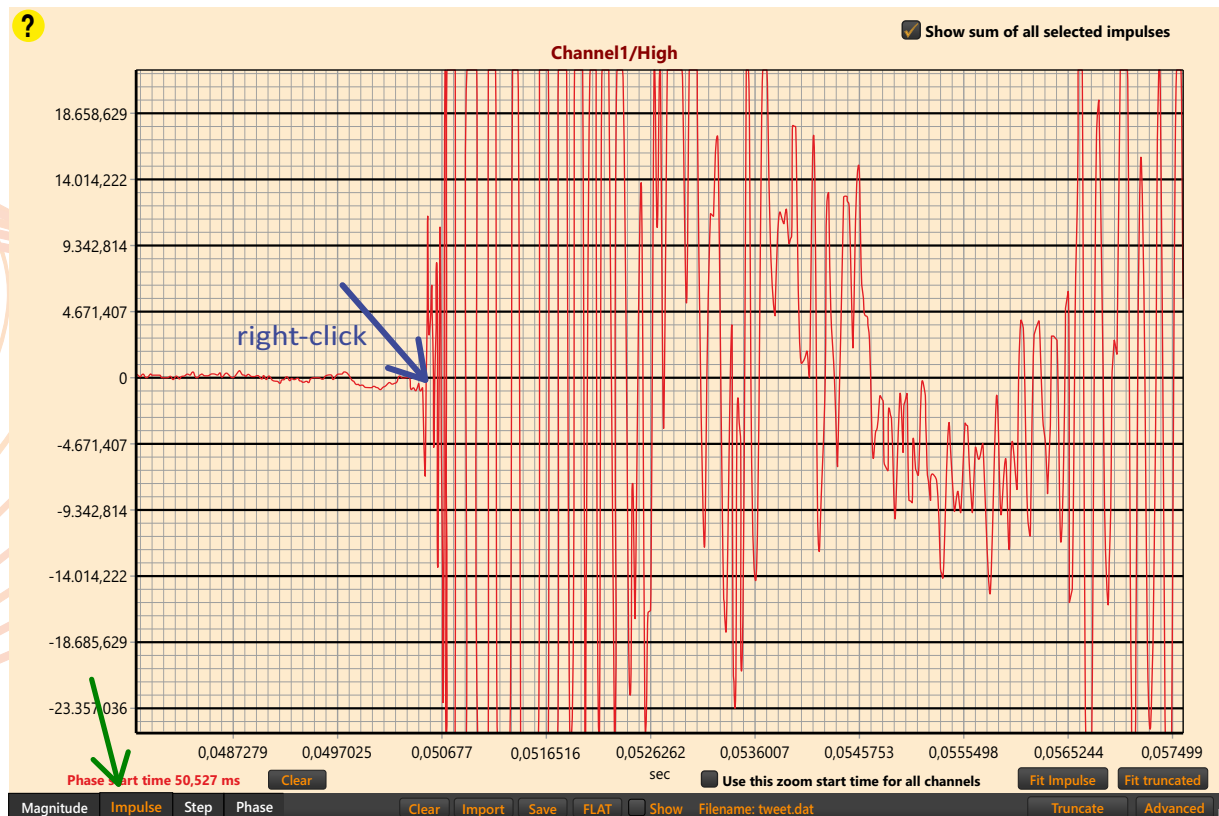
## Phase plot

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



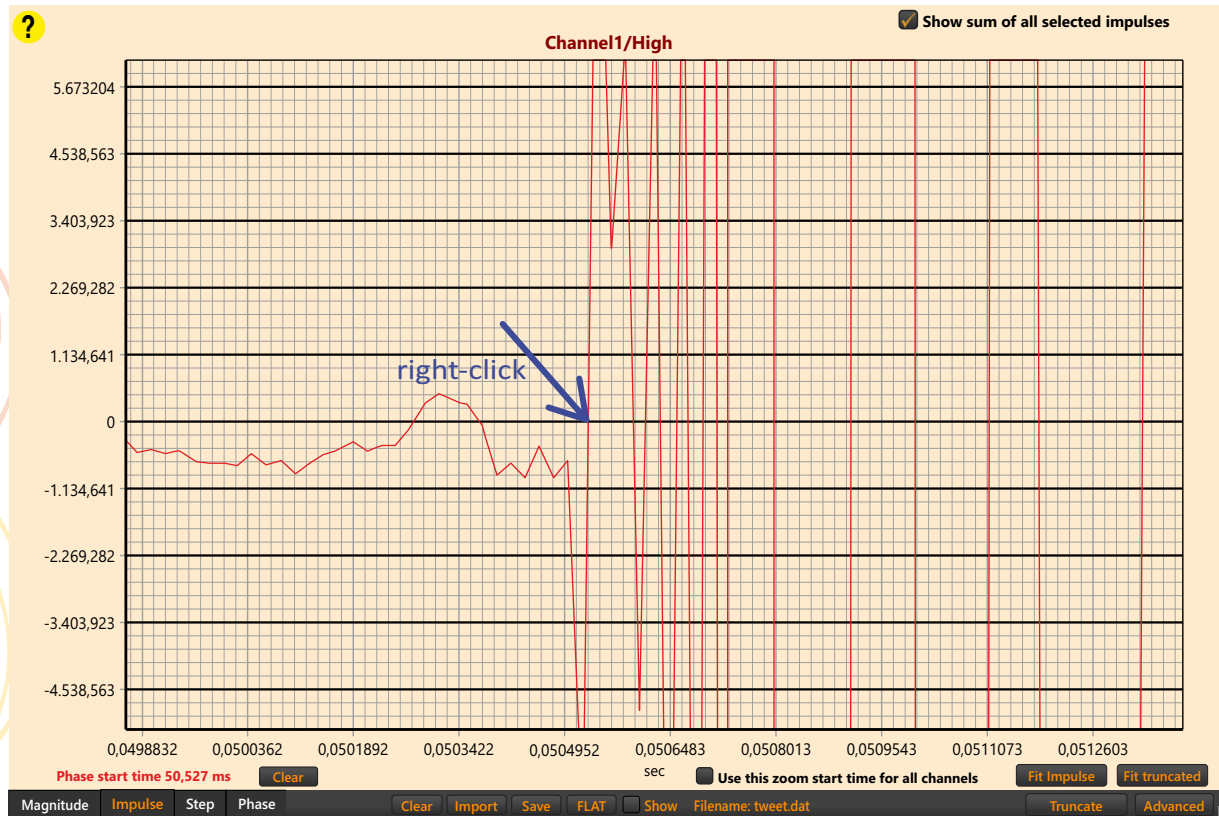
Picture 63: Phase plot

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 58, it will set the start time:



Picture 64: Set start time

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



Picture 65: 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:

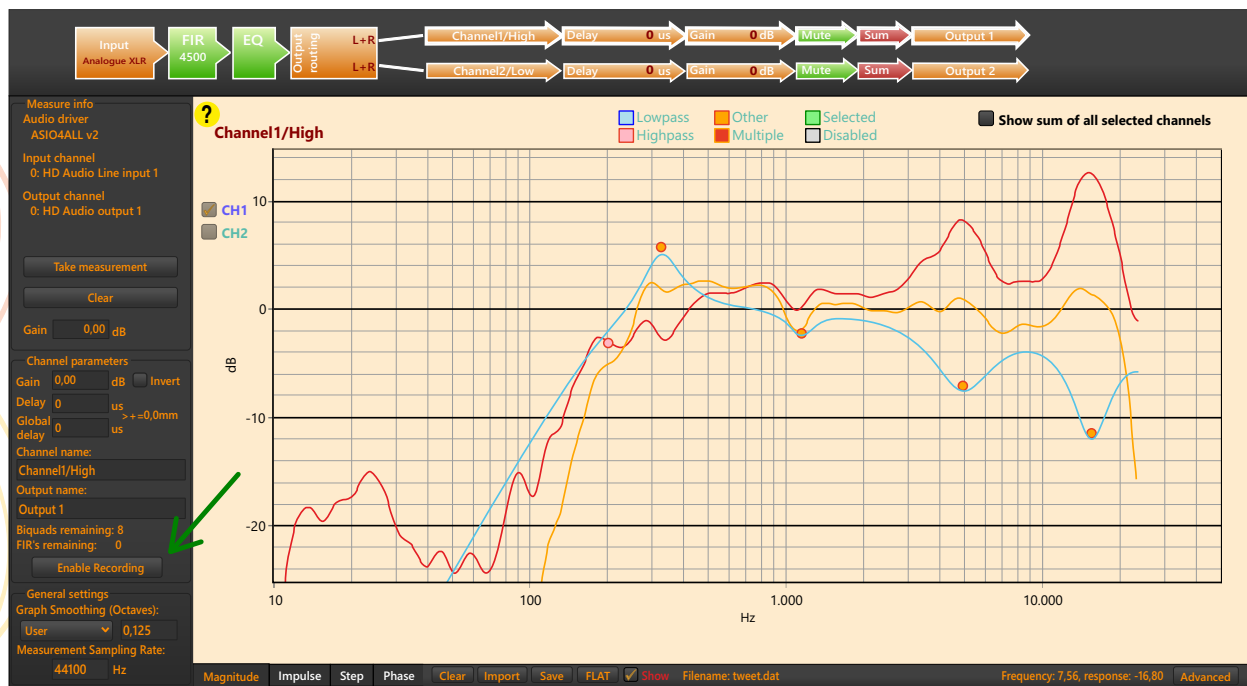


Picture 66: Phase plot after start time adjustment

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.

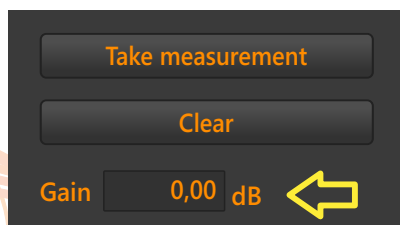
## Recording

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 67: 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:



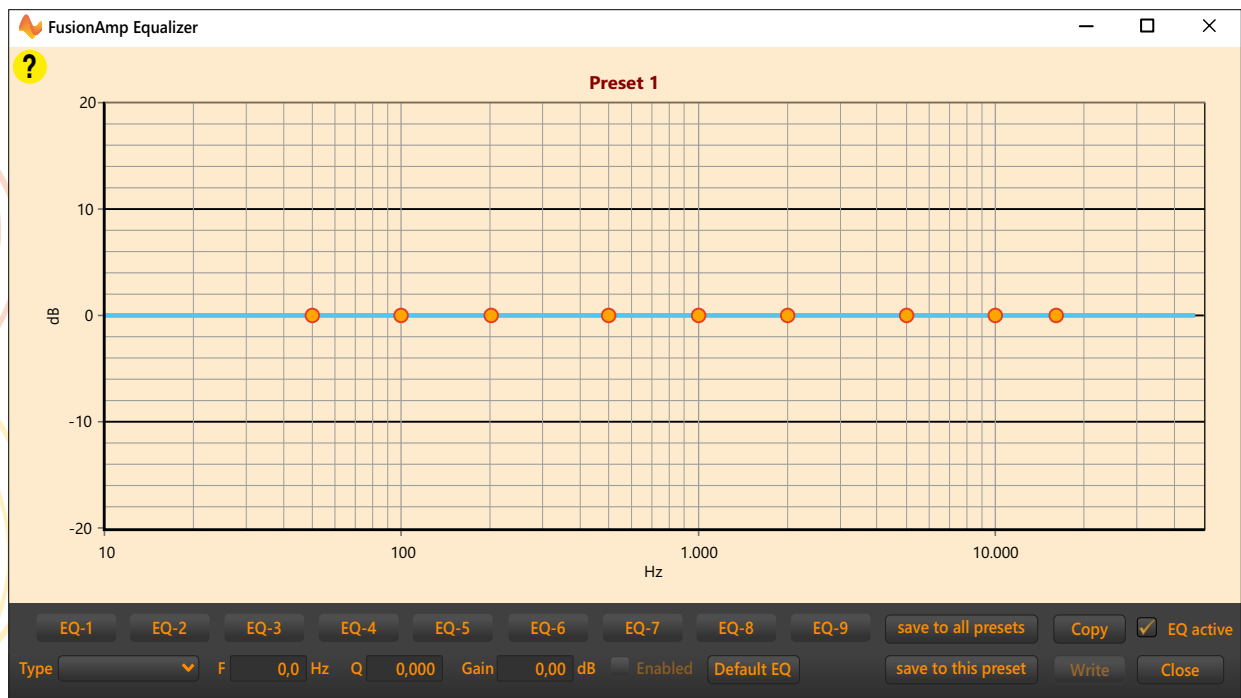
Picture 68: Change sweep output gain

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.

## 9. Equalizer

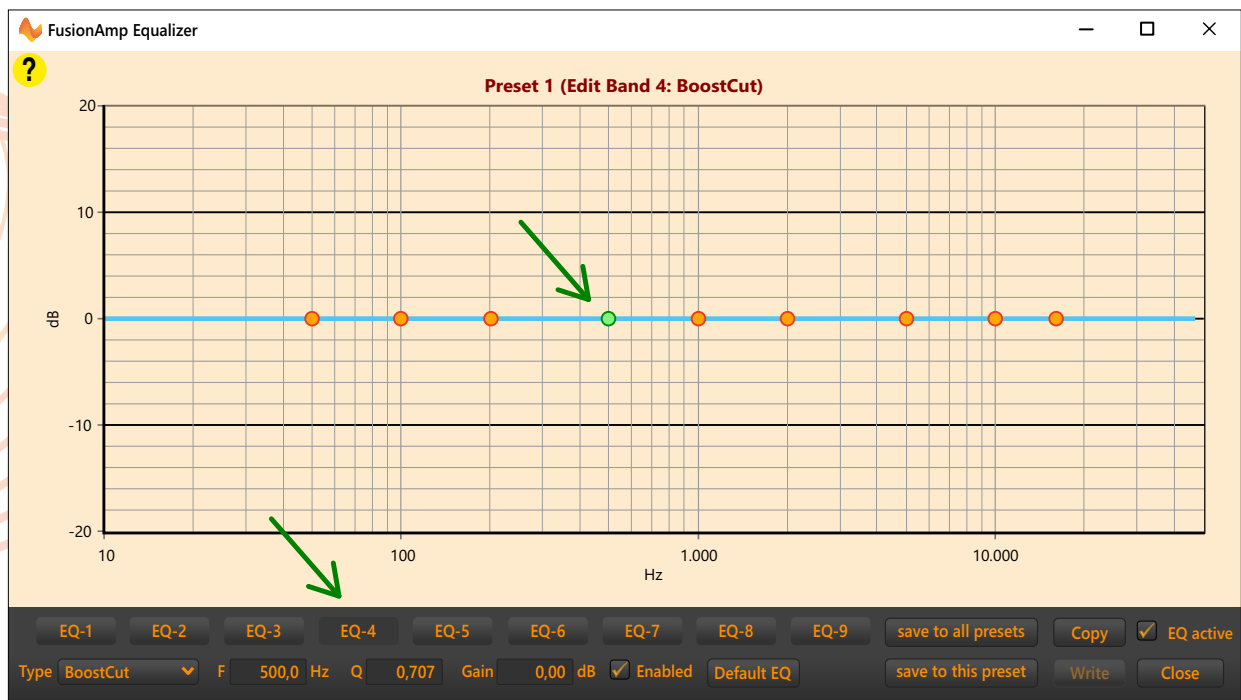
A parametric equalizer is available in the DSP3-224 and is added to the DSP3-213 as of firmware version v1.4. The equalizer is placed right after the input selection, before the signal is provided to the three channels.



Picture 69: Equalizer screen, no band selected

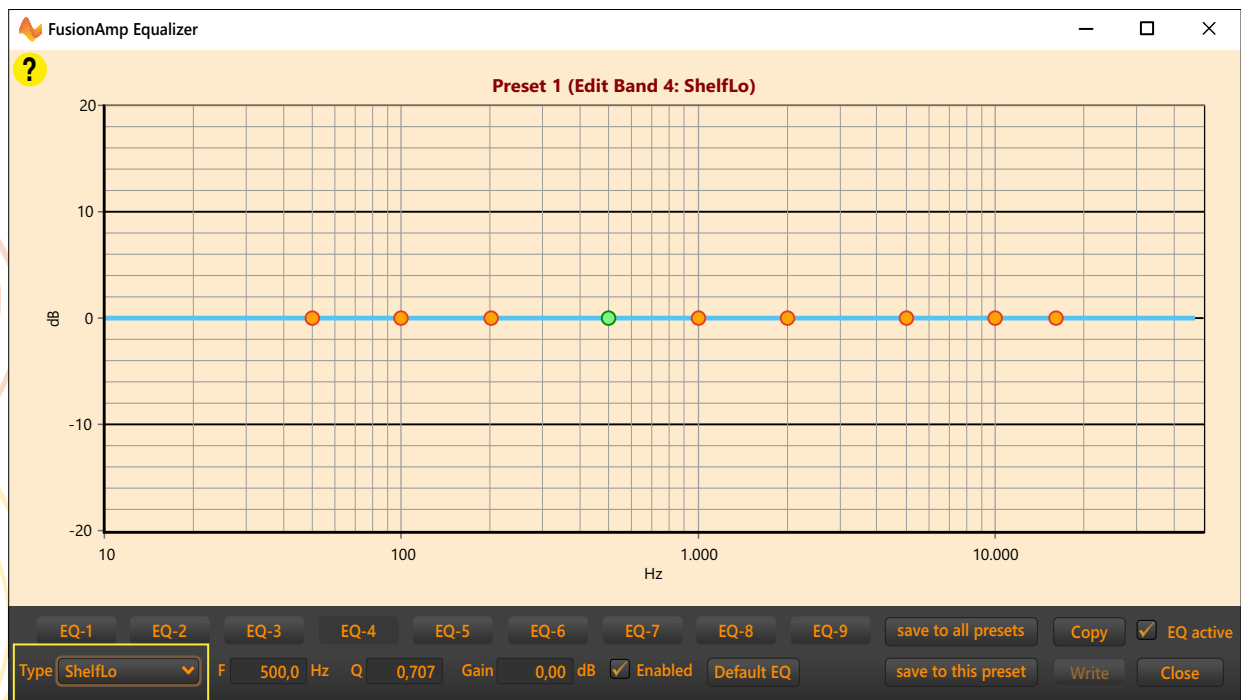
By default, all biquads are of type “BoostCut” with a gain of 0dB, making the equalizer to pass audio without any alterations. Up to 9 biquads are available to prepare the signal, before it is fed to the output router. Therefore, all three channels use the same equalizer. The equalizer settings are stored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



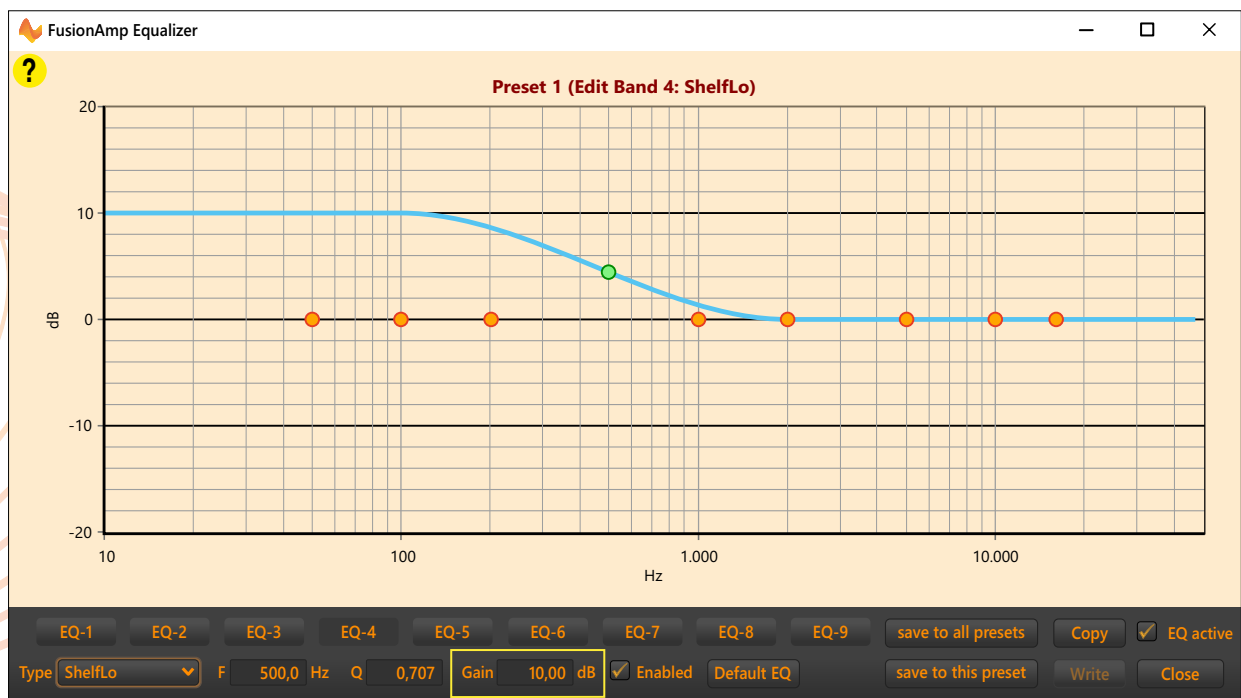
Picture 70: Equalizer screen, preset 1, band 4 selected: BoostCut

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.



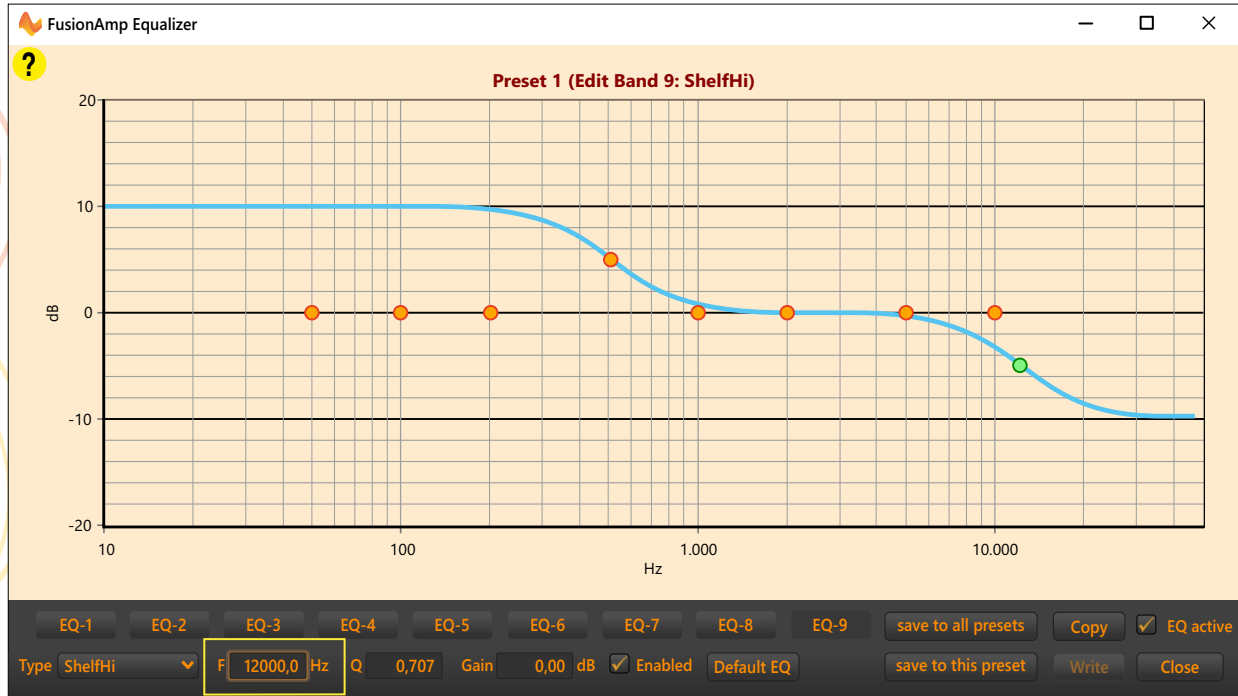
Picture 71: Equalizer screen, band 4 selected and filter type changed to ShelfLo

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 field and rotate the mouse wheel to change the value in the box under the cursor.



Picture 72: Equalizer screen, band 4 selected and Gain changed to 10dB

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

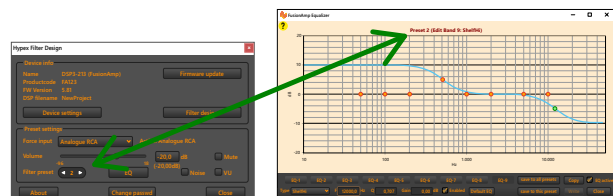
As you can see in the picture above, one biquad is coloured orange (not selected) and one is green (selected). You can also select the left biquad (of band 4) by clicking on it with your mouse.

### Saving presets

Updates to connected devices are not done automatically. Therefore, after changes are made, these must be saved to the DSP by uploading them, using the **save to .....** buttons. You can save the changes of the current selected preset or changes of all 3 presets at once. The equalizer will have immediate effect. After pressing **save to this preset**, the 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.



Picture 74: Select preset in main screen to edit in Equalizer screen

### 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 device. Hit the **Copy** button to copy the EQ parameters from the DSP. HFD will remember them for you, even if the device is disconnected. The parameters are NOT remembered after closing HFD.

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

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



## Default EQ

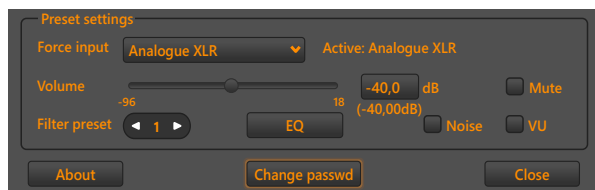
When you want to start all over with the current selected equalizer (that is, the eq for the current preset), you can set all nine biquads to their default values by hitting the **Default EQ** button. It will set all biquads to type "BoostCut", with a Q of 0.707, a gain of 0dB and a frequency of 50Hz(band 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.

## 10. Change password

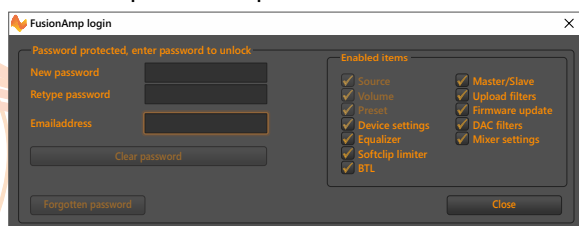
To prevent customers being able to change the settings above, the device should be protected with a password.

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



Picture 75: Change password

This will open a new password screen:

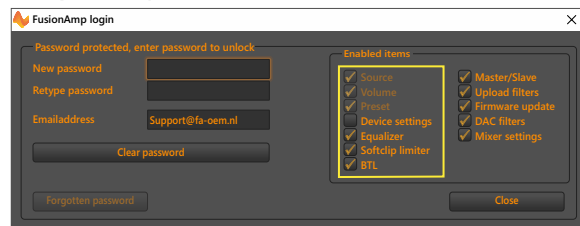


Picture 76: Password login dialog

On the left side, the standard password protection entry fields are visible, on the right (grouped as "Enabled items") you will find the options that 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 device without entering a password.

For example, if you want a user being unable to change any device settings, uncheck "Device settings" and protect the device with a password. Besides a password, a valid e-mail address is necessary. This e-mail address will

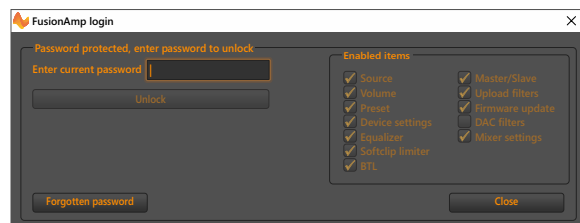
be programmed into the DSP also, in case of a forgotten password.



Picture 77: Disable Device settings

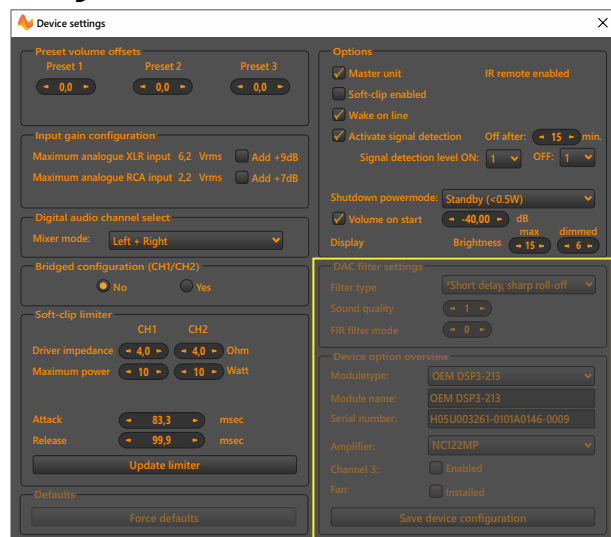
Every time a password protected device is connected to HFD, it should be unlocked first, to be able to change the "Enabled items". Once unlocked, clicking "Enabled items" will have immediate effect (no save is necessary). Needless to say that an unlocked device has all its protected items accessible.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

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




Picture 78: Enter password

In this example (for DSP3-213), 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 79: Disabled DAC settings for DSP3-213

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
R3.0	FIR implementation and DSP3-224 introduction	01-03-2025