

Hypex Filter Design

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General info

Hypex Filter Designer is a program to control Hypex Electronics B.V. amplifier modules.

The Hypex Filter Design v4 software consists of four areas:

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

On the main screen the user can control all basic settings of the amplifier like volume and input select. The device settings screen is the place to change channel assignments and fixed gain controls. In the filter designer the user can set the filters, delay, gain etc. for each channel. More about the functions can be read in the Filter Design chapter.

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.

Options

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

General functions

- Volume control
- Mute (global and per channel)
- Channel assignment
- Sub-out Control
- Load measurements for speaker (Impulse response)
- Take measurements
- Load/apply microphone correction data
- View Step-response / Impulse-response / Magnitude
- View sum of filters
- Invert filter functions
- Delay
- Graph smoothing
- Wide range of Biquad filter setups
- Input gain adapt

Volume control

Volume is normally controlled on the main screen of HFD and for some modules also through a controller/remote control.

Mute

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

Channel assignment

All version with multiple inputs have at least two ways of selecting them, actually there are three. The first is on the module itself, this functionality will be described in the manual of that module. The second and third options are controlled in the Main screen of Filter design under "Force Input". First there is a "auto detect" option. This option will automatically select one of the present inputs, for example when only SPDIF is present it will switch to SPDIF as a source. The third option is to force a input, by this you disable the automatic selection and the module will always switch to the selected source (when present).

Subout Control

On the AS2.100 there is a sub-out section added in the Control Panel. By this the source for the sub-output can be selected. For the left and right setting the same selection method is used as for the two main channels. When left is chosen on the master slave will use the right source and vice versa.

Load measurements for speaker

On the Graphic filter design screen, select the channel you want to import and click "import". On the advanced screen, right click a 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.

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.

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.

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.

Input gain adapt (only AS2.100/DLCP/FusionAmp)

The function Gain Adapt should be used to level the analogue and digital inputs. Note that this changes the maximum input voltage for the analogue input! For example (AS2.100) when you have a source with 1Volt output you can set the Gain for the analogue input to +6dB. This gives you a full scale input of 1,1Volt for the analogue input.

Getting started

Example with use of two AS2.100's in a 2-channel stereo setup (copied from the AS2.100 Manual)

This example will guide you through setting up your 2-channel stereo system with the AS.100.

The master module is always the one with the control panel connected.

First you got your 2 modules with no cables or speakers connected!

Master

1. Plug- in the power connector and switch the module to ON
2. Power up the module by the control panel or remote
3. Start the pc program, Hypex filter design
4. Plug-in the USB cable (assumed that you've already installed the software as described in chapter "Software installation")
5. Wait for automatic connection, at "Device info" on the main screen, information about the AS2.100 should be given. If there is no response, there is a problem with the USB connection or your operating system that cannot detect the USB device
6. Set the system to the desired setup, for this example 2-channel where the master represents the Left speaker. To do so, press the "Device settings" button.
7. When needed set the desired input, for now just leave it Auto detect. Note that in auto detect the USB source is selected when the module is connected to the pc
8. Switch to the Filter design Window, by pressing the button named "Filter design"
9. Fill in your biquads with the desired filters, the high channel is channel 1 and the low channel is channel 2
10. Upload the filters into your module with the "Load filter to DSP" button (found at the top of the screen)
11. When the upload is completed you can power down the module
12. Start with the SLAVE module

Slave

1. Start the pc program, Hypex filter design. Or close the filter design screen if the program is still running
2. Plug- in the power connector and switch the module to ON
3. Plug-in the USB cable (assumed that you've already installed the software as described in chapter "Software installation")
4. Wait for automatic connection, now the program detects a slave module. If there is no response there is a problem with the USB connection
5. Open the Filter design Window by hitting button "Filter design"
6. Fill in your biquads with the desired filters or use the same filters as the master module. The high channel is channel 1 and the low channel is channel 2
7. Upload the filters into your module with the "Load filter to DSP" button (found at the top of the screen)
8. When the upload is completed you can power down the module and disconnect the USB cable

9. Master and Slave are now set up as a 2 channel stereo system

Final setup

When both modules are installed it is time to complete your setup.

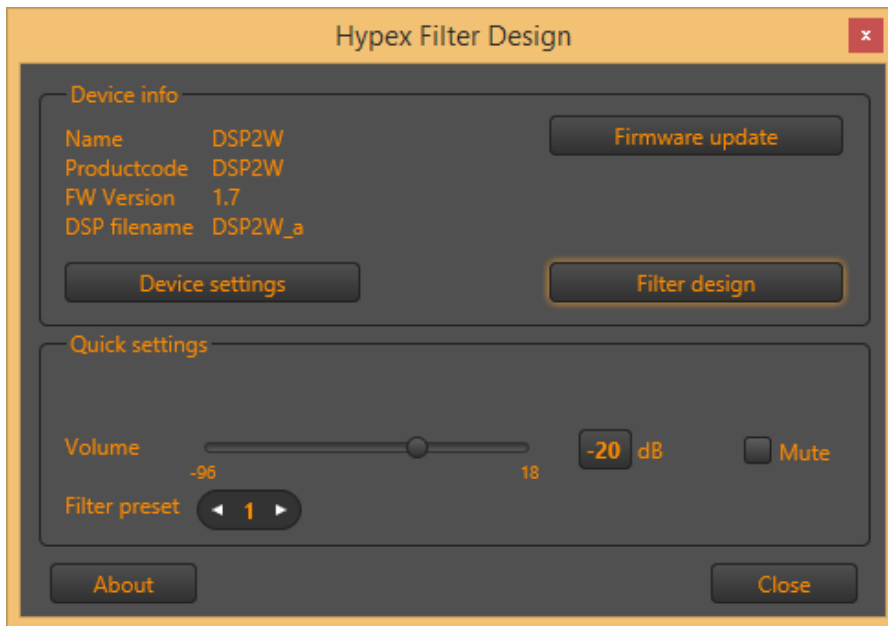
- First connect your speakers as explained in chapter Product overview
- Connect the trigger cable (mini jack) or SPDIF cable between both modules, depending on the used audio source
- Power up master and slave module
- Turn on the master by control panel or remote
- Lower you source volume, prevent loud start-up volume
- Connect your source to the AS2.100

Ready!

Now the system is fully setup and can be controlled by the control panel, remote control or by connecting the module to your pc. From now on you can update your filters at any time, just repeat the Master and Slave steps.

- * If your slave module is not turned on by the master try to power down the slave, turn the master off, wait 5 seconds, power up slave and turn the master back on*
- * When there is something that does not work try to repeat the installation*

Main screen



Picture 1: Main screen

Device info

When a device is connected via USB, it's name, product-code and firmware version will be displayed. Also, when DSP filter settings were uploaded, its project-filename is shown too.

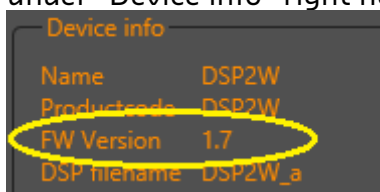
Supported modules

- AS2.100
- PSC2.400
- PSC2.700
- DSP2W
- DLCP
- Fusion amp

Firmware update

The latest Software and module Firmware can be found at our website. Please check if you have the latest versions if you want to keep the latest version. After a download only the executable file has to replace the old one. Firmware downloads and Hypex software can be found at www.diyclassd.com

Every module has the ability to update its firmware, when Hypex provides a new firmware version. The firmware version of the module is shown in the main screen under "Device info" right next to "FW version":



Picture 2: Firmware version of attached device

The bootloader used for the PSC2/DLCP and AS2 series are a bit different. The PSC2/DLCP has a different bootloader that requires an extra step for a firmware update. When you see “version 99,99” you are in the bootloader, from here you can update the firmware of the module. From this point updating is the same for both modules.

The firmware can be updated by USB, the same for master and slave module. Under option “download” you can find “Firmware update”. When this option is selected the user can select the new firmware file. This is a complete hex file provided by Hypex, no adapts can and may be made by the user!

After the file is selected the update begins.

DO NOT DISCONNECT THE MODULE AT THIS POINT!

When the progress bar is filled the update is completed. On a CRC error the update is automatically restarted, after three errors the update is aborted.

After restarting the module the new firmware is running.

**Note that the new firmware does not have any filters installed, so the filters must be reloaded with the correct values. This is however not always applicable to a FusionAmp update, because that module has it's filters stored in a different way and only need to be erased if the new firmware needs them to be reloaded.*

Quick settings

When a device is attached to your computer (and powered up), it is possible to change inputs and volume on the main screen. Use the volume slider or enter a volume-number in the edit box (and hit Enter to send the new volume to the device). Click “Mute” to mute or unmute all channels (the device is muted, when the box is checked)



Picture 3: Device is muted

Be careful when changing the volume up! When you selected the slider, it is also possible to change the volume up and down using the arrow keys of your keyboard. That's the safest way to increase the volume.

It is also possible to select a preset, if your device uses presets. Click on the arrows left or right to select the desired preset.



Picture 4: Selecting a preset

Filter design

This part describes the Filter Design part 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](#)

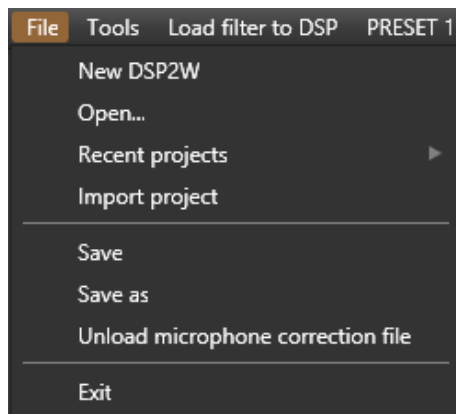
Filter design has two screens: one advanced screen (which represents the internal of the DSP and where biquads can be sorted) and a graphical screen, where filters are created using impulse data and adding biquads). When creating a new project, HFD always opens the graphical interface first. Use the button in the right-bottom 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 5: 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 you loaded a microphone correction file, 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.

Tools menu



Picture 6: 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.

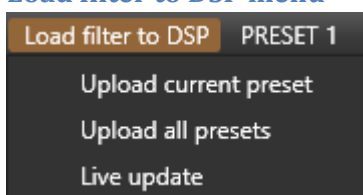
Also, you can hide or show the loaded microphone correction curve on the graphical filter design screen or edit device settings realtime when a device is connected to your computer.

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 filterdesign screen.



Picture 7: Set Asio preferences

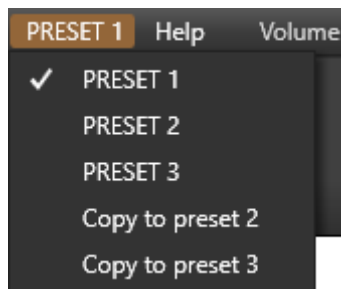
Load filter to DSP menu



Picture 8: Load filter to DSP menu

Filters can be uploaded to a device, preset by preset, or you can upload all presets automatically. 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. When a FusionAmp is connected, it is also possible to enable “Live update”. If enabled, a changed biquad will immediately be sent to the connected FusionAmp, so you can hear real time it’s effect on the audio of the selected channel and preset. A live updated biquad is not saved in memory however. So, to keep the settings of a live updated biquad (or more), you must upload the filters to the DSP by hitting “Upload current preset” or “Upload all presets”

PRESET menu



Picture 9: 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 10: Advanced screen overview

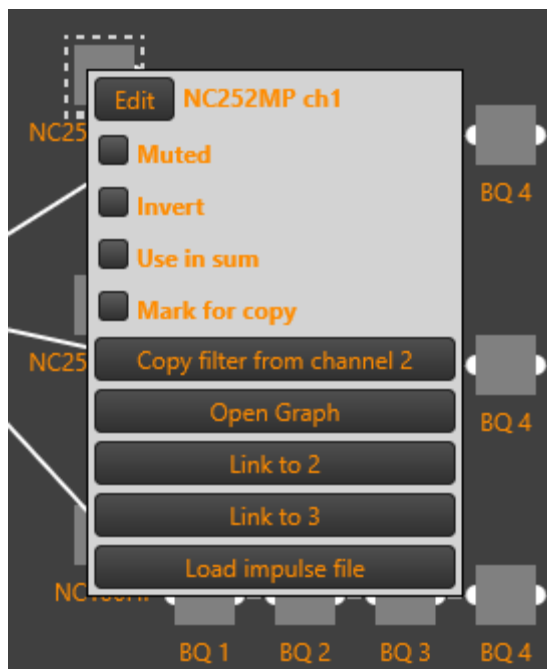
In the advanced screen you can:

- 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 masters 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 11: Right click on the channel box

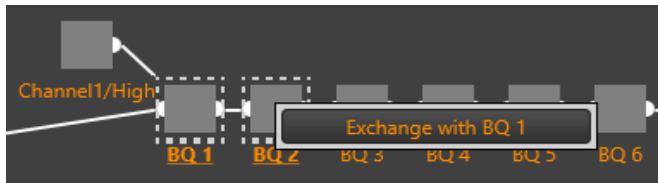
Here you can:

- 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

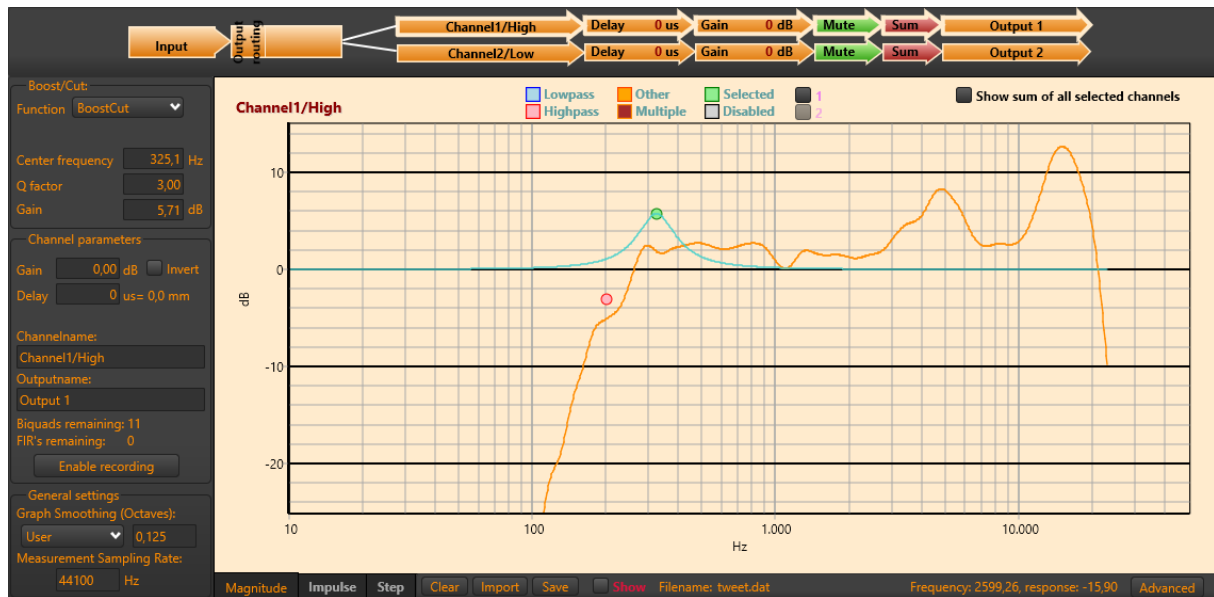
When you selected exactly two biquads on the same channel, you can exchange their parameters by right clicking on of the two and click on “Exchange with ...”



Picture 12: 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.

Graphical filter design screen



Picture 13: Graphical filter design screen

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.

The weapons of choice are 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 lest the cure be worse than the ailment. 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.

Biquad filter types

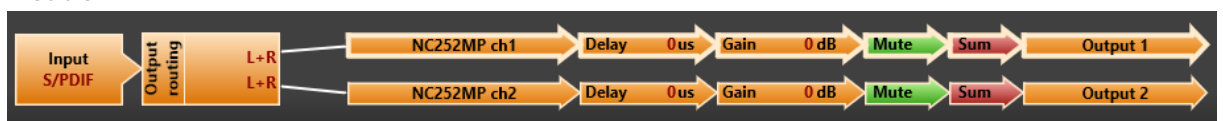
Biquad function	Parameters	Use
Unity	-	Section is not used
Lowpass 1	Cut-off frequency (always -3dB)	First order lowpass
Lowpass 2	Cut-off frequency (asymptotically) Q	Second order lowpass.
Highpass 1	Cut-off frequency (always -3dB)	First order highpass
Highpass 2	Cut-off frequency (asymptotically) Q	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.

Table 1: Biquad filter types

The graphical filter design screen has four main parts:

- Header



- Left pane with filter object parameters, channel parameters, general settings and recording options

Boost/Cut:

Function BoostCut

Center frequency 4810,5 Hz

Q factor 2,02

Gain -7,89 dB

Channel parameters

Gain 0,00 dB Invert

Delay 0 us = 0,0 mm

Channelname: Channel1/High

Outputname: Output 1

Biquads remaining: 10
FIR's remaining: 0

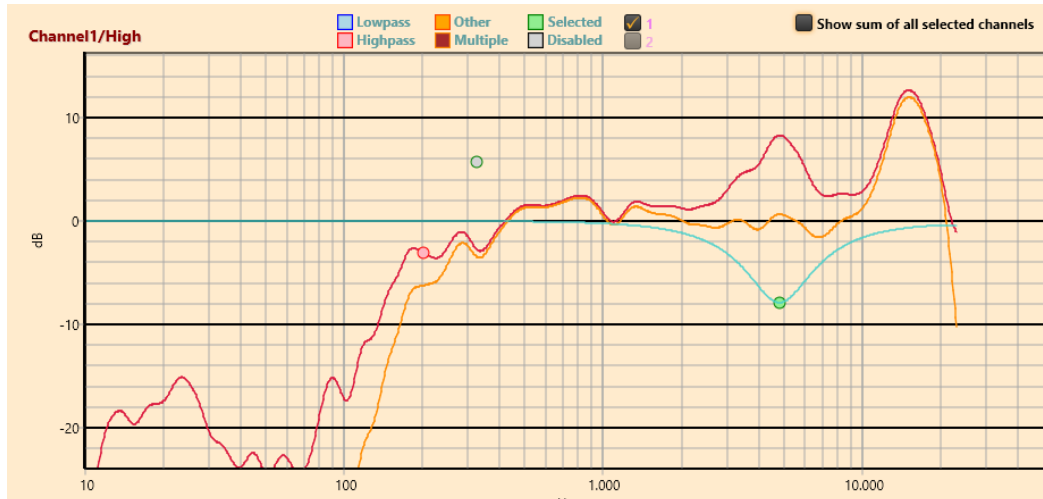
Enable recording

General settings

Graph Smoothing (Octaves): User 0,125

Measurement Sampling Rate: 44100 Hz

- Right pane with graphical representation of the selected channel



- Bottom pane with tabs for Magnitude, Impulse and Step graph, impulse load/clear/save buttons and cursor position.



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.

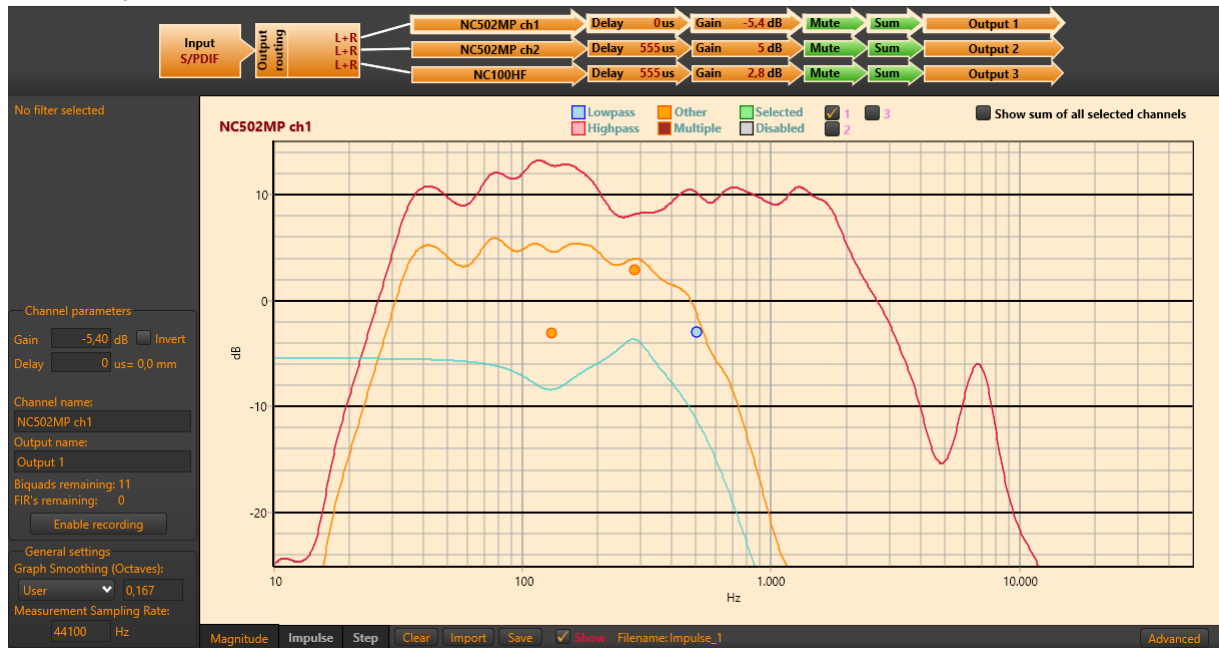
Graph right pane (filter graph)

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 14: 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, highpass 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 gray.

FusionAmp option: if you enabled “Live update” (found at the top of the window, under menu-item “Load filter to DSP”), any biquad that is changed on the screen, is also real time updated in the DSP. But remember: biquads changed in this way are not stored in the FusionAmps memory. To do so, upload the current or all presets to the DSP.

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.

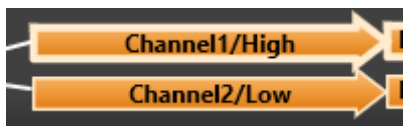
When channels are added to the sum, the sum-line can be displayed by checking “Show sum of all selected channels” at the right top of the graph. The sum is drawn black.

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.

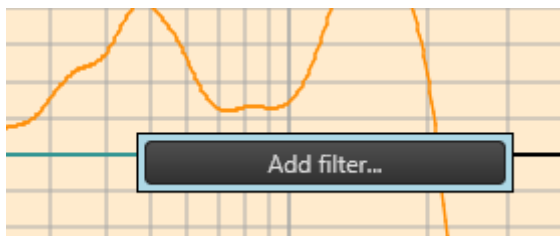
Adding a filter object

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



Picture 15: 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:



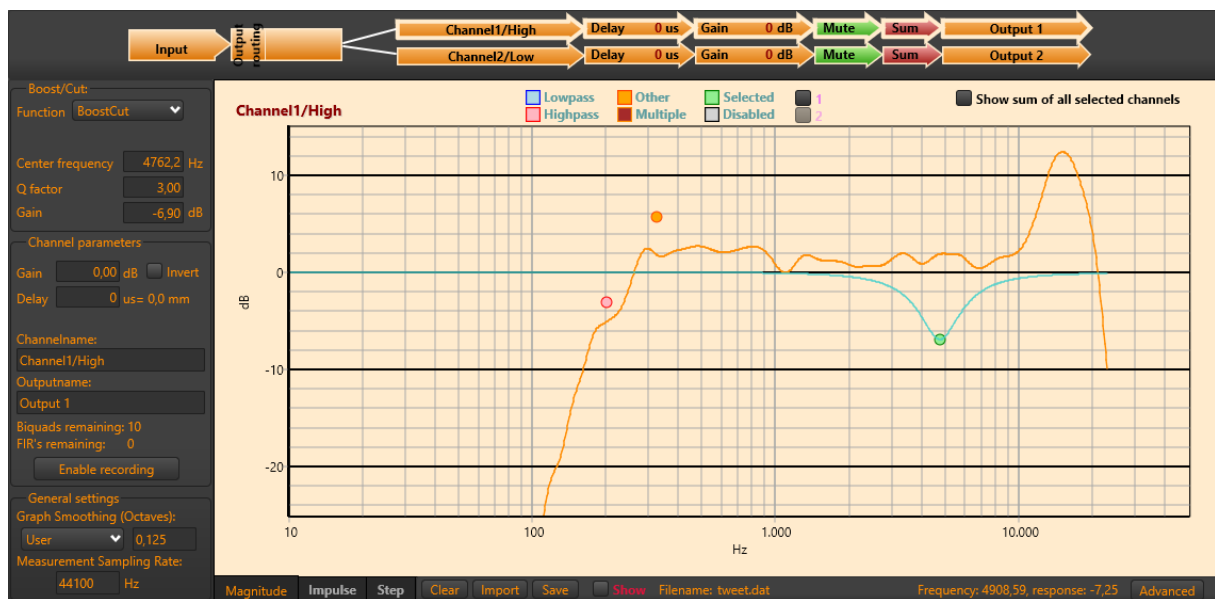
Picture 16: Add a filter object

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



Picture 17: 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”:



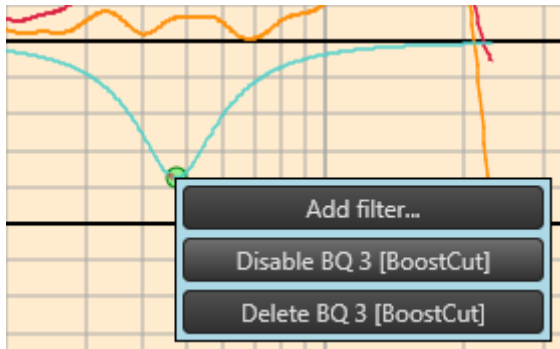
Picture 18: Filter object curve

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.

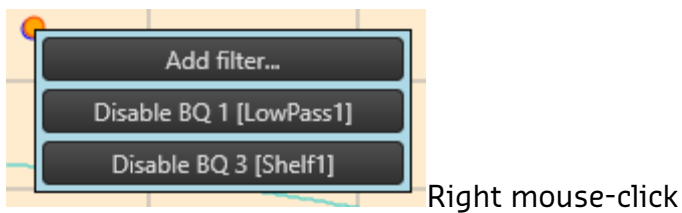
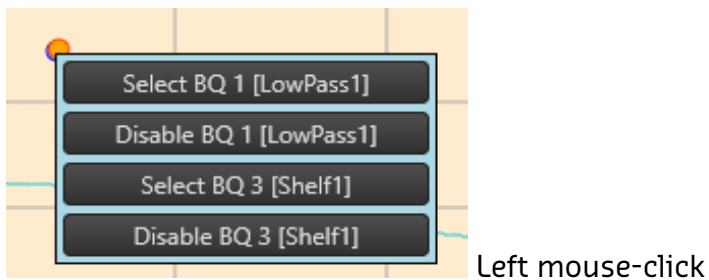
Disable or delete filter objects

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



Picture 19: 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.

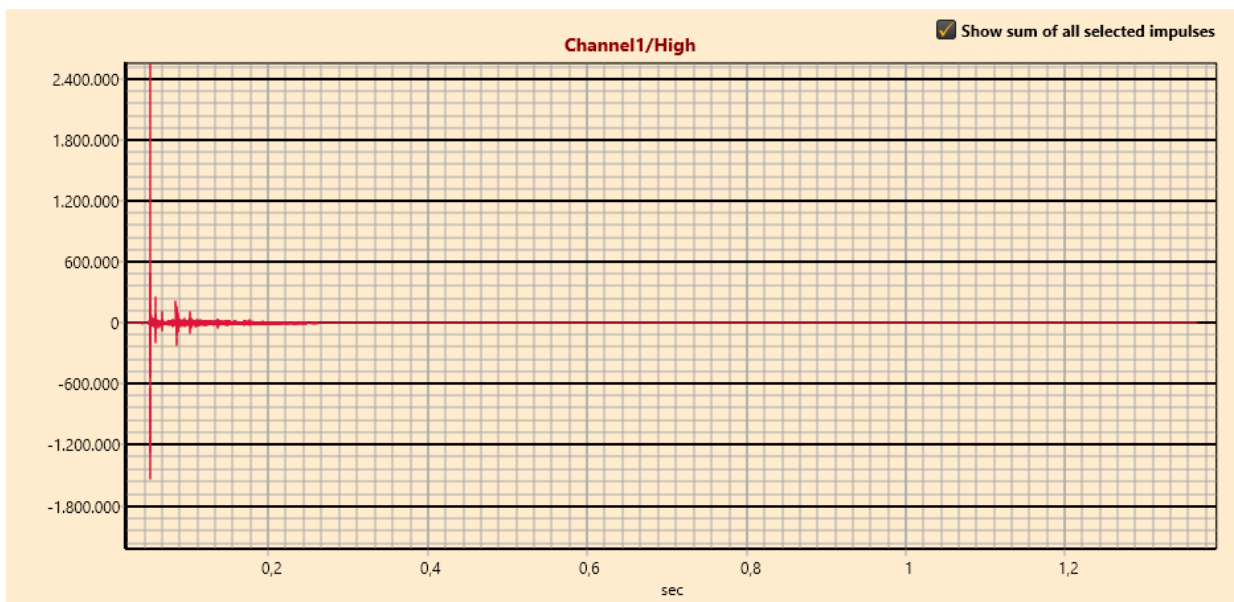


NOTE: a disabled filter object will not be uploaded to the DSP. You can use the disable function to temporary remove the filter object from your channel to hear its effect

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.

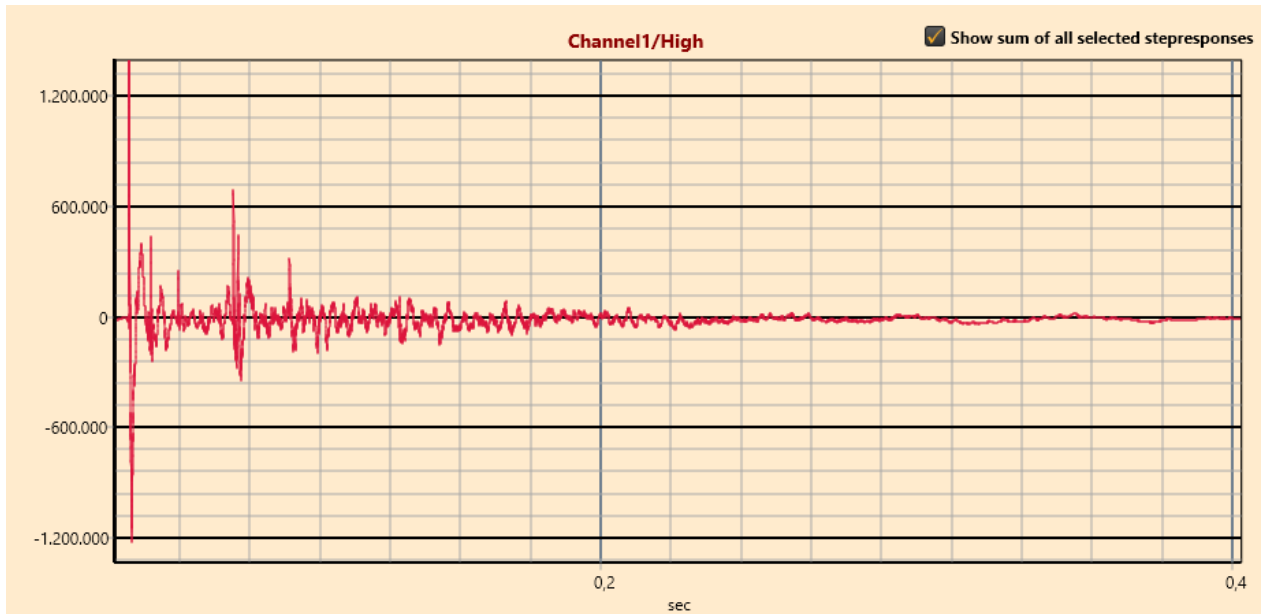
To zoom-out, just press the middle-mouse (or wheel) button and draw a rectangular box from right to left. The size does not matter. You can zoom-in/zoom-out multiple times.



Picture 20: Impulse response graph

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 21: 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.

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:

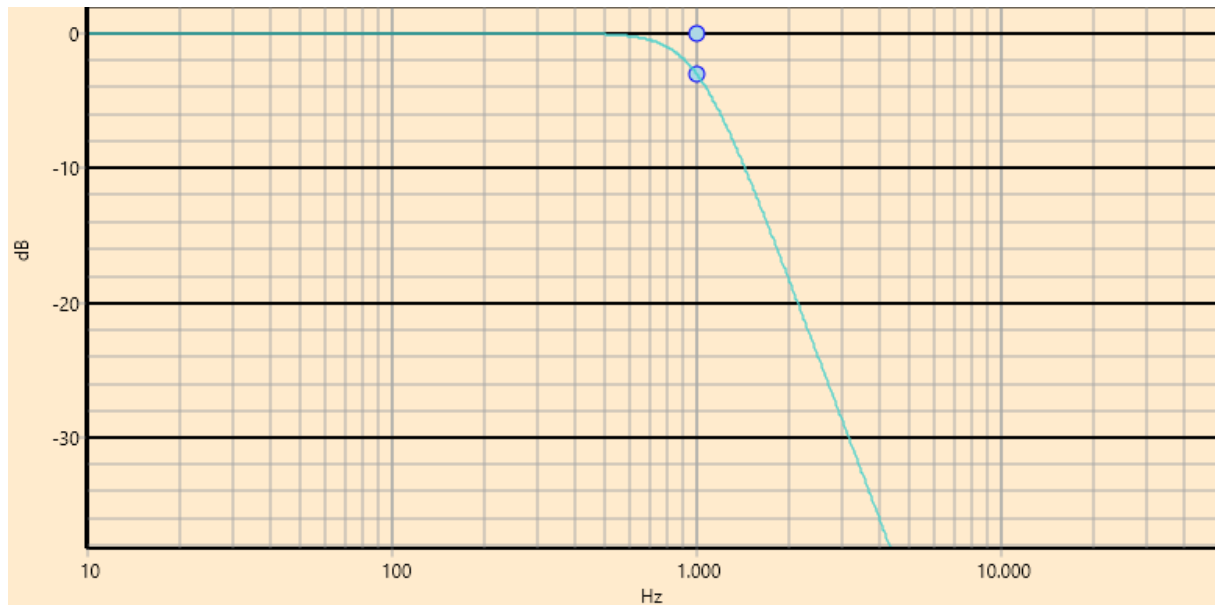
Order	Q first biquad	Q second biquad	Q third biquad
2	0.71		
3	1 st order filter, no Q	1.0	
4	0.54	1.31	
5	1 st order filter, no Q	0.62	1.62
6	0.52	0.71	1.93

Table 2: Butterworth coefficients

Third order lowpass Butterworth example ($F_c=1000\text{Hz}$)

First order lowpass filter:
Function
Cut-off frequency

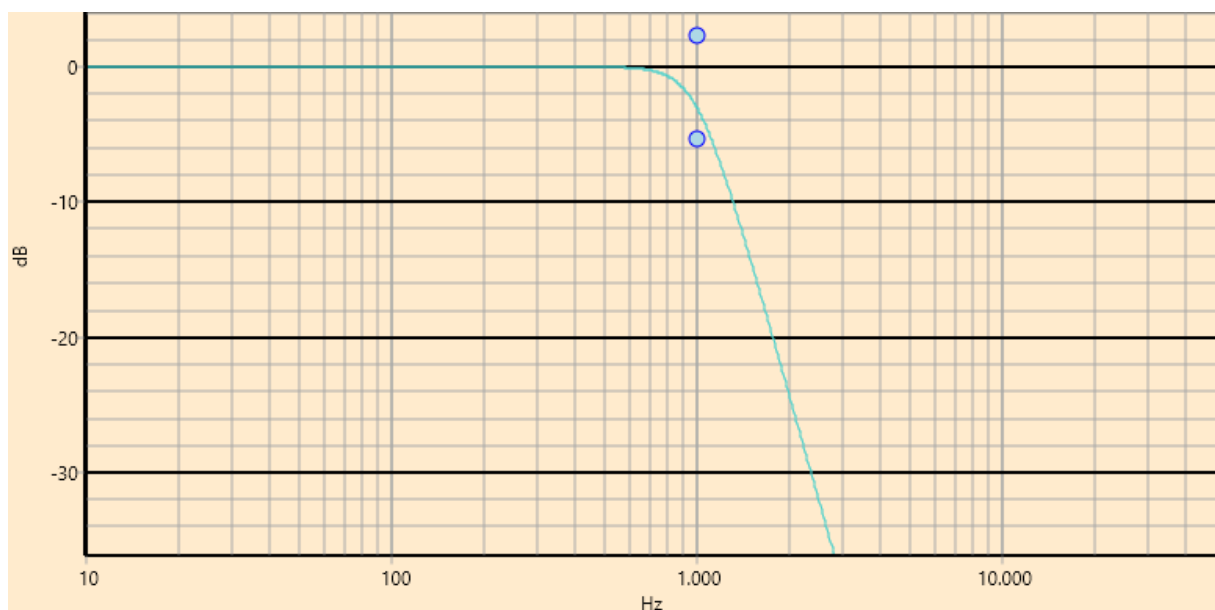
Second order lowpass filter:
Function
Cut-off frequency
Q



Fourth order lowpass Butterworth example ($F_c=1000\text{Hz}$)

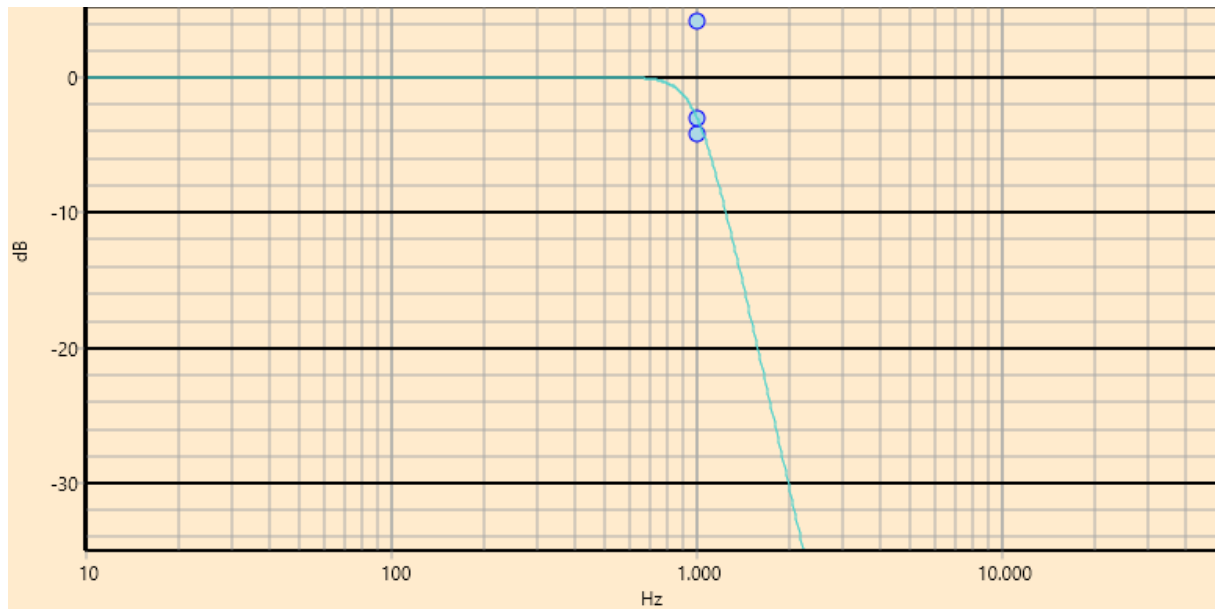
Second order lowpass filter:
Function
Cut-off frequency
Q

Second order lowpass filter:
Function
Cut-off frequency
Q



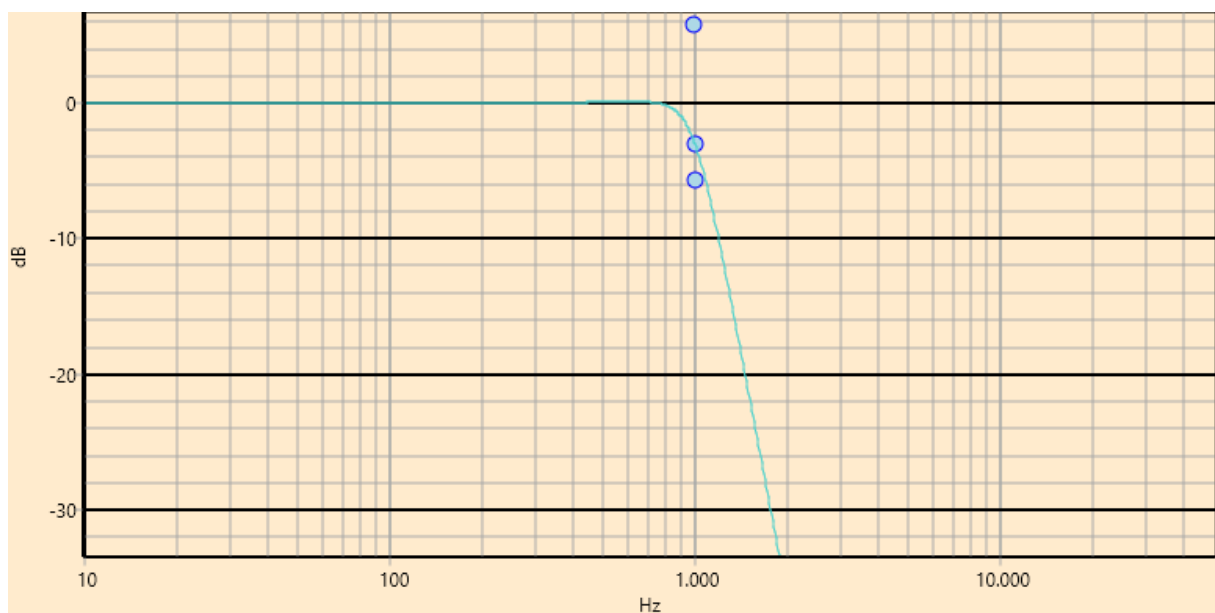
Fifth order lowpass Butterworth example ($F_c=1000\text{Hz}$)

First order lowpass filter: Function <input type="text" value="LowPass1"/> Cut-off frequency <input type="text" value="1000,0"/> Hz	+	Second order lowpass filter: Function <input type="text" value="LowPass2"/> Cut-off frequency <input type="text" value="1000,0"/> Hz Q <input type="text" value="0,62"/>	+	Second order lowpass filter: Function <input type="text" value="LowPass2"/> Cut-off frequency <input type="text" value="1000,0"/> Hz Q <input type="text" value="1,62"/>
---	---	---	---	---



Sixth order lowpass Butterworth example ($F_c=1000\text{Hz}$)

Second order lowpass filter: Function <input type="text" value="LowPass2"/> Cut-off frequency <input type="text" value="1000,0"/> Hz Q <input type="text" value="0,52"/>	+	Second order lowpass filter: Function <input type="text" value="LowPass2"/> Cut-off frequency <input type="text" value="1000,0"/> Hz Q <input type="text" value="0,71"/>	+	Second order lowpass filter: Function <input type="text" value="LowPass2"/> Cut-off frequency <input type="text" value="1000,0"/> Hz Q <input type="text" value="1,93"/>
---	---	---	---	---



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).

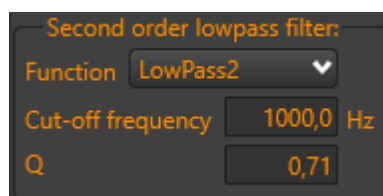
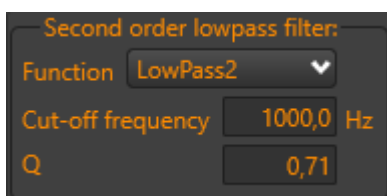
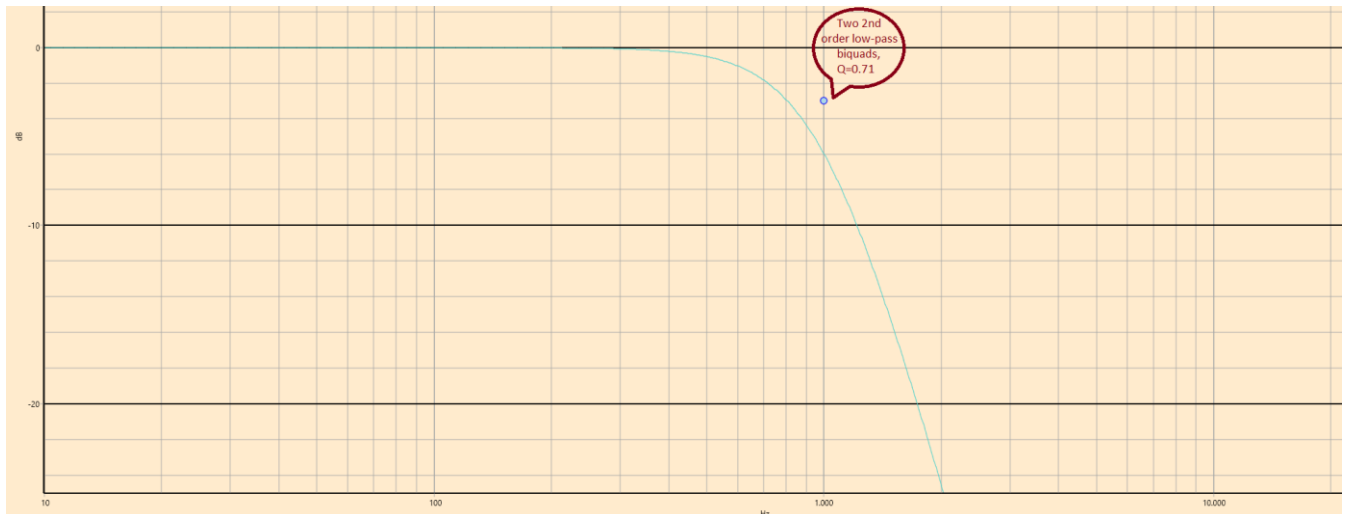
A second order low-pass or high-pass LR (short for Linkwitz-Riley) filter has a Q of 0.71 on its biquad. A 4th order LR filter (LR4) is a cascade of two 2nd order Butterworth filters (see [Butterworth](#)). An 8th order LR filter (LR8) is a cascade of two 4th order Butterworth filters, etc.

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.

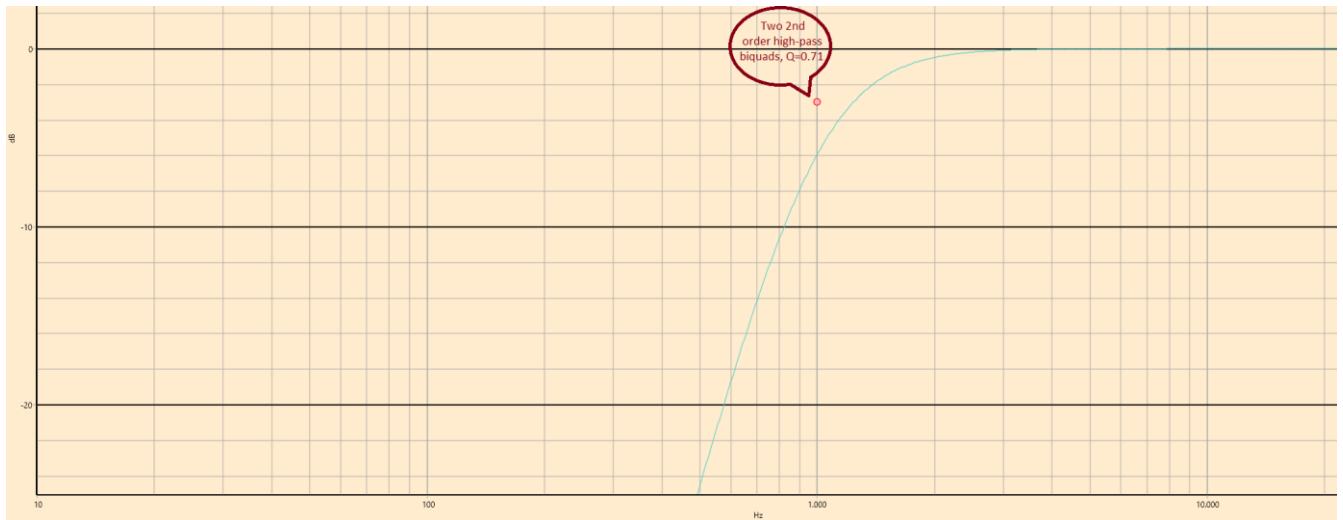
Example: Crossover filter, LR4, cut-off frequency 1 kHz.

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):



For channel 2 (high frequency driver):



Second order highpass filter:

Function **HighPass2** ▾

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

Q **0,71**

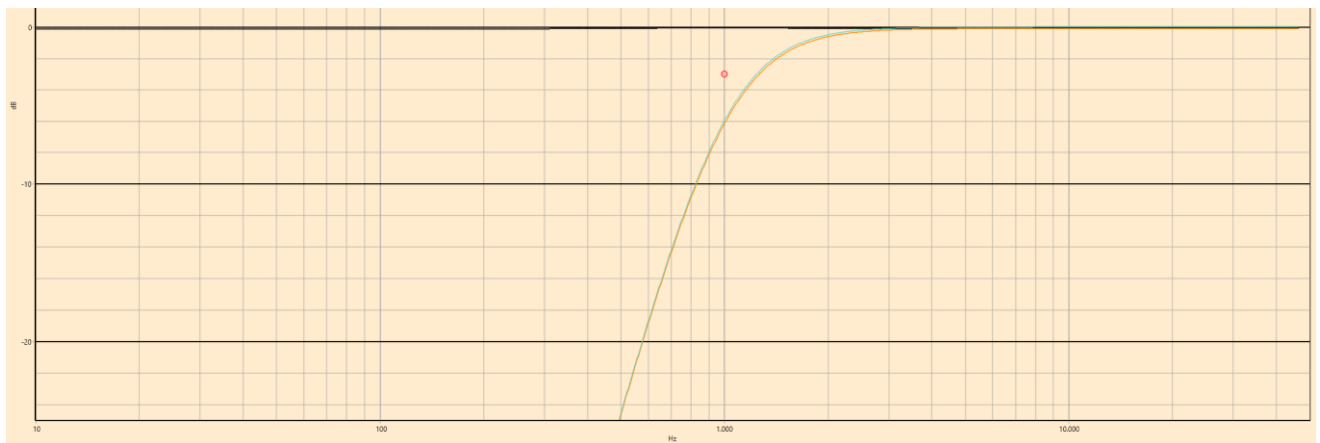
Second order highpass filter:

Function **HighPass2** ▾

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

Q **0,71**

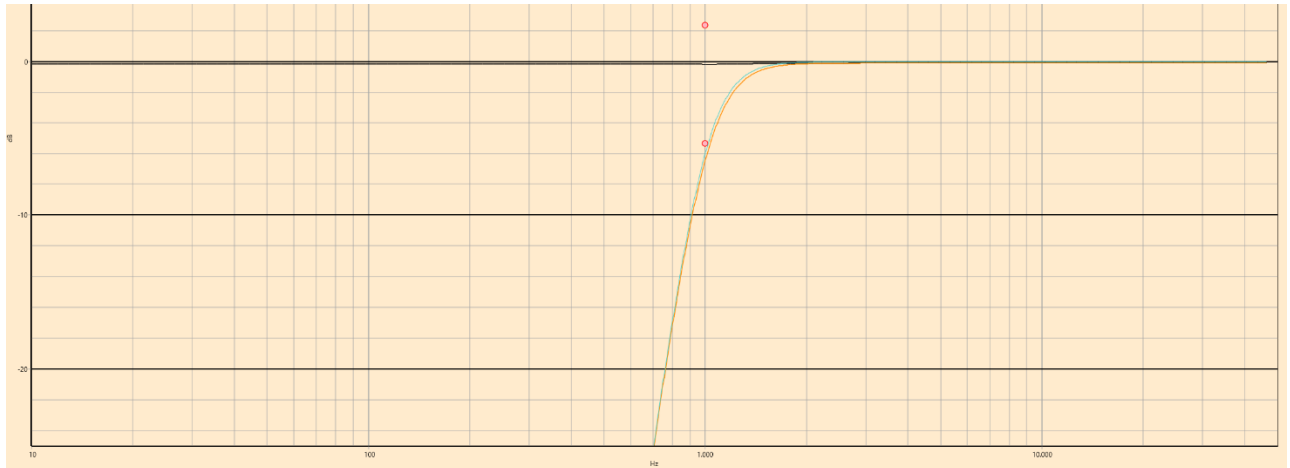
Looking at the sum-result of both channels (you can simulate this in HFD by loading a flat impulse file on both channels), you can observe that the result of the Linkwitz-Riley filter is flat over the total frequency range (black line just below 0dB):



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

Table 3: Linkwitz-Riley coefficients

Plot of a 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	1 st order filter, no Q	1.32x	0.69	1.45x		
4	0.81	1.60x	0.52	1.43x		
5	1 st 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 4: Besselfilter 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	$1.6 \times 1000 = 1600\text{Hz}$	0.81
2	$1.43 \times 1000 = 1430\text{Hz}$	0.52

Table 5: Example 4th order Bessel filter $F_c=1000\text{Hz}$

In case of a sixth 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 6: Example 6th order Bessel filter $F_c=1000\text{Hz}$

Fourth order lowpass Bessel example ($F_c=1000\text{Hz}$)

Second order lowpass filter:

Function LowPass2 ▾

Cut-off frequency 1600,0 Hz

Q 0,81

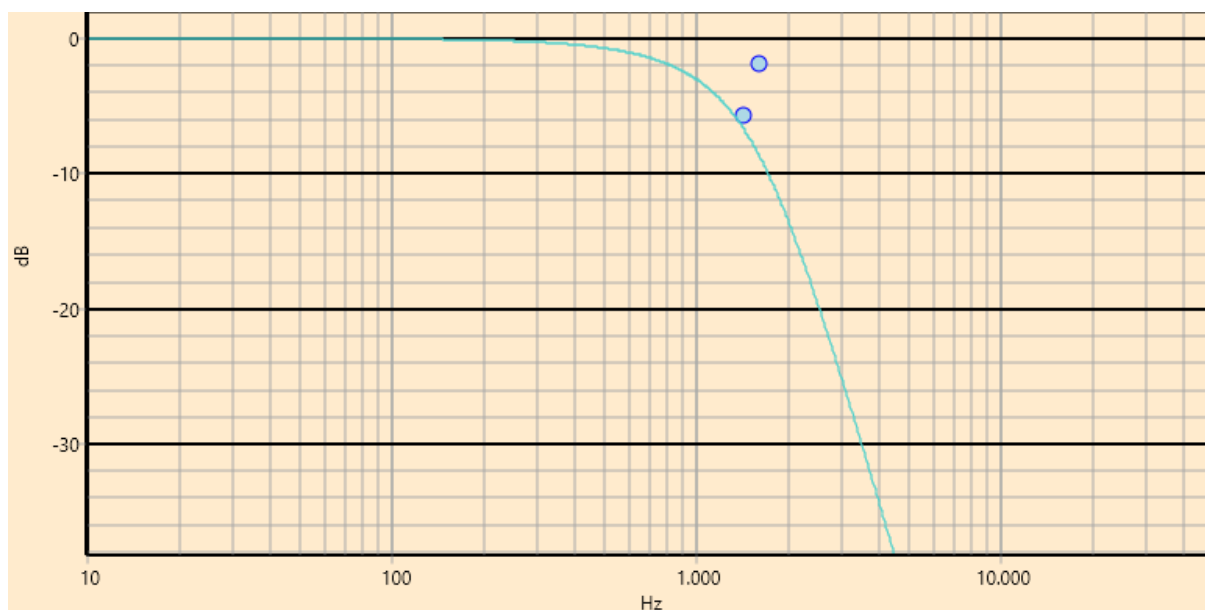
+

Second order lowpass filter:

Function LowPass2 ▾

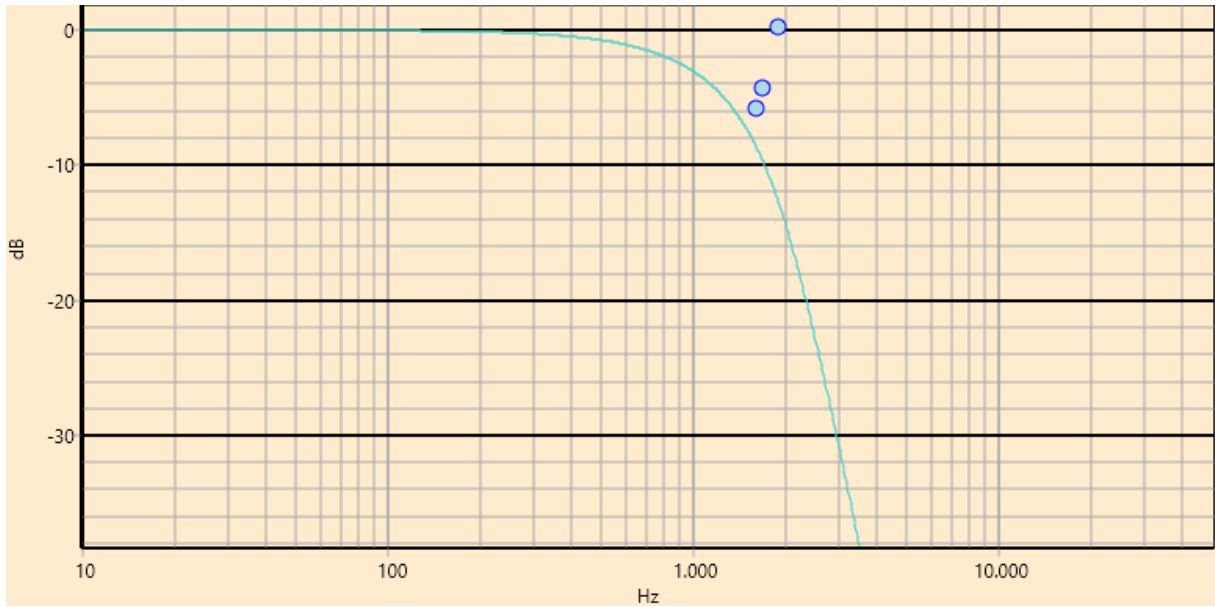
Cut-off frequency 1430,0 Hz

Q 0,52



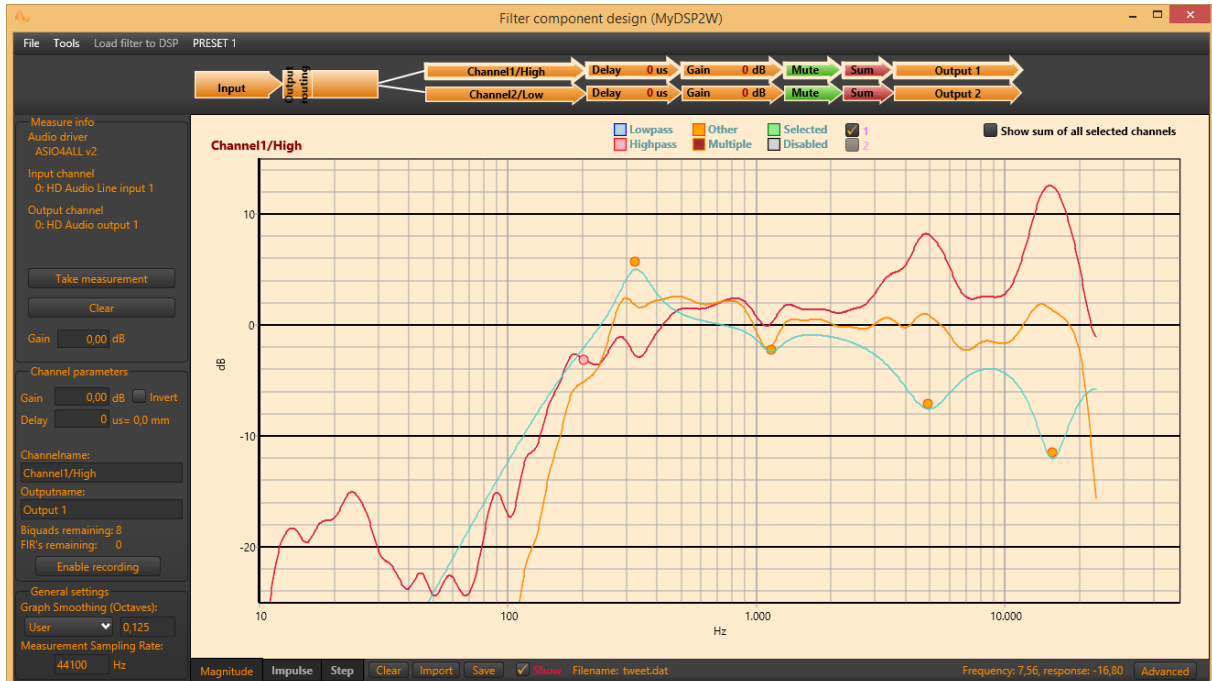
Sixth order lowpass Bessel example (Fc=1000Hz)

Second order lowpass filter: Function <input type="text" value="LowPass2"/> Cut-off frequency <input type="text" value="1900,0"/> Hz Q <input type="text" value="1,02"/>	+	Second order lowpass filter: Function <input type="text" value="LowPass2"/> Cut-off frequency <input type="text" value="1690,0"/> Hz Q <input type="text" value="0,61"/>	+	Second order lowpass filter: Function <input type="text" value="LowPass2"/> Cut-off frequency <input type="text" value="1600,0"/> Hz Q <input type="text" value="0,51"/>
---	---	---	---	---



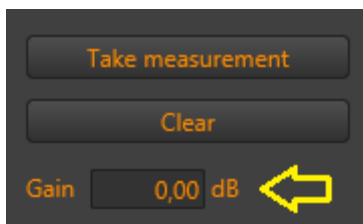
Recording

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 22: 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 23: Change sweep output gain

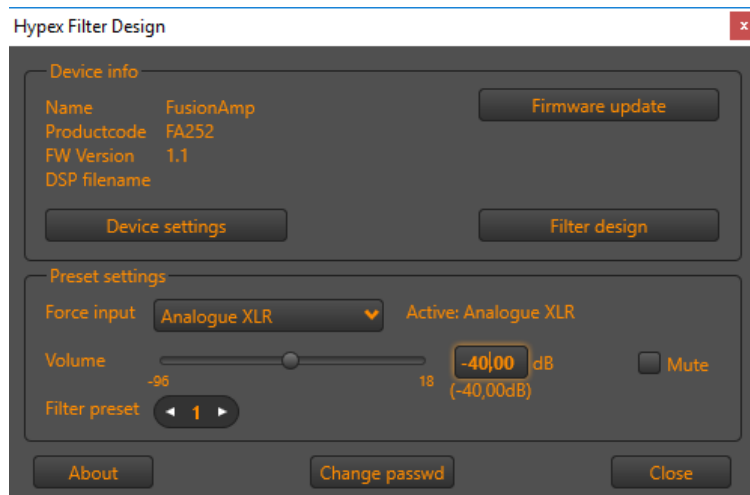
Hit the “Take measurement” button to start a measurement. HFD will output a sweep signal and record incoming audio. Afterwards, the impulse and step response are shown in the graphs. From here you can zoom in on the impulse response and truncate to remove echo’s and start editing a proper filter for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

Fusion amp device settings

Main screen

When a FusionAmp is connected to HFD, you can change its volume, source and preset on the main screen:



Picture 24: Main screen

Force input

The sources you can select depend on the type of FusionAmp that is connected to HFD. The sources “Auto detect”, “Analogue XLR” and “Analogue RCA” are always available. If your FusionAmp has high-level capability, the source “SUB” will be added to the list and when your FusionAmp is equipped with a SRC digital IO board, the sources “AES”, “SPDIF”, “Optical” and “Bluetooth” are added. All speak for themselves, but “Auto detect” needs a bit more explanation, see [“Auto detect”](#)

Volume

You can change the volume in hfd in two ways: with the slider and by entering a numeric value. **Be very careful using the slider though, it is very easy to set the volume level to a high level**, risking damaging your ears or your hardware. Best practise is to change the slider position with the arrow keys LEFT and RIGHT on your keyboard, or change the volume with the numeric input. The volume text beneath the numeric volume input (in picture 24 labelled “(-40.00dB)”) indicates the actual volume that is calculated by the system by adding all available offsets ([preset volume offset](#) and [high-level volume knob](#)) to the master volume.

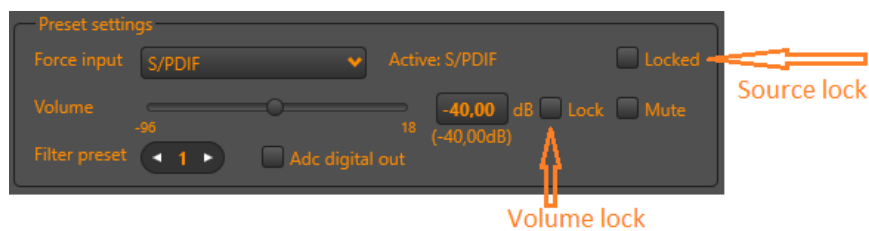
Auto detect

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) Bluetooth
- 5) Analogue XLR
- 6) Analogue RCA

Extra slave options

In the case, you instructed the FusionAmp to behave like a slave (see [Options](#) in Device settings), two extra settings come available on the main screen:



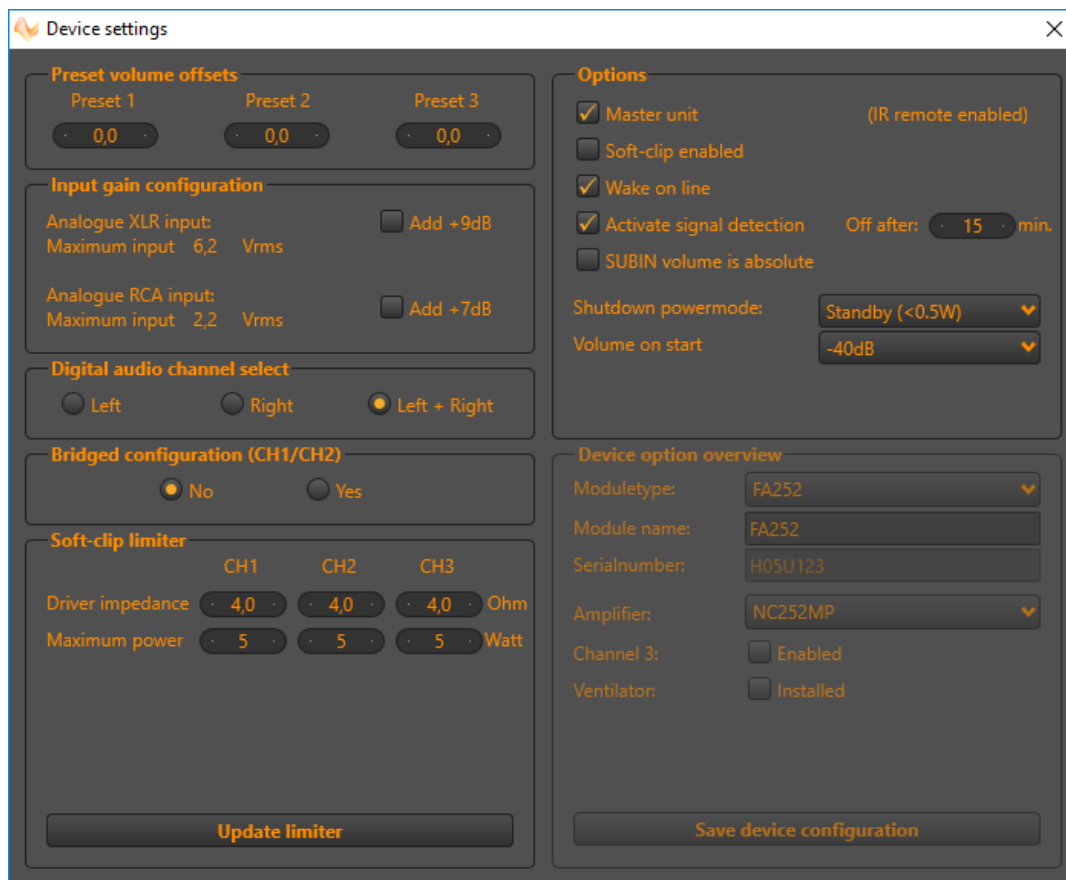
A slave can lock its source, so it will not change when the masters source is changed. This option will show itself to be very handy if, for example, your slave is connected to the master with a SPDIF cable and you provide audio through that cable. The master-source can be Optical, SPDIF or AES, whereas the slave source should always remain SPDIF. Another example is, when your slave is equipped with a high-level input board. In that case, you probably want that SUB is always selected as the source, no matter what source is selected on the master. To lock the source, check the control named "Locked".

The second 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 slaves source is connected to an output which has its own level control (high level (SUB) input for example).

Both extra options are added to the preset settings, so every preset can have its own source locked and volume locked.

Device settings

There are a few more parameters with which you can personalize your FusionAmp. To do so, hit the “Device settings” button on the main screen. A new window will pop up:



Notice that you cannot change all parameters. The ones in the “Device option overview” group for example, are only for Hypex to change. We will explain the parameters, group by group.

Preset volume offsets

The actual volume which is calculated by the DSP, is composed of several items. The master volume is set on the main screen with the slider or the numeric box. For every preset, you can add an offset to that volume between -12dB and +12dB.

If you have a FusionAmp with a high-level input, an extra offset is added with the volume knob, which also has a range of -12dB to +12dB

Digital audio channel select

When your FusionAmp is equipped with the digital IO board, you must instruct the DSP which audio channel in the digital signal it has to use. By default, it will add left and right, but you can change the setting to “Left” or “Right”, so the FusionAmp will output only the left or right signal.

Bridged configuration (CH1/CH2)

FusionAmps with at least two output channels have the possibility to bridge the two outputs to one. 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 not be in effect immediately, but only when the DSP is restarted. Please use the remote control or power switch to shut down the FusionAmp and switch it back on.

Soft-clip limiter

The DSP in the FusionAmp has a software limiter which can be used to limit the output power. To set the limiter to suit your needs, you must provide a maximum output power as well as the impedance of the connected drivers per channel. Power is set in units of 5Wrms. Default, the output power is set to zero, so before you enable the soft-clip limiter (under "Options"), please set the proper parameters and hit button "Update limiter" to save the settings in the FusionAmp.

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 have to 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, choose a source or preset with a RC5 remote control. Under "Options" check "Master unit" if the connected device is a master, uncheck it to force it to be a slave.

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

If you want the FusionAmp to start when it is connected to the mains, set this option. If it is unchecked and the FusionAmp is connected to main power, it will initialize and enter power down mode.

Activate signal detection

The FusionAmp has two power-down modes: "Standby" and "Low power" (See "Shutdown 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, the FusionAmp shuts down after 15 minutes of lack of audio on the selected source.

SUBIN volume is absolute

By default, the volume control knob on the high-level input board, gives you an offset of -12dB to +12dB on the master volume. If you want to control the master volume with the control knob, check this option. The range of the knob will then switch to -96dB to +18dB. **But use with care!** If checked and the knob is set to +18dB, the FusionAmp may produce maximum power and damage your drivers.

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 inputs when signal detection is enabled, or with a remote control, if your device is equipped with a remote control set. Most of the FusionAmp hardware is put in suspend mode to save power.

When the device entered "Low-power" mode, the SRC is not switched off, but is used to scan the digital inputs. When audio is detected on one of the digital sources, the system will start. In low power mode, the FusionAmp will consume less than 1.5W.

Volume on start

A nice feature of the FusionAmp is the ability of 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 choose from -3dB, -6dB, -10dB, -20dB, -30dB, -40dB, -50dB, -60dB, -70dB or leave it unconfigured. When not configured, the FusionAmp will always start with the last volume you selected.

FAQ

This is a list of Frequently Asked Questions about the Hypex Filter Design.

Q: My DSP module is not recognised by my computer, what can I do?

A: First check if your computer sees the device, if the computer does not show any reaction when you plug in USB then try another USB cable, it may be defected. If this still not giving any response the USB receiver may be broken, please contact us for more information.

If the computer does recognises some USB but does not detect the right device or is unable to install the USB device then please try to disconnect the USB for a about 30 seconds and try again. Or another computer if you have the ability. When this is not successful then try to setup the AS2.100 module as a Slave by removing the control panel and reconnect the module to the computer. When this solves the problem please update the firmware of the module.

Also be sure that the mCHID.dll file is in the same folder as the Hypex Filter Design software. If it is still not working then please contact us for more information.

Q: I cannot get the DSP module connected to my Windows7 computer, what is the problem?

A: Because of differences in newer systems the USB is not always handled properly. This is fixed in AS2.100 firmware V2.7.

Q: What is a "Biquad"?

A: A biquad is a filter section with 2 poles and 2 zeros. All of the biquads are in series so all kinds of filters can be made.

Q: What about the phase of the filters?

A: The biquad filters are all minimum phase, which means that phase is directly linked to magnitude response in exactly the same way as an analogue filter. Since individual speaker drivers are minimum phase too, EQ'ing them flat also equalises the phase response.

Q: I want to import a speaker measurement but what kind of file is expected and with what program can I use?

A: The file format is quite simple: a text file containing the impulse response. Each line contains one number, which is one sample of the impulse response. If your MLS tool supports exporting of the impulse response, you can usually tailor that file to work with the DSP tool.

Q: I don't have any measurements of my speakers, how can I see the sum of my filters?

A: When you load the Flat.dat file you have a flat measurement. This enables you to sum you filters.

Q: The AS2.100 responds on my TV remote control, how can I disable this?
A: The "OneForAll" remote we use has the same commands as some Philips TV's. In version 2.7 and up you are able to select a different remote, the Hypex remote, which uses different commands and therefore no longer responds on the TV remote.

Q: I want to use a universal remote for the AS2.100 what kind of Infrared format is used?

A: We use the RC5 Philips TV codes for controlling the AS2.100. Here's a list for the used commands.

OneForAll remote:

Device = 0 (TV1)
On/off = 12
On = 10
Off = 11
Vol up = 16
Vol down = 17
Mute = 13
Channel up = 32
Channel down = 33

Hypex remote:

Device = 16 (Pre-amp)
On/off = 50
On = 10
Off = 11
Vol up = 51
Vol down = 52
Mute = 53
Channel up = 54
Channel down = 55

Q: I forgot to save the filter settings I loaded into my module, is there a way to read them back to the filter design?

A: Since HypexFilterDesign V2.7 the module saves the name of the filter settings. By this you can read back what files was loaded last time. This information can be found under Help>About

Q: When I connect my AS2 the left part in the Control panel is cleared.

A: When you have an analogue AS2.100 this side is removed because these functions are not there.

Q: I would like to make filters without the need of connecting a device, how can I do this?

A: In the filter design window you can choose File>New>product. This will start a empty setup.

Q: What about the Subout of the AS2.100?

A: The Sub-out is a line output to drive an active (external) subwoofer. This signal can also be filtered by the AS2.100 so only an amplifier is needed for the active

subwoofer and you have your three-channel active system.

Q: I want to have the AS2.100 auto-on when the power is switched on, is this possible?

A: Since HypexFilterDesign V2.7 you can set "Automatic on during power-up". This setting can be found under Tools>options

Q: The Auto input detect function is not working.

A: The input detection scans for any valid signals on the digital inputs. Sometimes a CD-player provides a digital signal even when it is not running, this may disturb the auto input selection. And if the source is in DTS format the module will mute because DTS is not supported, only 2 channel-SPDIF.

Q: The module is not responding after firmware update.

A: After a firmware update all filter settings are cleared, so they have to be uploaded again to get the output back on.

Q: When I use a digital link between the AS2.100 master and slave, do I still have to use the jack trigger cable?

A: The SPDIF cable between the master and slave also carries the communication signals between the master and slave, like volume and module setup. But the Jack cable also sends the right analogue audio channel to the slave, together with the communication signals. If the right analogue audio is provided to the Slave directly and the SPDIF cable links between the master and slave, the jack trigger cable is no longer needed.

Q: I have a module that broke down. Do I have warranty?

A: You have a standard warranty of two years on a Hypex product. We consider ourselves to be very flexible when it comes to this subject. However, when we find extensive modifications made by yourself which prevents us from repairing the module or is even the cause of the failure itself, we may charge you for the repair or even decide not to repair the module at all.

Q: My volume settings are not stored after I power down the module, why is this?

A: The AS2 stores all settings when it is switched to standby. The PSC stores these settings automatically after a change. When the values are not stored the module is switched off the wrong way.

Q: Why is the processor sample rate in the settings window standard set to a weird value like 46875Hz?

A: This is because of the clock frequency we use on the board. The sample rate stays on the board and therefore will do no harm.

Q: When I select my digital source there is a small tick and then the module goes in mute.

A: The digital source has a non-SPDIF format like DTS. This signal is not supported by the module and therefore it is not possible to use this as a source.

Q: How do I set the AS2.100 to slave?

- A:** When there is no button panel connected the AS2.100 will act as slave. This means that it receives commands from the master to change volume and other settings.
- Q:** My AS2.100 slave module does not always follow the master correctly, what can I do?
- A:** In firmware V2.8 some things in the master-slave communication has been changed, try updating the firmware to V2.8 or later.