DSP Special Vol.1 – the Sound Tuning Magazine from Audiotec Fischer including operation and adjustment manual for digital sound processors

AUDIOTEC FISCHER

Sound Tuning Magazine

For DSP PC-Tool V2

DSP Special

Vol.1

Fantastic sound improvement thanks to DSP technology

- Multiband-equalizer for a natural sound experience
- Time alignment for precise sound staging
- Wide crossover features for perfect frequency range allocation

These products are equipped with DSP power:

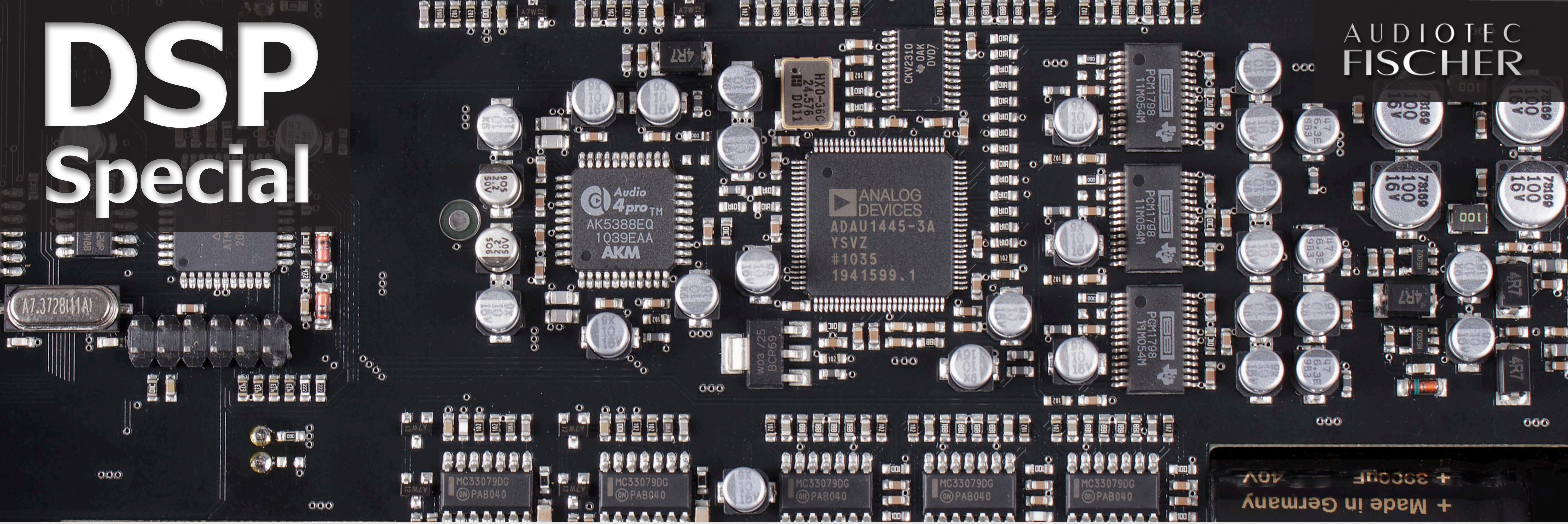
GERMAN CAR AUDIO BRAX

HELIX GERMAN CAR HIFI

MATCH BY AUDIOTEC FISCHER

DSP Special

AUDIOTEC FISCHER



Content

A.1. Installing the ATF DSP PC-Tool software

The installation of the ATF DSP PC-Tool software is a basic requirement before programming and adjusting our DSP devices

In order to use the ATF DSP PC-Tool software you first have to install it on your PC according to the following instructions.

Important: Ensure that the device or USB interface is not yet connected to your computer. Only connect the device or USB interface once the software has been successfully installed on the computer.

Download the installer file “SETUP_ATF-DSP-PC-Tool_Installer.exe ” from the Audiotec Fischer website (www.audiotec-fischer.de). The file can be found in the “Download” section. Save this file on the hard disk of your computer. Start the installer file with a double click.

Depending on your operating system you will see the security advice (picture 1). Double click on the „Execute“ button to confirm.

If you don't have „administrator rights“ for your PC, it might be possible that another security hint will appear which you have to confirm as well.

Then you should see a window, where you can select the language for the installation process (picture 2). This selection has no influence on the language of the PC-Tool software itself (English language only). Now the setup assistant will start – to continue please double click the respective button (picture 3).

Make sure that you carefully read the information in the following window (picture 4). After that you have to confirm this. Now select the location on your hard drive for the installation of the program. The installer will make a proposal, but you can change the location as you like (picture 5).

Now choose a name for the shortcut in your start menu. Here again the installer will make a proposal which you can change. Otherwise simply confirm to continue (picture 6). If you like to have a shortcut icon on your desktop for easiest access of the program, then checkbox the appropriate position (picture 7).

Click on „Install“ in the following window and the installation process of the DSP PC-Tool software will start. This installation includes the required USB drivers (picture 8).

Important: Carefully read the security advices in the following window (picture 9) before you continue. Now the installation is finished and the DSP PC-Tool software will be ready to use. If you want to run the software lately, then remove the checkmark in the checkbox (picture 10). Click „finish“ to complete the installation procedure.

The software „ATF DSP PC-Tool“ including the required USB drivers can be downloaded as a complete installation file from the Audiotec Fischer website.

A.2. System requirements

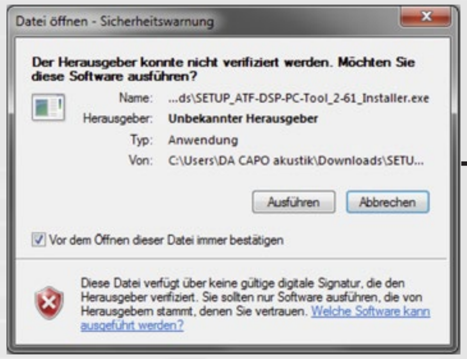
PC system requirements:
- At least 1 GHz processor
- At least 1 GB RAM main memory
- At least 25 MB free hard disk space
- 1 free USB port
- Screen resolution of at least 1024 x 768 pixels or better

The software has been tested in combination with the following operating systems: Windows XP (32 bit), Windows Vista, Windows 7 and Windows 8 (32 & 64-bit each).

PP50 DSP system requirements:
- You require the HELIX Optical & USB Interface (Part No. H424493).
- The relevant firmware must be installed in the PP50 DSP. If the firmware in the PP50 DSP is not up-to-date the PC-Tool software will show you a message which allows an automatically update.

C-DSP / P-DSP / HELIX DSP / NOX 4DSP / PP 52DSP / PP 82DSP system requirements:
- You require the supplied USB cable.
- The most recent version of the firmware must be installed in the device. If the firmware is not up-to-date, the PC-Tool software will show you a message which allows an automatically update.

A setup assistant easily guides you through the installation routine of the software

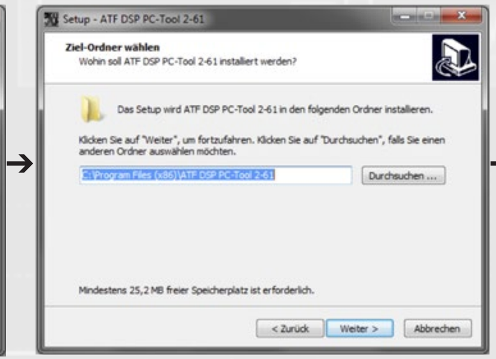


❶ Confirmation of the security warning is mandatory before starting the installation.

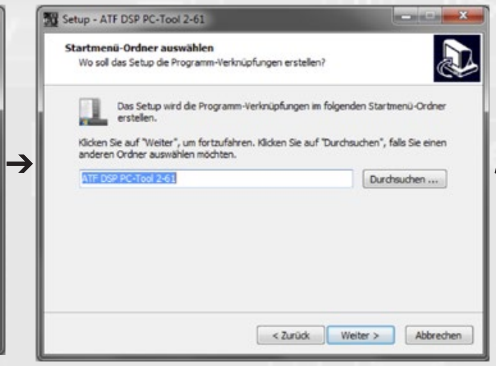
❷ The selection of the language only affects the installation routine.

❸ A click on “Continue” in the ”Welcome” window will start the setup assistant.

❹ Here you can read important information about the software itself.



❺ The target folder is already predefined; nevertheless it can be modified manually.



❻ Choose your „start menu“ folder here.

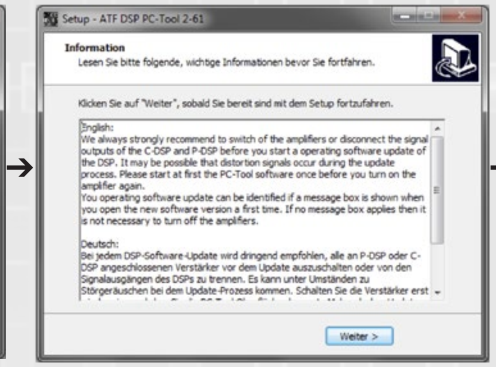
❼ The setup assistant will create a desktop symbol if you mark the checkbox here.

Important

Make sure that ATF PC-Tool software is properly installed before you connect the DSP device or USB interface to the computer.

Note that it is mandatory to connect the device prior to running the installed software.

❽ Here you can see all chosen parameters at a glance for checking purposes.



❾ The installation will start after you have read and confirmed the above information.

❿ After finalizing the installation you can directly start the software.

A.3. First operating steps with the software

Make sure that you hook up your DSP device before you start the „ATF DSP PC-Tool" software.

In most cases a USB cable establishes the connection between DSP device and your computer.

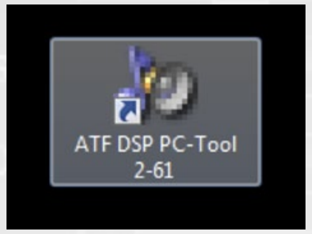
A simple USB cable (included in delivery) is typically all you need to connect any of our DSP devices with your personal computer. The PP50 DSP is the only exception - here you need a separate USB-interface.

Connect the red connector of the HELIX Optical & USB Interface to the red "CONTROL INPUT" on the PP50 DSP. Connect the HELIX Optical & USB Interface USB connector to a free USB port on your PC. Please note: The interface connector may only be inserted one way up. Do not attempt to force the connector into the "CONTROL INPUT" the wrong way up.

All other devices are connected directly to the computer using the supplied USB cable. It is not possible to use the HELIX USB Interface with this kind of processors. Use the USB port on the device only. Then insert the USB cable into a free port on the computer. Now switch on the device via the radio respectively the remote input or via the high level input (except NOX 4DSP).

Now you are able to start the software. The requirements are:

- The latest „ATF DSP PC-Tool" version has been downloaded from the Audiotec Fischer homepage.
- The software and the USB driver have been installed correctly.
- The device have been connected correctly to the computer.



In order to start the software, please execute the equal icon on your desktop.

Now the following window will appear on your screen. By pressing the „Connect" button, the device will be connected with the software.

Expert advice:

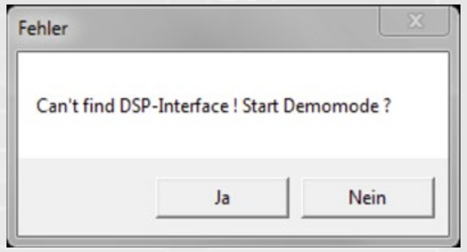
Recommended equipment for correct in-vehicle sound measurement

A device is required to measure frequency responses in order to optimize the PP50 DSP to your vehicle. There are very reasonably priced systems available which operate in combination with a PC/Notebook/Netbook and make it affordable for everyone to achieve reliable readings. The following list of components and software have been tested by us and judged to meet the operational requirements:

- „Behringer ECM8000" measuring microphone
- „T-BONE Micplug USB" microphone-USB interface with phantom power
- „Praxis" measurement software from Liberty Instruments (freeware)
- A music CD containing a pink noise test signal

The content of the memory will be read out by the software.

If no device has been hooked up or the device hasn't been turned on, then you will see the following error message:



This could happen as well if the USB driver hasn't been installed correctly. In this case please run the installation routine once again.

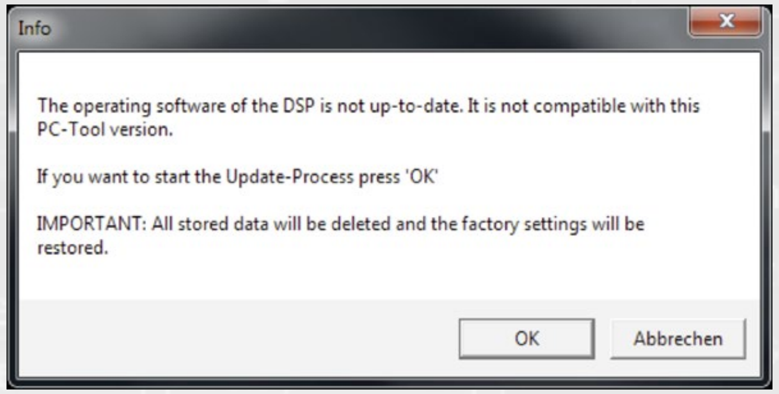
You can still start the software in demo mode. However, it is not possible to make any adjustments in demo mode as the device is not connected.

If you start in demo mode, you have the option of selecting the user interface for the specific device.

The user interface will open automatically on selection.

Please note that in demo mode it is not possible to make any changes in your device, even if you connect it to your computer thereafter. It is necessary to restart the software after hooking up the device. If the amplifier / processor has been connected correctly, it may happen that the following hint will appear within 5-15 seconds:

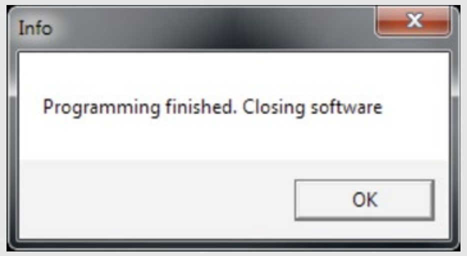
The DSP PC-Tool first checks the firmware of your DSP device and will update it automatically if necessary.



This simply means that the software in your device is not up-to-date and has to be updated prior using the PC-tool software. Confirm with "OK". Note that all stored sound setups will be erased.

After finishing the update (which will take 10 to 15 sec.) you will see the following window:

Confirm with „OK" – the PC-tool software will now be closed automatically



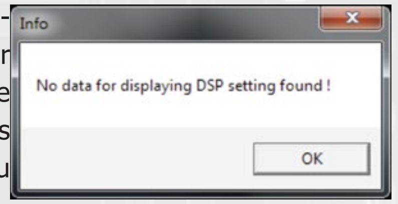
and has to be started once again.

Do not start the PC-tool software before the status LED on the amplifier/processor will light up „green". If you start the software too early it may happen that the update process will start once again!

After restarting the software the following hint will appear:

This indicates that no sound setup is stored in the internal memory of the device.

This hint only pops up after the initial installation or after a software update. As soon as you have created your own setup and stored it via the "Save DSP" button to the internal memory, the hint will not appear any longer.



Once all these steps have been done, you will see the user interface of the PC-tool software on your screen.

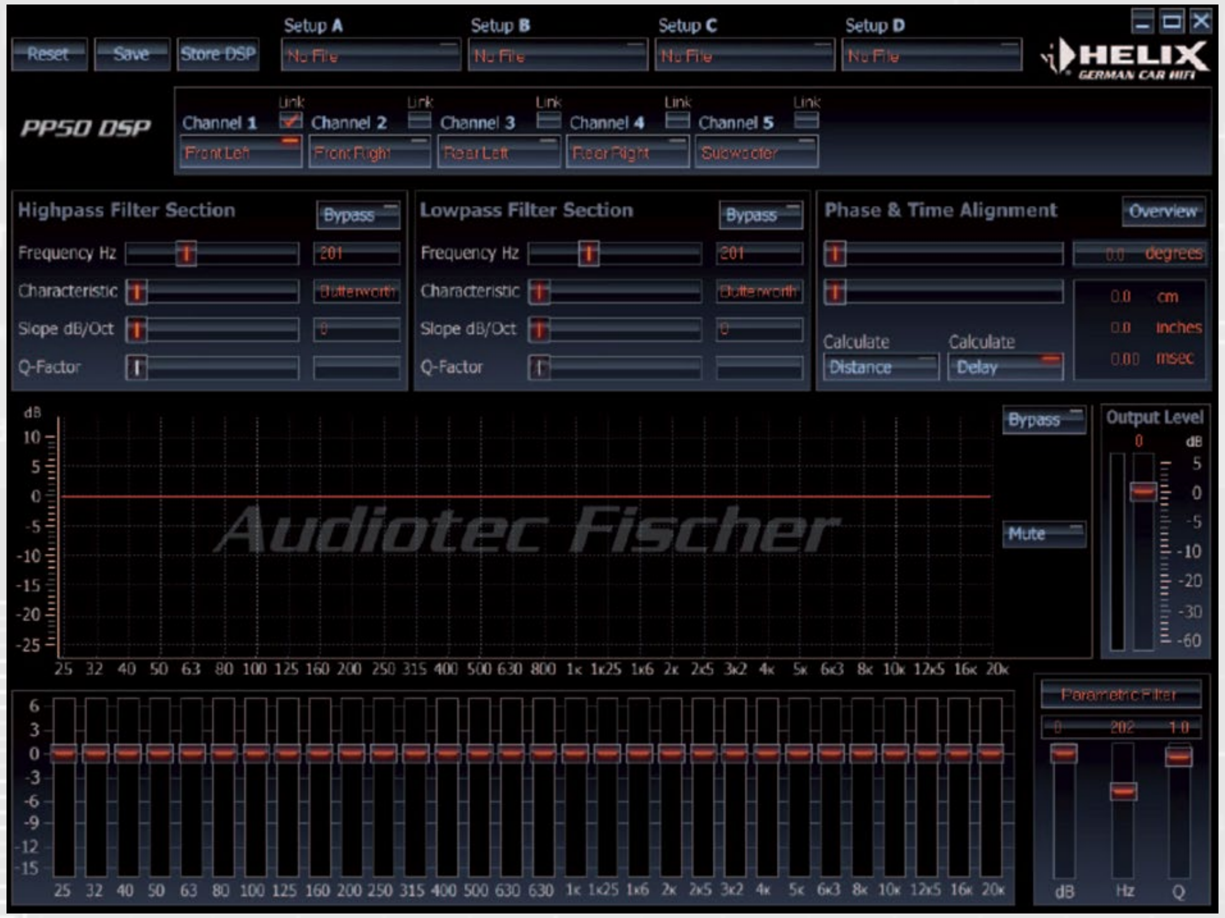
Important

If you start the software in demo mode, it will not recognize any device subsequently connected. It is then necessary to close the program completely and restart it.

Do not start the PC-tool software before the status LED on the amplifier/processor will light up „green". If you start the software too early it may happen that the update process will start once again.

B.1. The user interface

The user interface shows all sections like main navigation, channel selection, crossover, time alignment and equalizer at a glance.



The user interface is arranged top down for optimized adjustment workflow: preset selection – channel selection – crossover – time alignment – gain – equalizer.

B.1. The user interface

The user interface is divided in several sections. The upper line is the main navigation, responsible for saving of all adjustments to either your DSP device or hard disk.

The arrangement of the following sections is done in a logical order for easy adjustment. This either runs top down or from left to right:

1. Channel assignment: here you assign the inputs to the outputs of your DSP device depending on the constellation of your car stereo system.
2. Highpass and lowpass filters: here you adjust the appropriate frequency range for each channel.
3. Time alignment including phase adjustment: here you compensate the time delays caused by the different distances between the several speakers and your ears.
4. Equalizer: here you compensate the peaks and dips in the frequency response curve caused by the flaws of the loudspeakers and the car's interior acoustics.

After all settings have been done (item 1 to 4) you have to store the complete setup to your DSP device. We strongly recommend to additionally save the setup on your computer's hard drive.

! Important

If you accidentally press the "Reset" button while adjusting the DSP without having saved your setup, you will lose all your data.

It is not possible to use a „.ac1" or „.ac2" file in a different device. This may lead to severe malfunction of the unit and may even damage it.

It is then no longer possible to open, and therefore amend, the .ac1 file. We therefore strongly recommend saving the settings as .afp files as well.

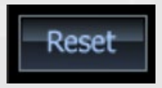
It is not possible to install the „.afp" files directly in the amplifier / processor via the micro SD card slot, the PC-Tool must be used instead

B.1.1. The main navigation menu

The main navigation line is intended for administrating the presets, the setups and data files which contain all settings.

Main navigation menu

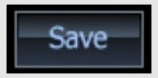
The main navigation serves memory administration. Here you save the content of each setup with all individual settings to your connected DSP device and/or to the hard drive of your PC.



Reset Button

The Reset button is used to reset all the adjustments made in the software. Pressing this button also resynchronizes the connection between the device and the software.

Important information: If you accidentally press the "Reset" button while adjusting the DSP without having saved your setup, you will lose all your data.



Save Button

Use the Save button to save an adjusted setup to a specified location on your PC. Clicking on the Save button opens a window which allows you to select the location and give the setup a file name. All setups are automatically given the file extension .afp. The .afp files contain all the PC-Tool software settings and may only be opened using this software.

The Save button is also used to save the DSP settings as a micro SD card file. To use this function, the amplifier/processor must be connected to the computer and all the specific settings already adjusted (the settings must be saved using the "Store DSP" button). Then right-click on the Save button. A window will now open which allows you to select the location. On confirmation, the program generates a micro SD card file with the file extension ".ac1" or ".ac2", depending on your choice.

An "ac1"-file will be stored in memory position 1 of the amplifier/processor whereas an "ac2"-file will be stored in memory position 2. This process can take up to a minute!

This file can then be simply installed via the micro SD card slot on any of the devices of the same type without the need for the device to be connected to the computer

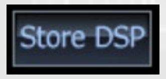
Most of our devices (here: P-DSP) allow a direct programming via PC software as well as a setup transfer via microSD memory card.

It is not possible to use such a file in a different device. For example never use an „ac1/ac2"-file that has been created for the PP 52DSP in a PP 82DSP (even not for testing purposes). This may lead to severe malfunction of the unit and may even damage it.

It is then no longer possible to open, and therefore amend, the .ac1 file. We therefore strongly recommend saving the settings as .afp files as well.

Saving the settings as an .afp file does not save them in the actual amplifier / processor. Please use the "Store DSP" button for this.

It is not possible to install the .afp files directly in the amplifier / processor via the micro SD card slot, the PC-Tool must be used instead.



Store DSP Button

The Store DSP button is used to permanently transfer the settings you made in the PC software to the amplifier's/processor's internal memory. The device then accesses this setup in its memory each time it is switched on.

The ".afp" file is also always saved in the amplifier/processor and loaded from the device when the software opens. Nevertheless, we still recommend that you also always manually save the ".afp" file on your computer.

All devices can handle two different setups in their memory. But it is not possible to store two different „.afp"-files. If you need two setups and want to switch between these, please choose the detour via generating „.ac1" and „.ac2" files (see topic "Save Button").

Setup selection

The PC-Tool software allows the direct comparison of a maximum of two different setups using the "Setup A" and "Setup B" buttons. First, load a previously adjusted setup into the selected memory by clicking with the right hand mouse button on the respective box under "Setup A - B". This opens a window in which you can retrieve the corresponding ".afp"-file from a chosen location. A setup is automatically active and audible as soon as it is loaded. Keep in mind that these setups aren't stored inside the memory of the amplifier/processor. This function is intended for direct comparison of different setups while working with the PC-tool software.

Important information:

The volume is temporarily reduced when switching between the various setups in order to avoid any potential temporary static noise when modifying the parameters.

Two setups can be directly compared with each other.

B.2. The device configuration menu (DCM)

In the „Device Configuration Menu" – abbreviated DCM – all general settings and additional configuration possibilities of the DSP device can be found.

The Device Configuration Menu (in the following simply called „DCM") allows to define several basic settings of the device or the optionally available cable remote control URC 2A.

The content of the DCM menu varies with the product – the PP 50DSP is the only amp that offers no DCM settings at all. Subsequently every single function within the DCM will be explained in detail by means of the PP 52DSP.

The following settings relate only to the DSP devices listed in the heading. The order of listing is based on the DSP device and is summarized as follows:

Clicking on „Device Configuration" (see right) will open the additional menu for several general settings.

B.2.1. Specific features in the DCM of PP 52DSP, PP 82DSP and HELIX DSP

B.2.2. Specific features in the DCM of PP 52DSP

B.2.3. Specific features in the DCM of PP 52DSP and PP 82DSP

B.2.4. Specific features in the DCM of HELIX DSP

B.2.5. Specific features in the DCM of P-DSP and NOX 4DSP

B.2.6. Specific features in the DCM of C-DSP

! Note

The content and complexity of the „Device Configuration Menu" depends on your DSP device.

B.2.1. DCM of PP 52DSP, PP 82DSP and HELIX DSP

Power management (PP 52DSP, PP 82DSP and HELIX DSP only):

The Power Save Mode allows to significantly reducing the current consumption of the device itself and connected amplifiers if no input signal is present for a certain time.

Keep in mind that many of today's cars are equipped with "CAN" or similar bus systems that keep the radio head unit "invisibly" alive for up to 45 min., even if you have already turned off the radio and left the car. Therefore any devices connected to the speaker outputs of the original radio will stay turned on for the same time and draw a lot of power from the battery.

Here's a numerical example for better understanding: A HELIX DSP is connected to the speaker outputs of an OE car radio and turned on by these as well. The „remote out" of the HELIX DSP controls two 4-channel power amplifiers, each with a typical quiescent current of 2 amperes (quiescent current= amp turned on, but no input signal).

After leaving the car the OE radio will remain invisibly turned on for 30 min. Means that the HELIX DSP will be turned on for the same time!

Current draw in Ah (ampere hours) without Power Save Mode:

0.4 A (HELIX DSP) + 2x 2 A (connected amps) x 0.5 hours = 2.2 Ah

Current draw with Power Save Mode:

0.4 A (HELIX DSP) + 0.0 A (amps turned off) x 0,5 hours = 0.2 Ah

Because the HELIX DSP turns off the amps as no input signal is present. While the "Power Save Mode" is active, in this specific example the current draw from the car's battery will be reduced by approx. 90%.

The higher the idle current of the connected amplifier is, the more extreme the power saving will be. Especially in winter time when the car's battery is already stressed by low outside temperatures, the „Power Save Mode" avoids any needless additional load.

The slider control allows to vary the time period until the device activates the „Power Save Mode" from 30 seconds to 300 seconds in increments of 30 sec. (default value: 60 sec.). As soon as a signal is detected on one of the inputs of the amplifier the device will switch back to normal operation within 1 sec.

If you like to fully deactivate the „Power Save Mode" then insert a tick on „Deactivate Power Save Mode".

! Expert advice:

The intelligent "Power Save Mode" can reduce the current draw from your car's battery by up to 90% when no music signal is present. The higher the idle current of your amps the bigger the energy savings will be.

You'll find further information about the DCM of HELIX DSP in chapter B.2.4. on page 16.

B.2.2. DCM of PP 52DSP

Impedance selector (PP 52DSP only)

Here you can choose the load impedance of the front and rear channels of the PP 52DSP.

Typically in cars speakers with an impedance of 3-4 ohms are used (set to "4 Ohm" position); nevertheless if you use speakers with lower impedance then choose the „2 Ohm" position. Note: This selector has no influence on the subwoofer speaker outputs. These can handle 2 ohms load at any time.

The „Impedance Selector" defines the minimum load for the speaker output channels.

B.2.3. DCM of PP 52DSP and PP 82DSP

DIP switch control (PP 52DSP and PP 82DSP only)

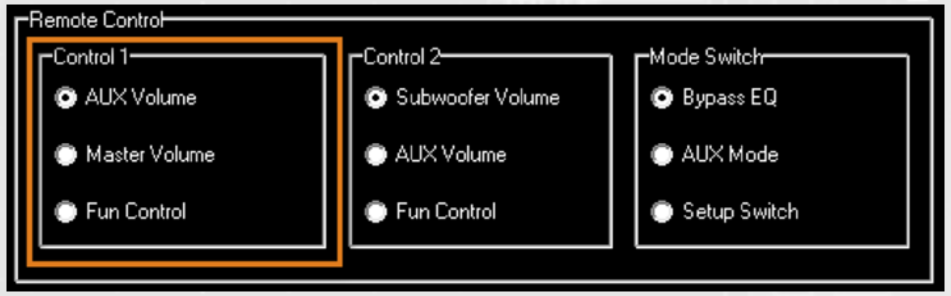
If you have an optional remote control (z.B. URC 2A) hooked up and you want to modify its functionality in the DCM, you have to checkbox „Activate Remote Control" first.

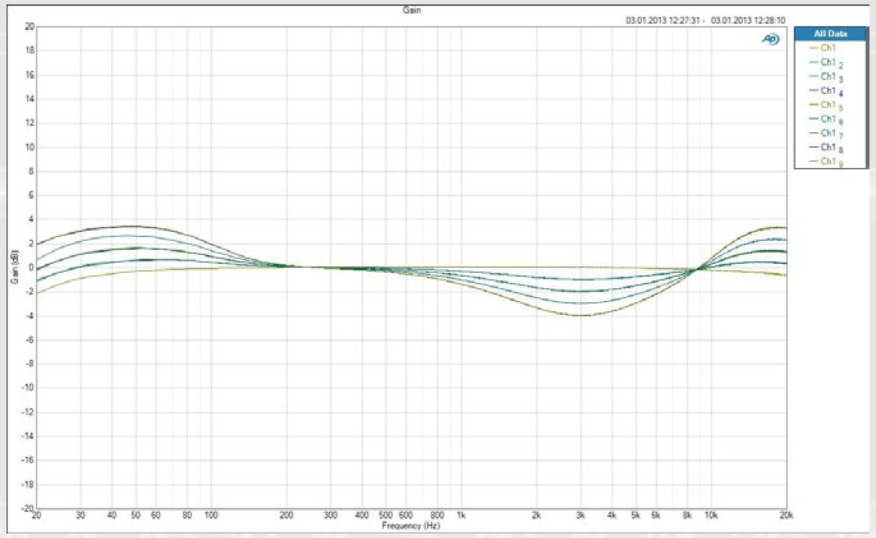
Remote control „Control 1" (PP 52DSP and PP 82DSP only)

This selector defines the functionality of the upper rotary knob "Control I" on the URC 2A remote:

Position „Aux Volume"
Allows controlling the volume of a connected AUX signal source. Therefore it is no longer necessary to adjust the volume on the source itself. All other signal inputs remain unaffected

Position „Master Volume"
Allows to control the volume of all outputs at the same time. This function makes sense if none of the sources have an own volume control.

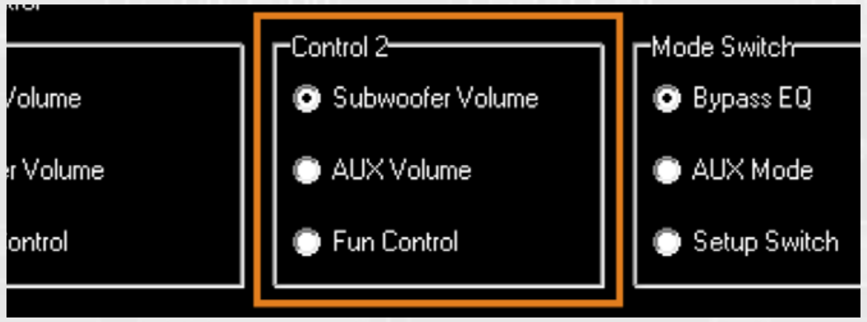




Position „Fun Control" (PP 52DSP only)
Allows to slightly adjust the sound of the amplifier according to personal preferences.
- Full counterclockwise position: no additional sound adjustment
- Full clockwise position: maximum additional sound adjustment.
The graph on the left illustrates the functionality of this rotary control.

Remote control „Control 2“ (PP 52DSP and PP 82DSP only)

This selector defines the functionality of the lower rotary knob „Control II“ on the URC 2A remote.



Position „Subwoofer Volume“
Allows adjusting the volume of all channels that have been defined as subwoofer channel in the “Input/Output-Matrix”.
PP 52DSP: adjust as well the output volume of the internal amps for the subwoofer channel.

Position „Aux Volume“
(PP 52DSP and PP 82DSP only)
Allows controlling the volume of a connected AUX signal source. Therefore it is no longer necessary to adjust the volume on the source itself. All other signal Inputs will not be affected.

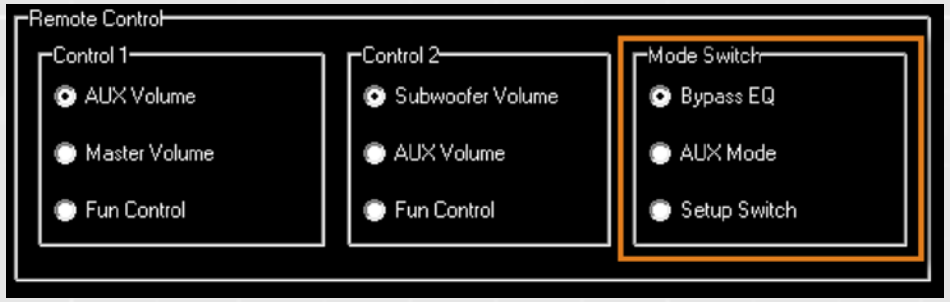
Position „Fun Control“
(nur PP 52DSP)
Allows to slightly adjust the sound of the amplifier according to personal preferences.

- Full counterclockwise position: no additional sound adjustment
- Full clockwise position: maximum additional sound adjustment

For better illustration see graph on previous page.

Remote control „Mode switch“ (PP 52DSP and PP 82DSP only)

This selector defines the functionality of the „MODE“ switch on the URC 2A remote:



Position „Bypass EQ“
(PP 52DSP and PP 82DSP only)
Allows to bypass all filters and equalizers as well as the time alignment of the front and rear channels (bypass = switch on remote is depressed). Additionally all other amplifier channels will be muted. So it is easily possible to toggle between the „original sound” without any DSP improvement and the optimized sound setting.

Position „AUX Mode“
(PP 52DSP and PP 82DSP only)
Allows to manually activate the auxiliary input. In this case the automatic signal detection for the aux input will be deactivated.

Position „Opto Mode“
(PP 82DSP only)
Allows to manually activate the optical digital input. This is the one and only possibility to select the optical input of the PP 82DSP.

Position „Setup Switch“
Allows toggling between two different setups. This function is only given under the following circumstances:

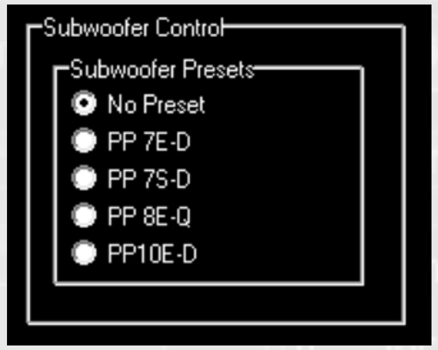
- Two different setups are stored in the internal memory (an „af1“ or „ac1“- file and an “af2” or “ac2”-file). If there is only one single setup stored in the internal memory, the “MODE” switch will have no effect.
- It is mandatory that in both setups the „Mode Switch“ in the DCM has been defined as „Setup Switch“.

In this case it is not possible to define a different functionality for the “Mode Switch” for each setup.
This „Setup Switch“ does not work in combination with car-specific setups that are available on the Audiotec Fischer Website. In these setup files the functionality of the MODE switch on the URC 2A remote is defined as “Bypass EQ” (no changes possible). This feature only properly works in combination with the remote control. Of course it will be possible to switch between the settings using the “control” key on the device. But the amplifier DSP will automatically switch back to setup 1 when turned on the next time if no URC 2A is connected

B Operation – B.2. Device Configuration Menu

Subwoofer control (PP 52DSP and PP 82DSP only)

If you connect a MATCH subwoofer to the PP 52DSP (8-pole connector “SUBWOOFER OUTPUT”) or PP 82DSP (8-pole connector “OUTPUT CHANNELS E-H”), then simply choose here the appropriate model for optimzed settings of highpass and lowpass filters. Of course you can modify these settings later on according to your personal demands.



AUX signal management (PP 52DSP and PP 82DSP only)

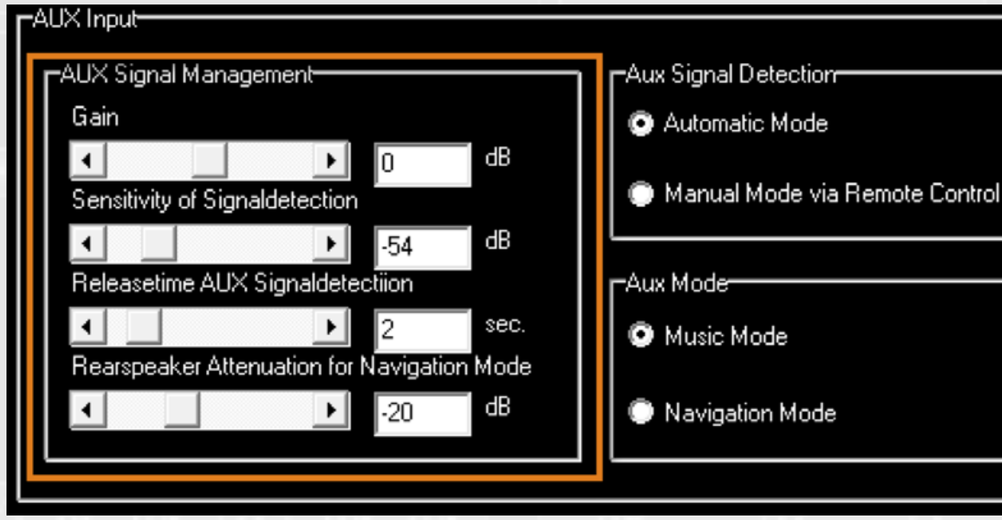
The sub menu „AUX Input“ defines the behaviour of the aux input.

Gain
The slider control „Gain“ defines the sensitivity of the aux input. It is possible to vary it from -6dB to +6dB in steps of 1dB (default value: 0dB).

Sensitivity of Signaldetection
The aux input will be activated automatically if an input signal is applied (precondition: selector „Aux Signal Detection“ is set to „Automatic Mode“). The slider control „Sensitivity of Signaldetection“ allows you to adjust whether even very weak signals will activate the aux input (position „-60dB“) or higher input level is required (position „-30dB“).
The latter may be necessary if electrical disturbances of other components inside the vehicle will activate the aux input though no music signal is present. The sensitivity of the signal detection can be varied from -60dB to -30 dB in steps of 1dB (default value: -54dB)

Releasetime AUX Signaldetection
The slider control „Releasetime AUX Signaldetection“ defines the time until the ampifier will automatially switch back to the radio source when no more signal is present on the aux input.
If you have connected a Sat Nav device to the aux input (so that you can hear the route guidance announcements via the car speakers) then a short releasetime of approx. 1 sec. is a good choice. On the other hand if you have connected an MP3-player a release time of 5 sec. or even more might be useful. Otherwise it may happen that during the pause between two music tracks (typically 2-3 sec.) the aux input will automatically switch back to the radio source for a short amount of time. The release time can be adjusted from 1 to 10 sec. in increments of 1 sec. (default value: 2 sec.)

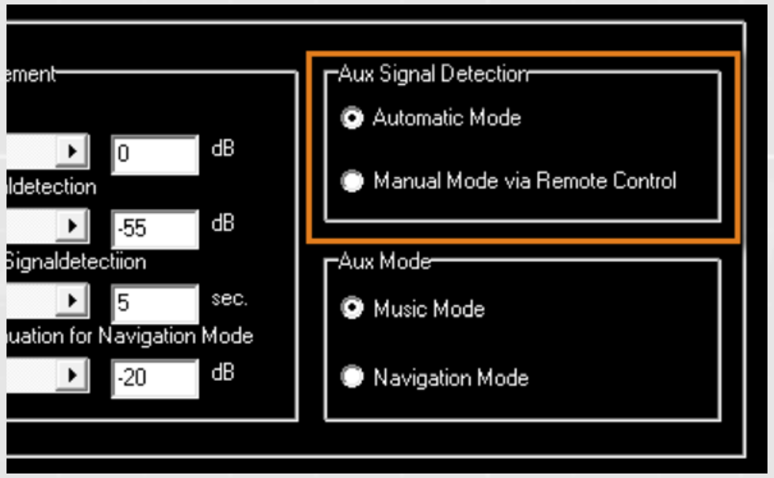
Rearspeaker Attenuation for Navigation Mode
If you have defined the aux input as „Navigation Mode“ then the signal of the aux input will only be routed to the front channels. The rear channels will continue playing back the signal of the radio source.
With the slider control „Rearspeaker Attenuation for Navigation Mode“ the level of the rear channels can be varied from -30dB to 0dB in steps of 1dB (default value: -20dB).



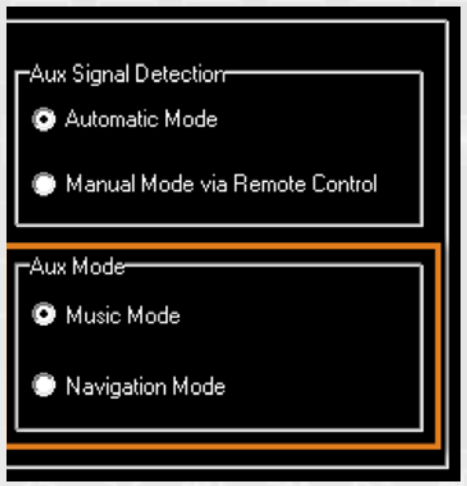
Aux signal detection (PP 52DSP and PP 82DSP only)

This selector defines whether activating the AUX input is done automatically (via automatic signal detection) or manually via the MODE switch on the URC 2A remote.

As soon as you choose „Manual Mode via Remote Control" the „Mode switch" selector in the DCM will automatically be set to „Aux Mode".



Aux mode (PP 52DSP and PP 82DSP only)



This switch defines what the Aux input is used for. Choose the "Music Mode" if you have connected an MP3-player or any other music source.

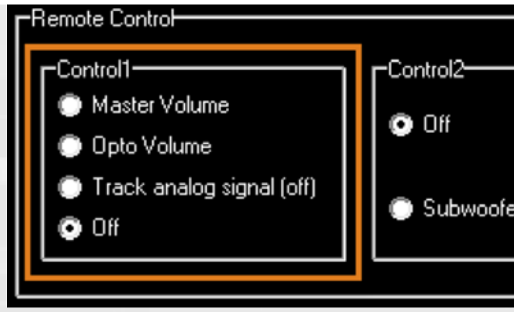
On the other hand, if you have connected a portable Sat Nav device and like to playback the route guidance announcements via the car speakers then the "Navigation Mode" is the right choice. Here are the differences described in detail:

„Music Mode": the signal of the source connected to the aux input will be routed to all output channels.

„Navigation Mode": the aux signal will in this case only be routed to the front channels; the rear channels continue to play back the other signal source (car radio). The volume of the rear channels in this specific mode can be adjusted using the slider control „Rearspeaker Attenuation for Navigation Mode". Hint: If you use the „Navigation Mode" we recommend setting the „Aux Signal Detection" to „Automatic Mode".

B.2.4. DCM of HELIX DSP

Remote control „Control 1" (HELIX DSP only)



This selector defines the functionality of the upper rotary knob "Control I" on the URC 2A remote.

Position „Master Volume"
Allows controlling the volume of all outputs at the same time. This function makes sense if none of the sources have an own volume control.

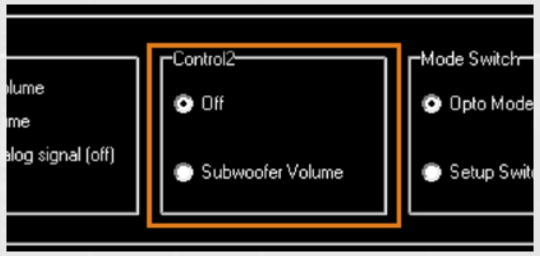
Position „Opto Volume"
Allows controlling the volume of the optical input only. All other signal inputs will not be affected.

Position „Track analog signal off"
If this position is chosen, the rotary knob „Control I" will have no function. In this case the volume of the optical input depends on the level of the audio signal that is applied to the analog inputs A and/or B.

This input signal can be identical to the optical input signal. A much better volume control is realized when you apply a constant noise signal (e.g. from the tuner which isn't set to any radio station) to the analog inputs.

Position „Off"
The rotary knob „Control I" on the remote has no function.

Remote control „Control 2" (HELIX DSP only)



This selector defines the functionality of the lower rotary knob "Control II" on the URC 2A remote.

Position „Off"
The rotary knob „Control II" on the remote has no function.

Position „Subwoofer Volume"
Allows adjusting the volume of all channels that have been defined as subwoofer channel in the "Input/Output-Matrix". In case of the HELIX DSP this could be the channels G and/or H.

Remote control „Mode switch" (HELIX DSP only):

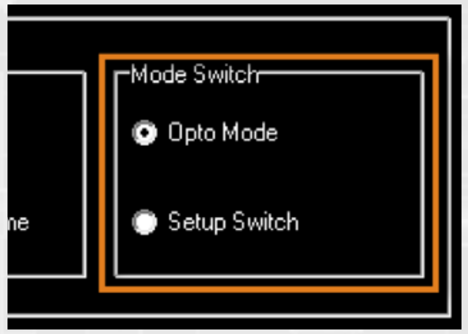
This selector defines the functionality of the „MODE" switch on the URC 2A remote:

Position „Opto Mode"
Allows to manually activate the optical digital input. The automatic detection of the optical input will be deactivated.

Position „Setup Switch"
allows toggling between two different setups. This function is only given under the following circumstances:

- Two different setups are stored in the internal memory (an „af1" or „ac1"- file and an "af2" or "ac2"-file). If there is only one single setup stored in the internal memory, the "MODE" switch will have no effect.
- It is mandatory that in both setups the „Mode Switch" in the DCM has been defined as „Setup Switch".

In this case it is not possible to define a different functionality for the "Mode Switch" for each setup. This feature only properly works in combination with the remote control. Of course it



will be possible to switch between the settings using the "control" key on the device. But the HELIX DSP will automatically switch back to setup 1 when turned on the next time if no URC 2A is connected.

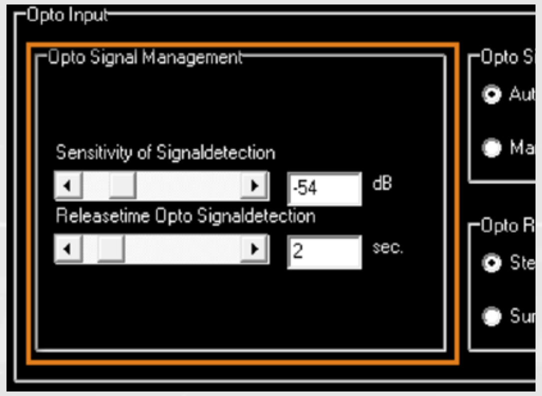
Opto input (HELIX DSP only)

This sub menu „Opto Input“ defines the behaviour of the optical digital input.

The device will automatically switch to the optical input as soon as any input signal is detected (pre-condition: selector "Opto Signal Detection" is set to "Automatic Mode").

The slider control „Sensitivity of Signaldetection“ allows you to adjust whether even very weak signals will activate the optical input (position „-60dB“) or a higher input level is required (position „-30dB“).The sensitivity of the signal detection can be varied from -60dB to -30 dB in increments of 1dB (default value: -54dB).

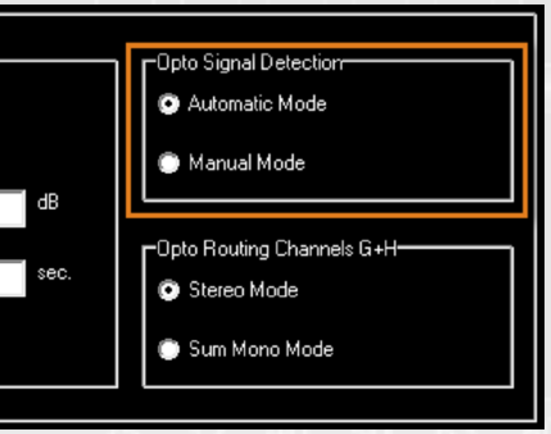
The slider control „Releasetime Opto Signaldetection“ defines the time until the processor will automatically switch back to the analog source when no more signal is present on the optical input. The release time can be adjusted from 1 to 10 sec. in increments of 1 sec. (default value: 2 sec.).



Opto signal detection (HELIX DSP only)

This selector defines whether activating the optical input is done automatically (via automatic signal detection) or manually via the MODE switch on the URC 2A remote.

As soon as you choose „Manual Model“ the „Mode switch“ selector in the DCM will automatially be set to „Opto Mode“.



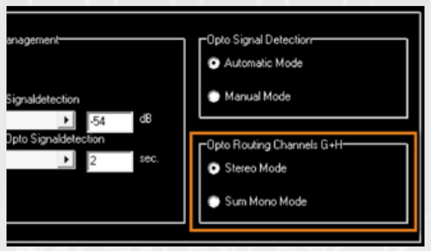
Opto routing channels G + H (HELIX DSP only)

This switch defines whether the channels G and H will reproduce a stereo signal or a sum mono signal of the optical input.

Here are the differences in detail:

„Stereo Mode“
The signal of the left channel of the optical signal input will be routed to the output channel G, whereas the signal of the right optical input channel will be routed to output channel H.

„Sum Mono Mode“
Both input channels of the optical digital input will be mixed together to a sum mono signal and routed to both output channels G and H.



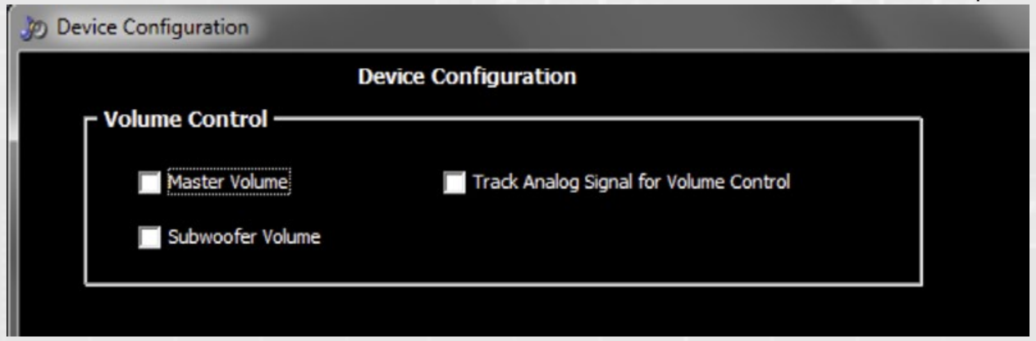
B.2.5. DCM of P-DSP and NOX 4DSP

Volume Control (P-DSP and NOX 4DSP only)

Position „Master Volume“
If you tick this function, then the upper rotary knob "CONTROL I" on the remote URC 2A will influence the volume of all outputs at the same time. This function makes sense if a digital source is connected to the optical signal input.

Position „Track Analog Signal for Volume Control“
If this position is chosen, the rotary knob „Control I“ will have no function. In this case the volume of the optical input depends on the level of the audio signal that is applied to the analog inputs A and/or B. This input signal can be identical to the optical input signal. A much better volume control is realized when you apply a constant noise signal (e.g. from the tuner which isn't set to any radio station) to the analog inputs.

Position „Subwoofer Volume“
Allows adjusting the volume of all channels, that have been defined as subwoofer channel in the "Input/Output-Matrix"



B.2.6. DCM of C-DSP

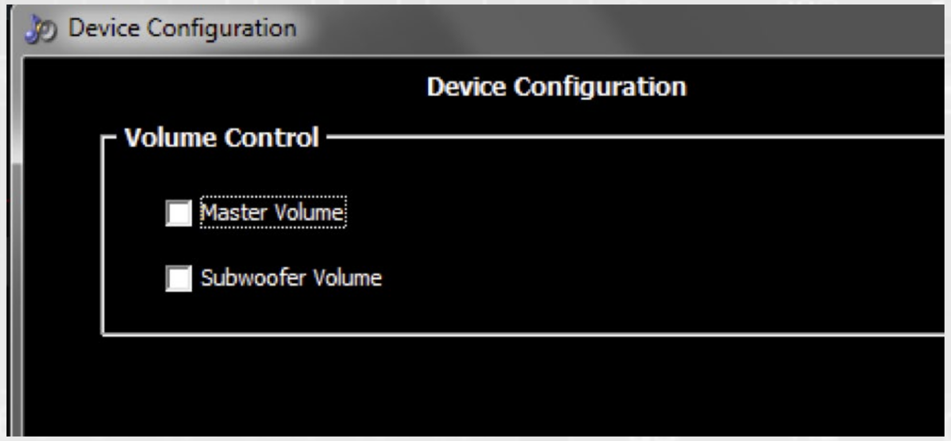
Volume Control (C-DSP only)

Position „Master Volume“
If you tick this function, then the upper rotary knob "CONTROL I" on the remote URC 2A will influence the volume of all outputs at the same time. This function makes sense if a digital source is connected to the optical signal input.

Position „Subwoofer Volume“
Allows adjusting the volume of all channels, that have been defined as subwoofer channel in the "Input/Output-Matrix".

Expert advice:

If you setup your DSP device for the very first time or define a new sound system configuration please make sure that you properly do the right settings in the DCM prior to all other adjustments! After that you can continue with the channel routing (page 20).



B.3. Channel routing – Configure the Inputs & Outputs

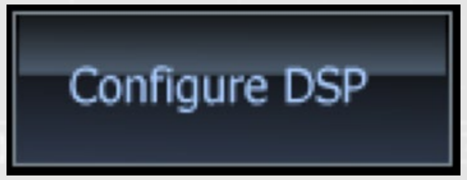
The channel routing „organizes“ all inputs and outputs. Summing up input signals is done here as well.

The HELIX DSP provides three sets of stereo preamplifier inputs (RCA/Cinch; A-F) and an optical stereo digital input. In addition the inputs A/B and C/D offer highlevel inputs which can be directly connected to the speaker outputs of a car radio.

The „Configure Inputs / Outputs“-Button

This button is used to switch to the channel routing overview in which you can configure any of the inputs and outputs. This allows you to assign names to each of the individual channels, to add up input signals and allocate a specific input signal to each output. To protect the connected loudspeaker, default parameters for each of your settings for each channel will be loaded automatically, which may, of course, be amended as required.

The button „Configure DSP“ is used to return to the main PC-Tool page from the channel routing user interface.

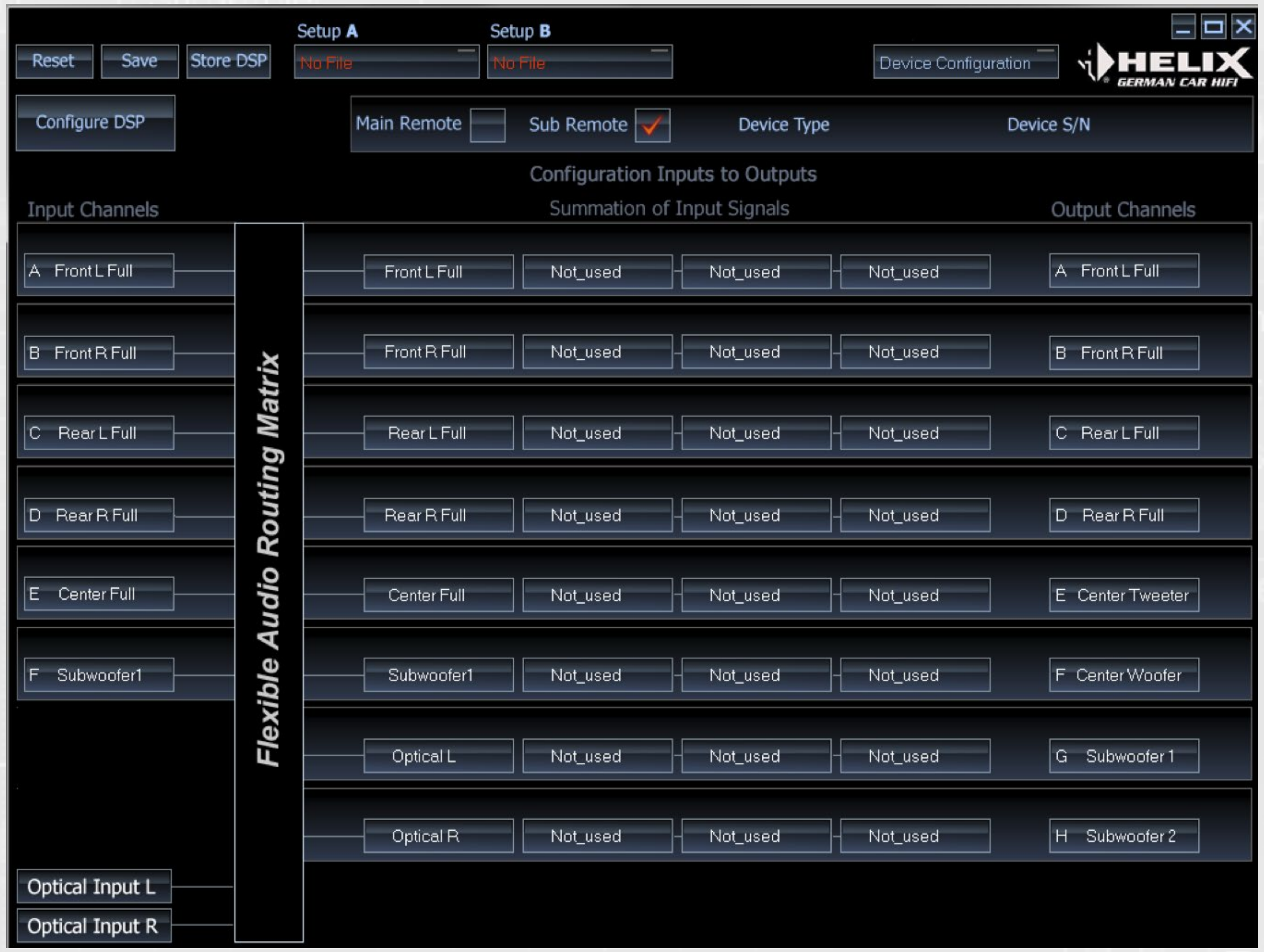


Note

The channel routing overview may only be found in the PC-Tool user interface of all products except the PP50 DSP.

The extent of possibilities and number of allocable channels depends on the device itself. Only the available channels are shown in the channel routing overview.

The HELIX DSP offers eight output channels in total – therefore you can find four output pairs (RCA/Cinch; A–H) on its back panel. The routing matrix (see next page) defines which input channel will be allocated to which output channel. Even mixing of several input channels to one output channel is possible.



This is the input mask for the channel routing (HELIX DSP shown here). The left side shows all inputs, the right side all output channels. Each output channel can be the sum of up to four arbitrary input channels, displayed in the same line as the output itself.

B.3.1. Naming inputs

On the left hand side of this mask you can allocate a name to each of the input channels (C-DSP: "A-H", P-DSP and HELIX DSP: "A-F and NOX4 DSP „A-D“) to give you the best possible overview. Make the transfer using the right and left mouse buttons. PP 52DSP and PP 82DSP do not allow to change names of the inputs.

This setting does not affect the functionality. It is merely used as an overview and helps you to better differentiate between the channels. The following names may be allocated:

Note

This setting does not affect the functionality. It is merely used as an overview and helps you to better differentiate between the channels.

- Front L Fullrange
- Front L Tweeter
- Front L Midrange
- Front L Woofer
- Front R Fullrange
- Front R Tweeter
- Front R Midrange
- Front R Woofer
- Center Fullrange
- Center Tweeter
- Center Woofer
- Rear L Fullrange
- Rear L Tweeter
- Rear L Midwoofer
- Rear R Fullrange
- Rear R Tweeter
- Rear R Midwoofer
- Subwoofer

B.3.2. Naming the outputs (C-DSP, P-DSP, HELIX DSP and NOX 4DSP only)

Just as with the inputs, you are also able to allocate names to the outputs "A-H" (C-DSP, P-DSP and HELIX DSP) respectively „A-F" (NOX 4DSP) on the right hand side of the input mask. They are selected using the right or left mouse button.

In this case, however, the selected configuration does immediately affect the default settings of the high-pass and lowpass filters in the DSP main menu. This therefore prevents, for example, a tweeter connected to output A accidentally being assigned a fullrange signal which may damage it. Any unused output channels should ideally be muted ("Not_used").

Default settings of individual channel types

Output channel	Default Highpass	Default Lowpass
Not used	Channel output is muted	
Front L Fullrange	–	–
Front L Tweeter	3000 Hz, 12dB, Butterworth	–
Front L Midrange	500 Hz, 12dB, Butterworth	3000 Hz, 12dB, Butterworth
Front L Woofer	–	3000 Hz, 12dB, Butterworth
Front R Fullrange	–	–
Front R Tweeter	3000 Hz, 12dB, Butterworth	–
Front R Midrange	500 Hz, 12dB, Butterworth	3000 Hz, 12dB, Butterworth
Front R Woofer	–	3000 Hz, 12dB, Butterworth
Center Fullrange	200 Hz, 12dB, Butterworth	–
Center Tweeter	3000 Hz, 12dB, Butterworth	–
Center Woofer	–	3000 Hz, 12dB, Butterworth
Rear L Fullrange	–	–
Rear L Tweeter	3000 Hz, 12dB, Butterworth	–
Rear L Midrange	500 Hz, 12dB, Butterworth	3000 Hz, 12dB, Butterworth
Rear L Woofer	–	3000 Hz, 12dB, Butterworth
Rear R Fullrange	–	–
Rear R Tweeter	3000 Hz, 12dB, Butterworth	–
Rear R Midrange	500 Hz, 12dB, Butterworth	3000 Hz, 12dB, Butterworth
Rear R Woofer	–	3000 Hz, 12dB, Butterworth
Rear Fill	200 Hz, 12dB, Butterworth	–
Subwoofer 1	–	80 Hz. 24dB, Butterworth
Subwoofer 2	–	80 Hz. 24dB, Butterworth

! Note

These default settings may, of course, be modified in the DSP main menu. They are only used to protect the connected loudspeaker and are not recommendations for correct settings.

Default settings of individual channel types (PP 52DSP and PP 82DSP only)

For PP 52DSP and PP 82DSP only two different settings can be chosen. Either a fullrange signal or a preset for subwoofer applications.

As soon as you select one of the MATCH subwoofers in the Device Configuration Menu under "Subwoofer Control", the appropriate optimized setting for the highpass filter will automatically be applied.

There are two modes selectable for the preamp outputs of the PP 52DSP – either "full" for "fullrange" or "sub" for subwoofer mode.

Output channel	Default Highpass	Default Lowpass
Out Full #(A-H)	–	–
Out Sub #(A–H)	depending on setup in „DCM"	80 Hz 24dB, Butterworth

B.3.3. Allocating inputs to outputs

In the center of the input mask, you can allocate any of the input signals to each of the eight outputs A-H (NOX4 DSP: A-F).

You can also use either the right or left mouse button to make this selection.

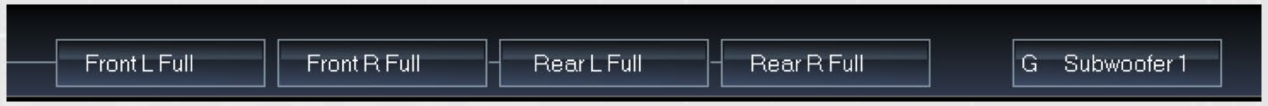
It is mandatory to allocate at least one input to every output that you like to use.

If you add two or more input channels to one output, each input channel is weigthed equally.

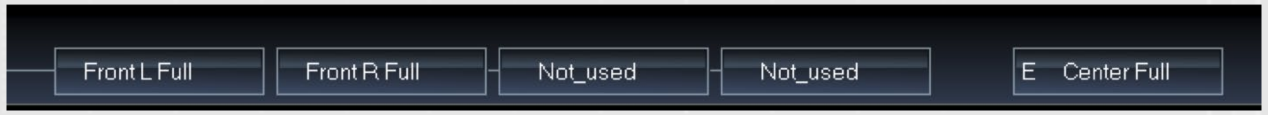
You do not have to use all the inputs. It is entirely possible to generate all outputs channels from only two input channels.

We recommend setting all unused inputs and outputs to "Not_used".

Here are some examples:



If you want to generate a sum signal for a subwoofer output from all four front and rear channels, then please choose this adjustment.



If you like to generate a front center signal from both front channels then this adjustment makes sense.

If you only use the optical digital inputs, a multichannel application with center speaker and subwoofer would look like this.

You will find further examples in the appendix starting on page 42.

! Note

In previous versions of the ATF DSP PC-Tool it was mandatory to assign all four input fields – otherwise it wasn't possible to reach the maximum output level. The current version doesn't require this anymore as the gain will be adjusted automatically. Even if you enter the same input channel in all four input fields this will not have any effect on the output level.

B.4. High- and lowpass filter settings

The crossover with highpass and lowpass filters allocates the right frequency range for each specific speaker type.

Selecting channels for adjustment / Linking channels

Select the channel you want to adjust. To do so, simply click on the respective box with the left mouse button. The active channel is indicated by the red LED (top front left in the example).

Linking channels

The software allows each channel to be adjusted separately. In spite of this, it often makes sense to link the two front channels, or the two rear channels, for example, and therefore adjust them simultaneously. This is particularly recommended for the high- and lowpass filters since having different values for the right and left channels should be avoided. Even with the equalizer, a separate adjustment of channels only makes sense if sound optimization is only required for a single listening position.

Simply insert a tick by the channels you want to adjust simultaneously (in the example above, the two front channels).

! Important

When linking two channels, the adjustments made previously to one channel are not automatically transferred to the other channel. Only those adjustments made after the link has been created and then applied in an identical manner to the linked channels. Therefore, please consider whether two or more channels should be adjusted simultaneously before making any adjustments.

Adjustment of high- and lowpass filter

This section makes it possible to configure almost any type of high-pass filter (left side) and lowpass filter (right side) for the selected channel. When doing so, the following parameters are adjusted separately:

- crossover frequency
- filter characteristics
- slope
- Q-Factor

(The „Q-Factor"-slide control is only active if you select the "Self-Define" characteristics).
The functions of the individual parameters are described below.

The adjustment for the highpass filter has to be made in the left section whereas the adjustment for the lowpass filter is done in the right section. Both filters affect the selected channel (or more than one channel if you have linked them before).

B.4.1. Adjustment of crossover frequency

Use the slide control to select the required crossover frequency for the high-pass or lowpass filter. The setting accuracy depends on the device type and is shown in the chart.

DSP device	Channel	Filter	Steps	frequency range
PP50 DSP	Front / Rear	High- / Lowpass	1/6 Octave	20 Hz – 20.480Hz
	Subwoofer	Highpass	1 Hz	20 Hz – 100 Hz
		Lowpass		20 Hz – 300 Hz
P-DSP	Fullrange	High- / Lowpass	1/12 Octave	20 Hz – 20.480Hz
	Subwoofer	Highpass	1 Hz	20 Hz – 100 Hz
		Lowpass		20 Hz – 300 Hz
C-DSP NOX 4 DSP HELIX DSP PP 52DSP PP 82DSP	Fullrange	High- / Lowpass	1/24 Octave	20 Hz – 20.480Hz
	Subwoofer	Highpass	1 Hz	20 Hz – 100 Hz
		Lowpass		20 Hz – 300 Hz

Important Note about PP 50DSP

If you are using the PP50 DSP in combination with the MATCH subwoofers, we recommend to adjust the subwoofer channel highpass filter settings as follows:

PP 7E/PP 7E-D/PP 7S/PP 7S-D:
• Crossover frequency: 49 Hz
• Charakteristik: Self-Define
• Q-Faktor: 1.5

PP 8E-Q:
• Crossover frequency: 46 Hz
• Charakteristik: Self-Define
• Q-Faktor: 1.2

PP 10E / PP 10E-D:
• Crossover frequency: 38 Hz
• Charakteristik: Self-Define
• Q-Faktor: 1.2

B.4.2. Adjustment of filter characteristics

Butterworth (Q = 0,707)
- Rather good impulse response
- Sharp transition from the passband to the stopband
- 3 dB gain in the crossover frequency range
- The most popular characteristics, suitable for almost all applications

Bessel (Q = 0,577)
- Very good impulse response
- Very gradual transition from the passband to the stopband
- Requires loudspeakers that have a smooth frequency response even outside their passband
- Less than 1 dB gain in the crossover frequency range
- Only practical in combination with very high quality loudspeaker systems; not recommended for subwoofers

Linkwitz (Q = 0,5)
- Very good impulse response
- No steep transition from the passband to the stopband
- No gain at the crossover frequency
- Only useful in combination with very high quality loudspeaker systems; not particularly recommended for subwoofers

Tschebyschev (Q = approx. 0,9)
- Very sharp transition from the passband to the stopband
- Poor impulse response
- No flat frequency response (1 dB "ripple")
- Only recommended as a lowpass for subwoofer or as a high-pass as required for tweeters which are operated close to their resonance frequency

Self-Define
- Only available as a 12 dB filter, with adjustable Q factor
- Useful as a high-pass filter in so-called "filtered bass reflex systems", whereby the crossover frequency usually corresponds to the tuning frequency of the bass reflex port

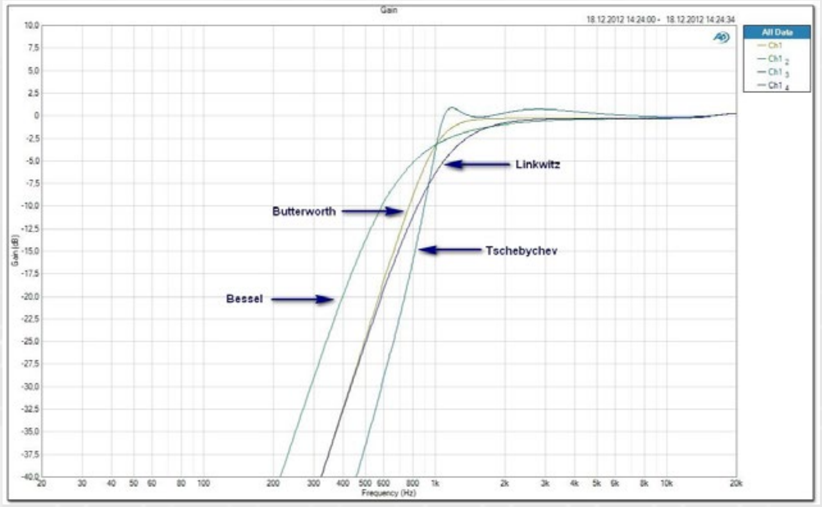
There are four different filter characteristics available for selection and the option of setting your own highpass and lowpass filter using "Self-Define". The table on the left presents an overview of the typical properties of the different filter characteristics.

If in doubt, choose preferably „Butterworth". This filter function also use normal commercial crossovers.

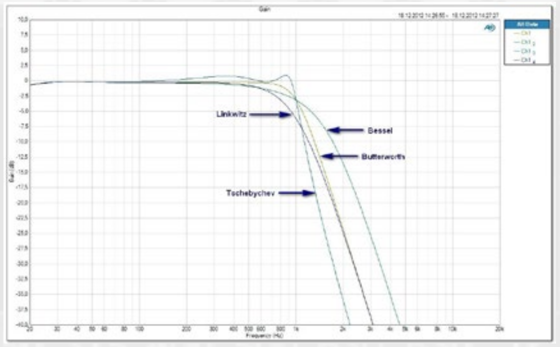
Note

If you have selected the "Linkwitz" filter characteristics, you are only able to set the values to "-12 dB" and "-24 dB" (-36dB as well for C-DSP and NOX4 DSP).

The "Self-Define" characteristic always has a slope of -12 dB per octave.



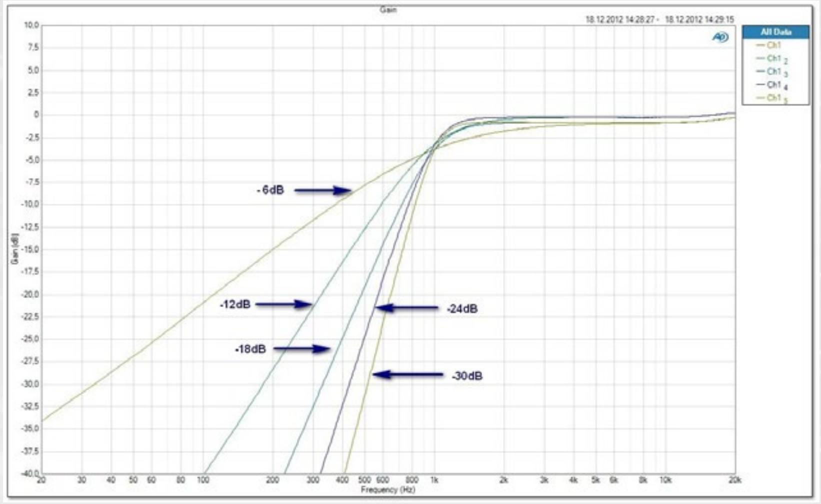
Examples of the different filter characteristics are presented for a 1000 Hz highpass filter (above) and lowpass filter (right) with a slope of 24 dB per octave.

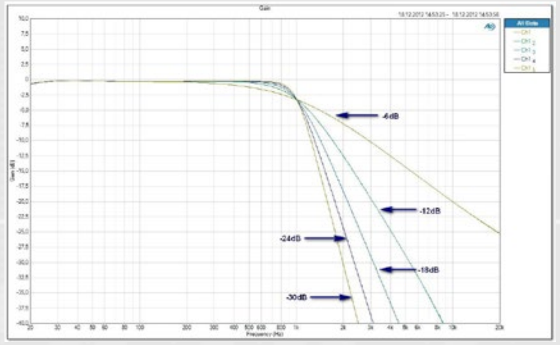


B.4.3. Adjustment of slope

Use this slide control to adjust the slope of the filter in 6 dB increments from "0 dB" per octave (filter not active) to a maximum of "-42 dB" per octave on C-DSP / NOX4 DSP and a maximum of 30 dB per octave (= filter with very steep slope) on the P-DSP, HELIX DSP, PP50 DSP, PP 52DSP and PP 82DSP.

The effect of the slope on the frequency characteristics of a filter is illustrated in the grafs by using the examples of two 1000 Hz filters.





The effect of the slope on the frequency characteristics of a filter is illustrated using the example of a 1000 Hz highpass filter (above) and a lowpass filter (left), both with Butterworth characteristic.

Tips for adjustment

If in doubt, choose preferably „Butterworth". This filter function also use normal commercial crossovers.

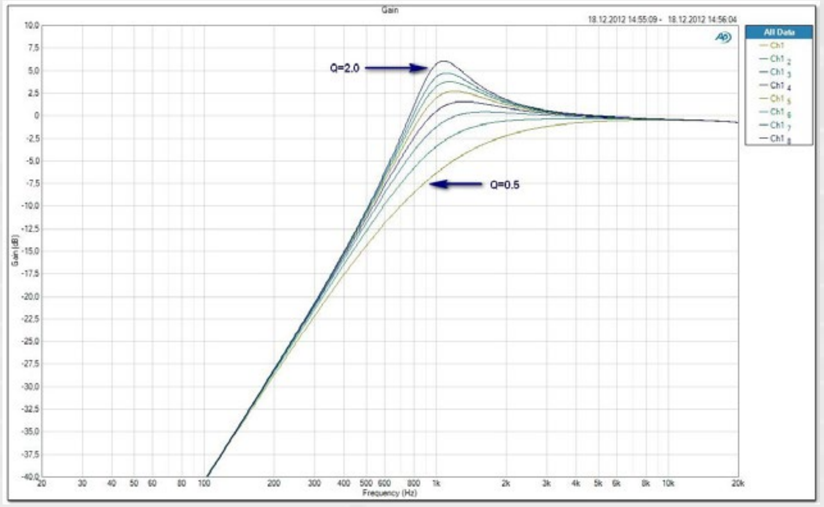
The value you select for the slope depends heavily on the type of application. The following points may aid your decision:

- The steeper the slope, the worse the filter impulse response.
- The preferred slope for the subwoofer channel lowpass filter is "-24 dB".
- A standard value for the crossover network between the woofer and the tweeter in fully active systems is "-12 dB".
- If the frequency response of a midwoofer shows a lot of strong peaks outside its typical operational frequency range, it may be useful to select a steeper slope (e.g. -24 dB per octave) for the lowpass filter.
- A small 19 mm tweeter operated in a fully active system at up to 3000 Hz also requires a steeper slope (-18 dB to -24 dB) to avoid overloading and causing a considerable amount of distortion.
- In most cases, a high-pass filter for a woofer or subwoofer is sufficiently dimensioned with "-12 dB" and is only necessary when small loudspeaker systems are used.
- Take care when you choose a slope of just "-6 dB" in fully active systems, particularly with tweeters. Such filters shall only be used with a suitably selected crossover frequency.

B.4.4. Adjustment of the Q-Factor

This slide control is only active if you select the "Self-Define" characteristics. The slope for this characteristics setting is fixed at -12 dB and there is the option of adjusting the Q-factor of the high- or lowpass between 0.5 and 2.0 in increments of 0.1.

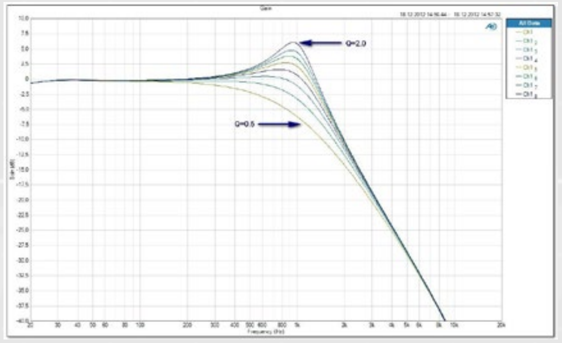
The Q-factor effect is illustrated below using the example of a high- and lowpass filter with a crossover frequency of 1000 Hz. The high-pass behaves analogously to the filter characteristic accordingly reversed.



Expert advice:

Adjusting an appropriate crossover frequency and slope is more important than the selection of „characteristic" or „Q factor".

The Q-factor effect is illustrated using the example of a highpass filter (above) and a lowpass filter (right) with a crossover frequency of 1000 Hz.

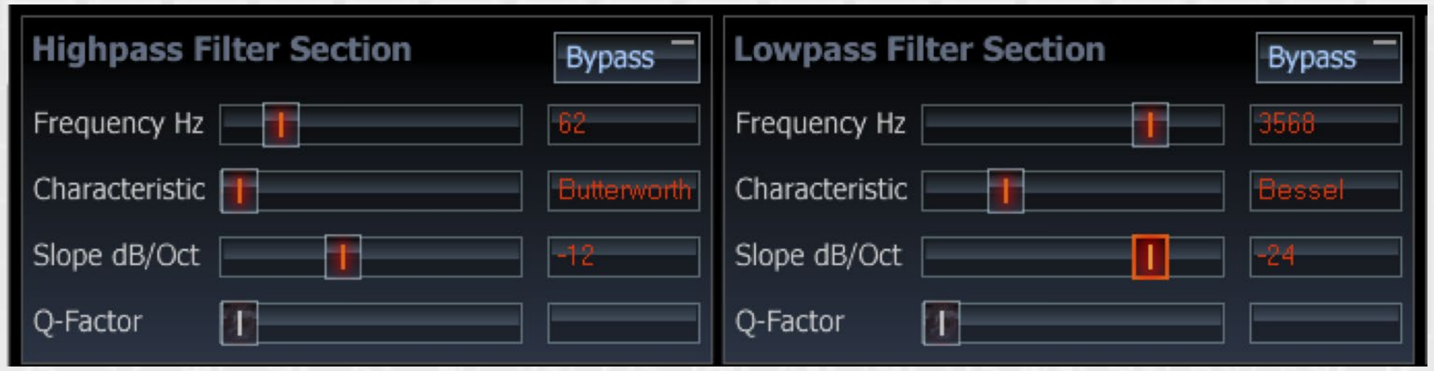


B.4.5. The bypass function of high- and lowpass filter

Use the bypass button to completely eliminate the effect of a high- or lowpass filter with a single push of a button to obtain a simple acoustic comparison "with and without filter".

Important information:

In fully active systems, do not simply bypass the tweeter highpass filter. Without any frequency crossover, irreparable damage may be caused even at low volumes! Bypassing a lowpass filter, on the other hand, is generally uncritical and will not result in any damage to the loudspeaker.



B.5. Output level

Use the gain controls or each channel only for fine tuning – the coarse adjustment has to be made on the connected amplifier itself.

B.5. Adjustment of output level

The output level of each of the channels may be adjusted using the "Output level" slide control.

Its increments depend on the position of the slider. The "–10" to "+5 dB" band has increments of 1 dB, the "–30 dB" to "-10 dB" band increments of 2 dB and that below "–30 dB" has increments of 6 dB.

The exact value is also indicated on the scale next to the slide control and as an absolute value above the slide control.

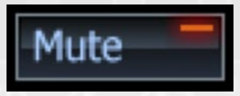
When adjusting the slide control, values greater than "0 dB" should be avoided, otherwise there is a risk of overloading the signal processor. Ensure that the red area of the level indicator is never visible at any volume. This may possibly occur if you have used the full boost of "+6 dB" in an equalizer frequency band, for example. The red bar in the level indicator is a clear sign that the signal processor is overloaded, which may cause a dramatic and sudden rise in the distortion factor. In contrast to an analog amplifier, which overloads with a rather smooth increase of distortion, digital stages are very "intolerant". Digital over modulation sounds extremely harsh and may quickly damage the connected loudspeaker (especially the tweeters). It is therefore preferable to set the quietest channel to "0 dB" and adjust all other channels downwards accordingly until the volume balance meets your requirements.

Each channel has its own gain control. Be careful with gain settings higher than 0dB – reduction are always uncritical (see left). The red area of the level control (see right) indicates overmodulation which should be avoided at any times.

The C-DSP has rotary volume controls for the input sensitivity – the output gain is controlled by the software of the "DSP PC-Tool".

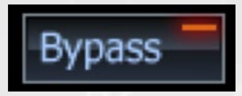
Muting an output channel („Mute")

The "Mute" button is used to mute the output of a channel. Clicking on this function again demutes the channel.



Bypassing the equalizer and the time alignment („Bypass")

Activating the "Bypass" button on the left next to the Output Level slide control bypasses the graphic 1/3 octave equalizer, the parametric filter (PP50 DSP only), the fine-EQ functions and the time alignment settings. The "Bypass" may be removed by clicking on the function for a second time. The orange lightning bar means: „Bypass is activated".



B.6. Phase and time alignment

A sensitive adjustment of the phase and time alignment is the basis for a natural and precise sound staging in your car.

B.6. Phase and time alignment

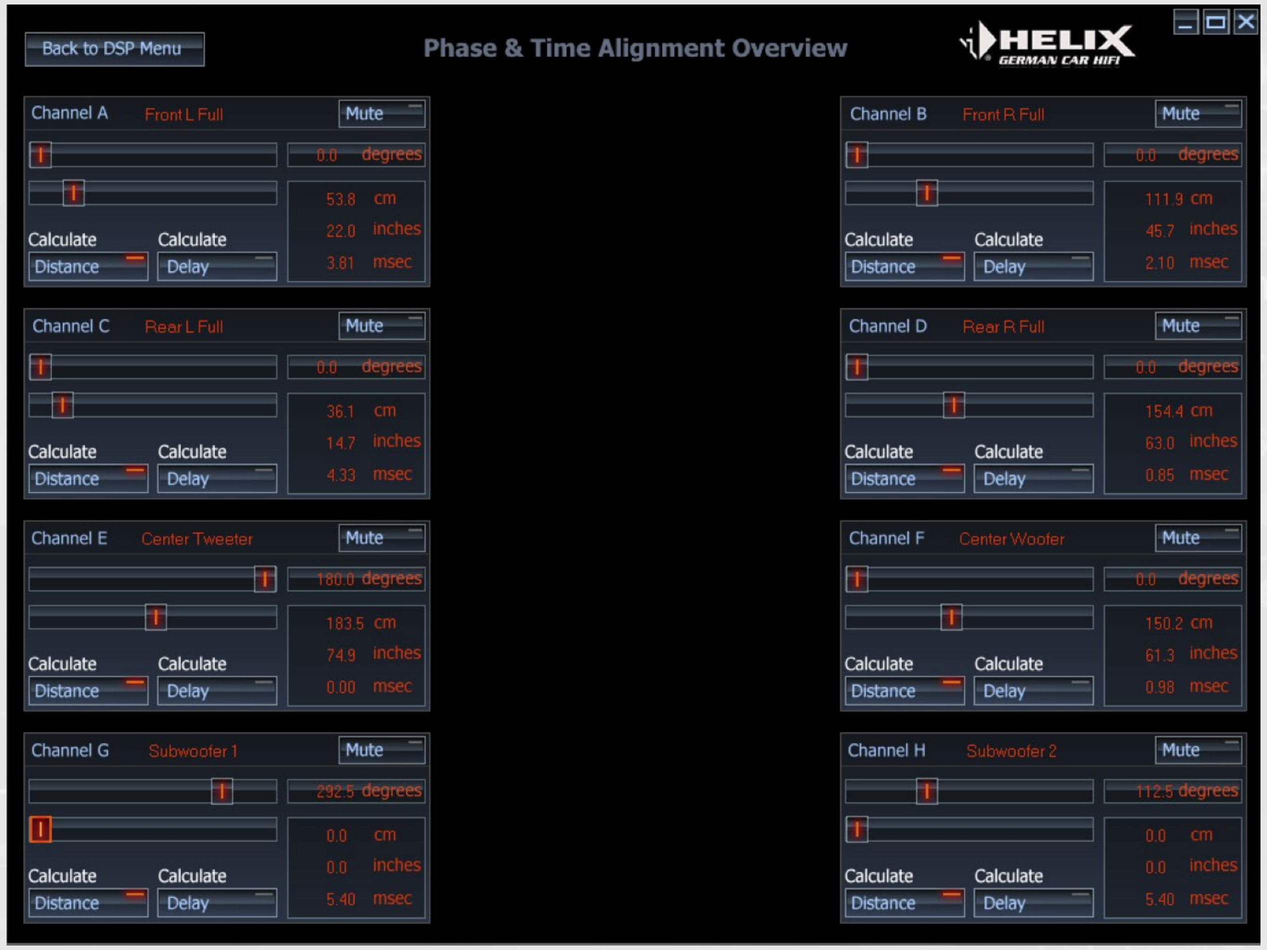
The window section for time alignment and phase adjustment shows all settings for the selected channel. Next to the two sliders you find the value for the phase adjustment in degrees as well as the delay time displayed in centimeters, inches and milliseconds.

The adjustment of phase and time alignment is also possible in the little window, but we recommend the Overview. So that you don't have to constantly switch backwards and forwards between the individual channels when adjusting the time alignment and phase, you can use the Overview button to display all the channel values simultaneously. This presents you with the display shown on the right page.

The phase and time alignment can either be adjusted for each channel individually in the small window below or by clicking on the „Overview“ button for all channels .

The Time Alignment Overview page („Overview“)

The time alignment overview displays the distance and delay values of all channels at a glance. Use the "Back to DSP Menu" button located in the top left hand corner to return to the previous screen.



B.6.1. Adjusting the phase

The phase of each of the front and rear channels can be switched between 0 and 180 degrees. Regulation of the subwoofer channel phase is particularly sensitive and may be adjusted between 0 and 360 degrees in increments of 22.5 degrees. This makes it possible to acoustically adapt the subwoofer precisely to the front and rear channels.

B.6.2. Adjusting the time alignment

It is only possible to achieve a correct "front staging" of the music if the time alignment is adjusted correctly. The software therefore allows sensitive adjustments in increments of 7 mm / 0.021msec. The software provides two different ways of adjusting the time alignment:

„Calculate Distance“

This mode requires the input of the distance between each loudspeaker and the ears of the listener. Simply use a tape measure to determine the distance. The program uses this to calculate the necessary time delay for each channel.

„Calculate Delay“

This mode requires the direct input of the preferred time delay value if available.

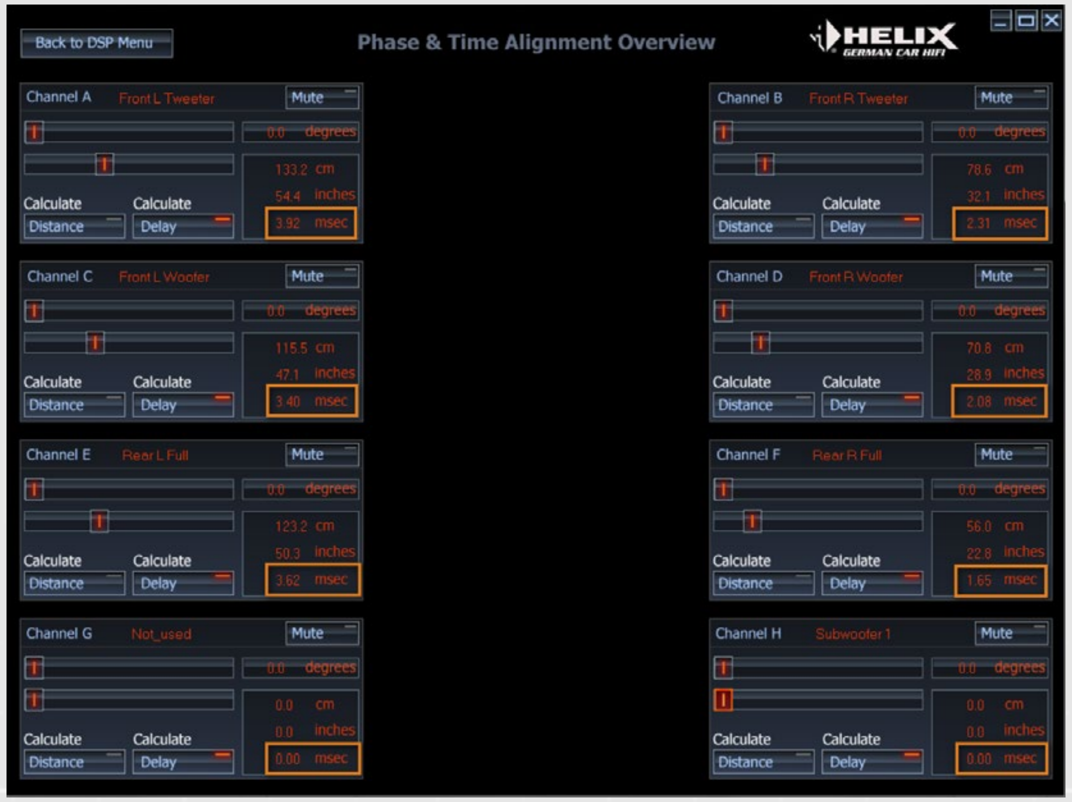
! Expert advice:

The velocity of sound equals to 340 meters per second, so that each millisecond (ms) corresponds to 34 centimeters. 0,1ms mean 3.4 cm and therefore 0.02ms correspond to 7mm.

Expert advise for adjusting the time alignment („Time Alignment“)

The optimum setting is most easily calculated using the "Calculate Distance" mode as follows:

1. Use a tape measure to measure in centimeters / inches the distance between the front left loudspeaker (Front Left ...) and your head in your sitting/listening position.
2. Adjust this distance value using the slide control.
3. Repeat these steps for each of the other channels. The program uses these distances to calculate the required time delay for each of the channels and displays these values accordingly in "msec". The loudspeaker furthest from the listening position (usually the subwoofer channel) will not be delayed at all, whereas the loudspeaker closest to the listener usually experiences the longest delay. It is not necessary to measure/adjust the values as accurately as possible as they merely serve as initial values for the fine adjustment.
4. Now mute the rear channels and the subwoofer using the "Mute" function.
5. Select a music track which preferably consists of only a voice and no accompanying instruments.
6. In small increments, adjust the time alignment of one of the front loudspeakers (Front left ... or Front right ...) until you can hear the voice either directly from the front, or slightly from the right (1 o'clock position). It is usually only necessary to adjust the time alignment by only a few centimeters to reach the desired result.
7. Now mute the front channels using the "Mute" function and reactivate the rear channels.
8. Listen to the same music track again and adjust the time alignment of the rear channels so that the voice can be heard from directly behind you, or slightly from the right (5 o'clock position).



! **Important**

The time alignment function may only be used to achieve optimum stereo reproduction for one single listening position in the vehicle. However, spatial reproduction in the other positions is usually even worse with time alignment.

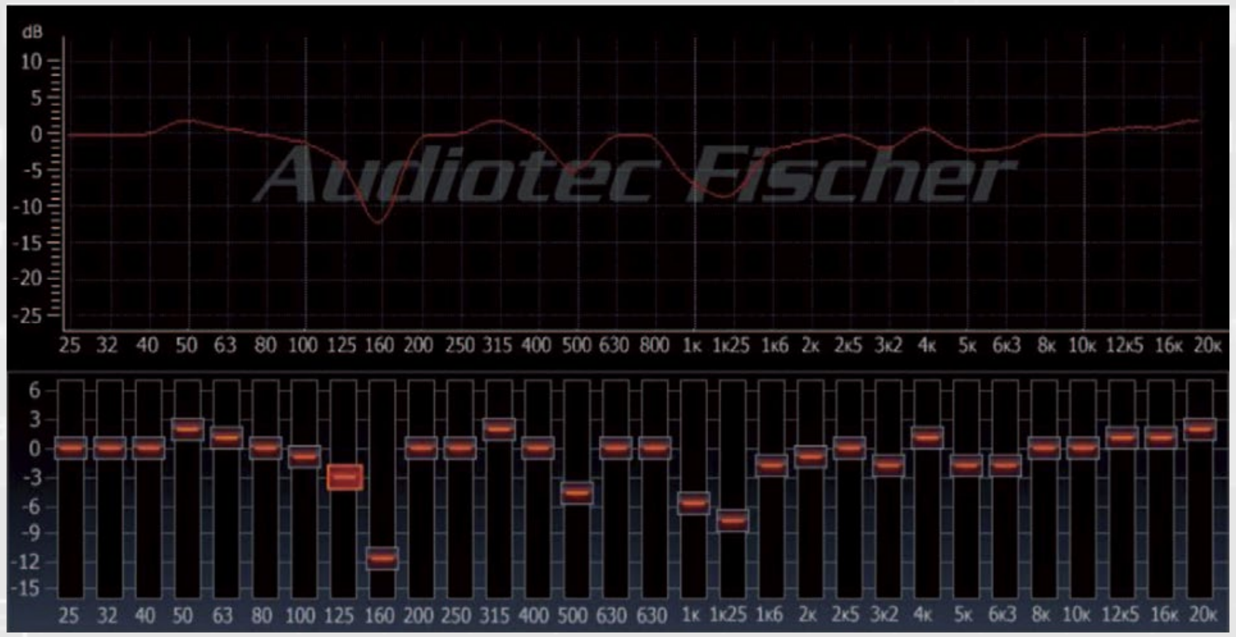
Always switch on the time alignment before you adjust the frequency response with the equalizer.

It is not usually necessary to adjust the time alignment of the subwoofer more accurate than using the "tape measure method". Using the "Phase" slider is often more effective in this respect.

9. Reactivate the front channels (keeping the subwoofer channel muted as before).
10. The amplifiers PP50 DSP, PP 52DSP and PP 82DSP also have the option of using the "Additional Delta Delay Front/Rear" slide control to shift the rear channels backwards until it is almost impossible to hear them or they no longer have any negative effect on a proper "front staging" of the front loudspeaker.
11. If you have a device without this feature, you have to manually enter an extra delay for each rear channel. According to experience values of approx. 100cm (corresponds to 3 msec. delay) are sufficient. If you have done the time alignment setup using the „Calculate Distance" method then it is not possible to add such an extra delay by simply using the sliders of the rear channels. This would affect all other channels. So you have to do the following: Take a note of the delay of each channel in "msec" switch to "Calculate Delay" method. Enter the delay for each channel in msec. using the individual sliders (upper picture). Now you can vary the values for the rear channels without affecting the other channels. In the example (lower picture) the two rear channels have been additionally delayed by 100.6cm (= 2.96 msec.).
12. If you have done the time alignment for the front and rear channels correctly, then demute all the other channels. This completes the adjustment of the time alignment.

B.7. The Equalizer

An equalizer is able to significantly improve the tonality of a car stereo system. The target of a properly adjusted EQ should be a smooth overall frequency response.



A graphic equalizer allows boosting or cutting specific frequency bands. Always try to cut the "peaks" in the measured frequency response instead of filling up the "dips". Strong boosts should be avoided at any time! The curve displayed above the slide control indicates the effect of the equalizer on the frequency response.

B.7.1. The graphic 1/3 octave equalizer

Each of the DSP channels has its own equalizer which affects the frequency response in high resolution 1/3-octave increments. The functionality of the EQ depends on the device itself and may be different for the individual channels:

- All devices have 30 slide controls in 1/3rd octave steps between 25 Hz and 20 kHz for the channels A to D.
- Channels that have been defined as "subwoofer channel" have 12 slide controls in 1/3rd octave steps between 25 Hz and 300 Hz.
- The channels E and F of NOX4 DSP and channels E to H of C-DSP respectively also have 30 controls in 1/3rd octave steps between 25 Hz and 20 kHz if defined as fullrange channels.
- The channels E to G of P-DSP and HELIX DSP have 30 controls in 1/3rd octave steps between 25 Hz and 20 kHz if defined as fullrange channels. Channel H is always a subwoofer channel with 12 controls in 1/3rd octave steps between 25 Hz and 300 Hz.
- The channels F and G of PP 52DSP and F to H od PP 82DSP have 15 controls in 2/3rd octave steps between 25 Hz and 16 kHz.

Each band of the equalizer allows a maximum boost of "+6 dB" and a maximum cut of "-15 dB". This asymmetrical design was chosen because boost levels of more than "+6 dB" are rarely required. Any deep, narrow band dips measured in the frequency response are usually due to signal cancellations caused by phase shifts which may not be eliminated by the equalizer, even with greater boost levels. Experience shows that more attention should be paid to the elimination of peaks in the frequency response when adjusting it. The human hearing is considerably more sensitive to narrow peaks than to narrow frequency dips. You can read more about adjusting the equalizer in the chapter entitled "C Adjustments". There you'll find more tips about the practical application of the equalizer function.

! Expert advise:

For best results it's neccessary to use a measurement system. You'll find further informations in the chapter „C.1. Adjustment guide" on page 39.

B.7.2. Further equalizer features

The feature set of the equalizer section strongly depends on the DSP device. The table below describes the different feature sets in detail:

DSP device	Equalizer functions in detail
PP50 DSP	1 additional parametric filter per channel 0 – -30dB, center frequency in 1/6th octave steps (subwoofer: 1/12th octave steps), Q factor adjustable from 0,5 to 15
P-DSP HELIX DSP NOX 4 DSP PP 52DSP PP 82DSP	FineEQ = fine adjustment of all graphic EQ bands, center frequency in 1/24th octave steps and Q factor adjustable from 0,5 to 15 (overlapping of center frequencies not possible!)
C-DSP	Para-graphical EQ = fine adjustment of all graphic EQ bands with free selection of center frequencies in 1/24th octave steps (overlapping of center frequencies possible), Q factor adjustable from 0,5 to 15
PP 52DSP PP 82DSP	Channels F–G (PP 52DSP) and channels E–H (PP 82DSP) respectively only 15 EQ bands in 2/3rd octave steps; all can be tuned with FineEQ in 1/12th octave steps and Q factors between 0,5 and 15

B.7.2.1. The parametric filter (PP50 DSP only)

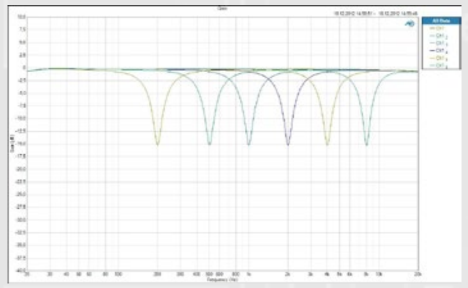
In addition the graphic 1/3rd octave equalizer, the PP50 DSP also has one parametric filter for each channel which may be used to make even finer acoustic adjustments.

This filter is intended exclusively for compensating peaks in the frequency response (range: 0 to -30 dB). It does not allow any boosts and therefore cannot eliminate any dips in the frequency response.

A parametric filter is typically used for a narrow band resonance of a door panel, for example, which cannot be eliminated with the 1/3rd octave equalizer.

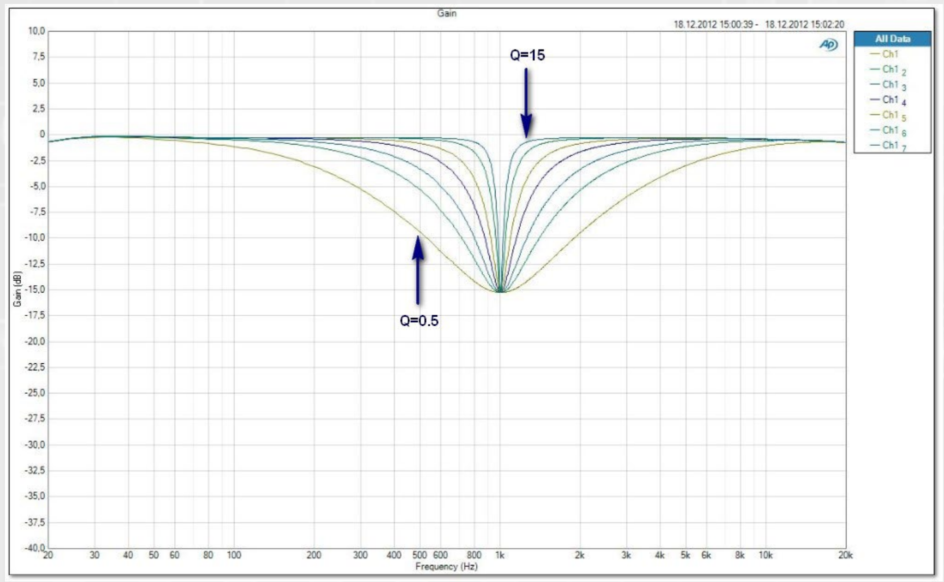
The center frequency
The center frequency of the parametric filter may be adjusted in 1/12-octave increments; the corresponding value is displayed above the slide control. The upper diagram on the right illustrates the effect on the frequency response.

The „Q"-factor
The "Q" slide control is used to adjust the bandwidth of the frequency cut. A low Q-factor equates to a broad band reduction, while a high Q-value equates to a particularly narrow band cut. The effect of the "Q" control is illustrated in the lower diagram.



The graph shows the effect of different center frequencies of a parametric filter. In this case the Q factor and the gain remain unchanged.

The effect of the „Q" of the parametric filter



The graph clearly shows the effect of different Q factors of a parametric filter. In this case the center frequency (here 1 kHz) and the gain (here -15dB) remain unchanged.

B.7.2.2. Graphic equalizer fine adjustment function „Fine EQ"
(P-DSP / HELIX DSP / NOX 4DSP / PP 52DSP / PP 82DSP only)

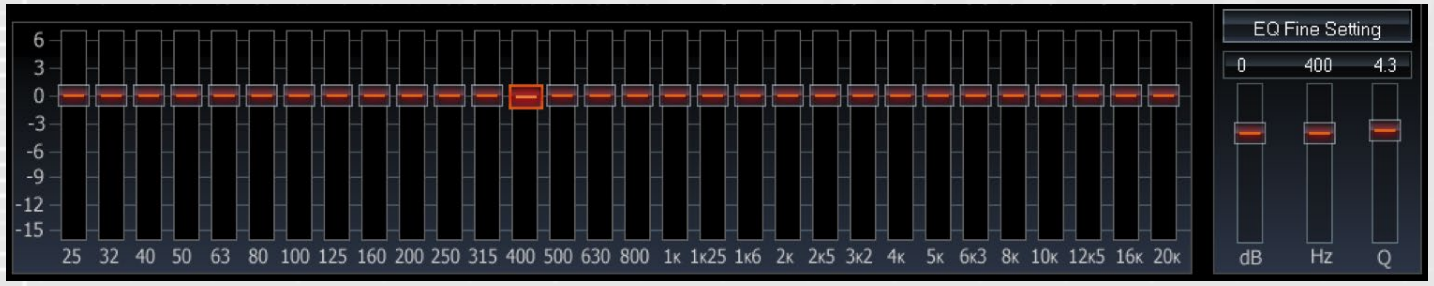
The FineEQ function is used to precisely adjust the 1/3 octave equalizer and provides a similar function to the parametric filter of the PP50 DSP. The FineEQ function makes it possible to adjust each of the 30 bands of each channel even more accurately. Therefore, both the center frequency and the Q-factor of each band may be adjusted with great precision. To finely adjust the selected band, click with the mouse on the level control of the specific band. This should turn red. On the right hand side in the "EQ Fine Setting" area, you can now adjust the center frequency of the band up and down. This adjustment is made in 1/24th octave increments.

Anyhow, an overlapping of frequency bands is not possible. Here's an example:

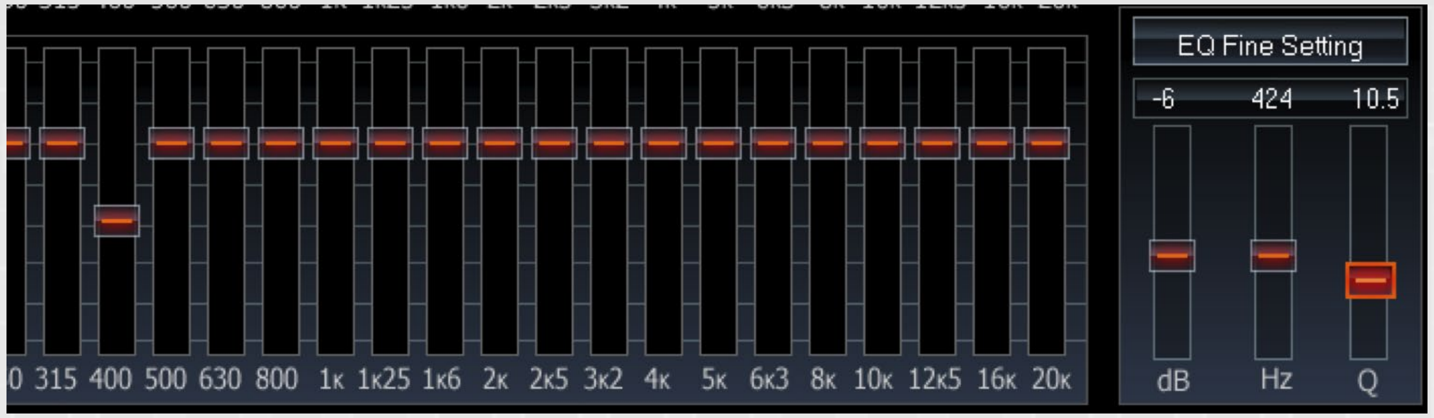
The center frequency of the 1kHz band can be raised with the "Fine setting" up to 1150 Hz. The next "1.25kHz" band can be lowered to 1175 Hz only. So it is not possible to set two EQ bands to exactly the same center frequency.

The quality (Q-factor) of each band may also be adjusted. The default value for each band is 4.3 which corresponds to 1/3rd octave bandwidth. With the right slide control you can lower or increase the bandwidth of each EQ band (range 0.5 to 15). The effect of the "Q" slide control is identical to the parametric EQ of the PP50 DSP (see chapter 7.2.1. for explanation and graphs).

Example for fine adjustment of the 400 Hz EQ band



Step 1: Choose the band you want to fine adjust by left clicking the slider. In this case we choose the 400 Hz band. The slider will be highlighted as shown on the picture.



Step 2: The chosen band can now be fine adjusted with extra sliders on the right side of the EQ. This includes adjusting the center frequency and the Q-factor.

B.7.2.3. The para-graphical Equalizer (C-DSP only)

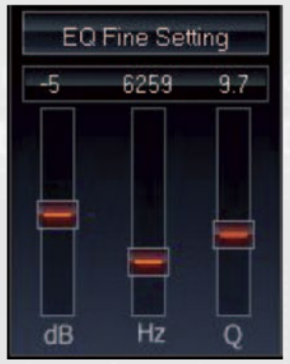
The C-DSP also has a fine adjustment function for each individual band in the 1/3 octave equalizer. Its functionality is similar to that described before although the setting options are even more comprehensive.

This makes it possible to adjust the center frequencies of each of the bands of the C-DSP from 20 Hz to 20 kHz by increments of 1/24th octaves. This means that each band can be set to a frequency within the complete frequency range. This function therefore offers almost the same setting options as a purely parametric equalizer.

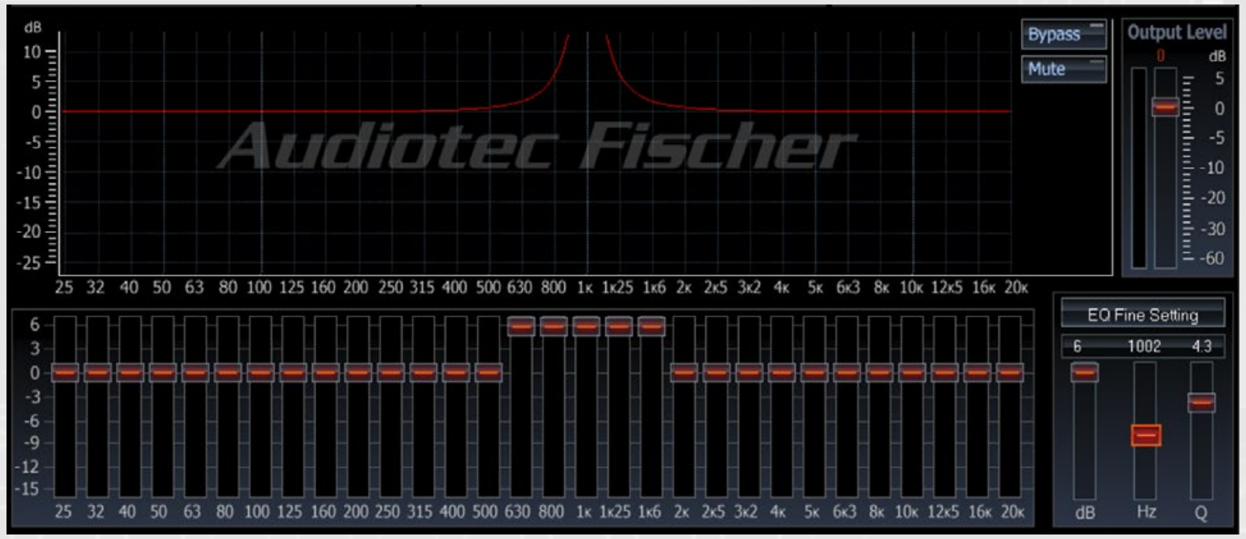
Important notes

First of all this functionality allows using the EQ bands which could normally not be used. That means for example that you can use the lower frequency bands of a tweeter channel also for the higher frequency range instead of not being used. So you can configure e.g. the 25 Hz band to 10 kHz to optimize the frequency response in this range.

Please make sure not to choose the same center frequency for more than one band if you want to boost a specific frequency range. The following image shows what may happen, if you set four bands to the same center frequency with maximum boost. Such settings will automatically lead to an overload of the DSP and severe digital distortion. Chances are high that you destroy your loudspeakers within seconds. In your own interest, please use this function very carefully!



The para-graphical equalizer of the C-DSP is extremely flexible and allows adjustments that may even lead to problems. Especially if you adjust more than one band to a specific center frequency and boost all these bands at the same time.



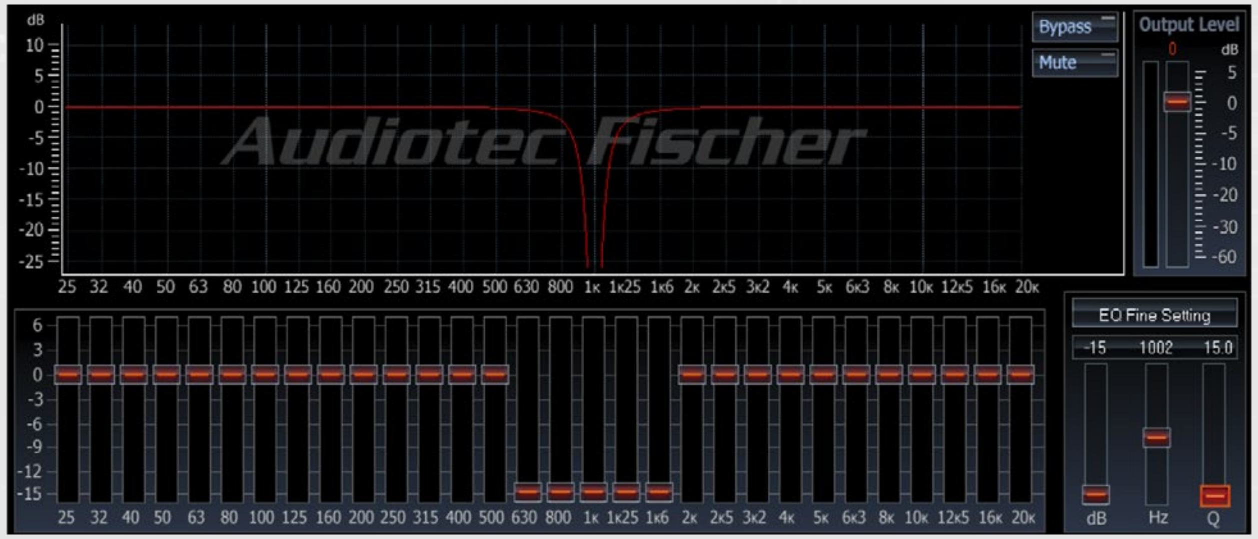
Warning

Make sure not to choose the same center frequency for more than one band if you want to boost a specific frequency range. Such settings will automatically lead to an overload of the DSP and severe digital distortion. Chances are high that you destroy your loudspeakers within seconds!

Please always check the level indicator in the output level section. The level indicator has to be always below the red area which will be shown if your channel is boosted too much. The red area should never be visible at any output volume because it may result in digital distortion.

continue: The para-graphical Equalizer (C-DSP only)

On the other hand it isn't critical if you choose the same center frequency for several bands to cut a specific frequency range. If multiple bands are set to the same center frequency, it will be possible to realize stronger attenuations than only one EQ band can do as the attenuations will be added. In opposite to added boosts of several EQ bands this has no negative impact and might make sense if you want to eliminate narrow band vibration or resonance problems of your speakers or door panels that cannot be solved mechanically.



Setting more than one EQ band to the same center frequency allows stronger attenuations without any negative impact. The attenuations of each band will be added.

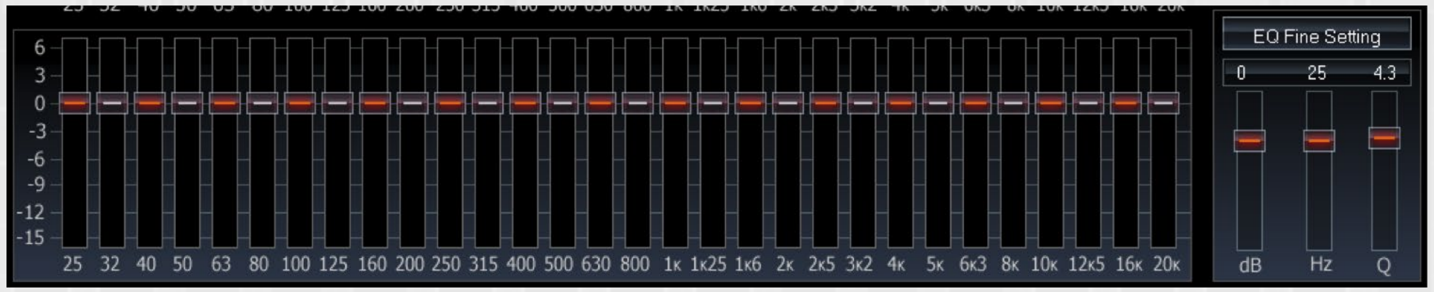
B.7.2.4. The graphical 2/3rd octave Equalizer (PP 52DSP and PP 82DSP only)

The channels E to H of the PP 82DSP and the channels F to G of the PP 52DSP respectively have 15 frequency bands in 2/3rd octave steps (as long as those have been defined as full-range channels in the channel routing menu).

Each band can be precisely fine-tuned using the „Fine EQ" functionality which already has been described in detail before.

Example:

The center frequency of the 1 kHz band can be raised in 1/12th octave steps up to 1330 Hz. The center frequency of the next 1.6 kHz band can be lowered down to 1424 Hz, so that even 15 EQ bands are sufficient to cover the whole frequency range from 20 Hz to 20,000 Hz. Nevertheless it is not possible to set two bands to exactly the same center frequency. Additionally it is possible to vary the Q factor of each band. The standard value is Q = 4.3 which can be changed in the range from 0.5 to 15. Lower values correspond to broader effects on the frequency response whereas higher values correspond to more narrow effects. For further details and illustrations please refer to the chapter where the parametric EQ of the PP50 DSP is explained.



All 15 bands of the 2/3rd octave graphic equalizer can be adjusted almost like a parametric EQ using the „Fine EQ" setting.

C. The correct adjustment of the DSPs

The real potential of a DSP can only be exploited to the full extent, if it will be adjusted reasonably and precisely. Here you can read several useful hints.

To optimally adjust setting of the equalizer using the "ATF DSP PC-Tool" it is absolutely necessary to be able to measure the frequency responses of the loudspeakers in your vehicle. Even absolute professionals are unable to make perfect adjustments using their hearing alone.

Thankfully, good measuring equipment is affordable and is definitely worth the investment. As an alternative several dealers offer a complete DSP setup service, if you don't rely on your own skills.

C.1. Adjustment guide

1. Make sure that all sound controls of your car radio are set to either "linear" or the center position. Deactivate any existing loudness functions. The balance and fade controllers must be adjusted to the "center position" as well.

2. Connect your amplifier / processor with your PC and start the software.

3. Start with the setup of channel routing matrix first (except PP50 DSP). On page 20 and in the appendix you will find a couple of examples for typical applications.

4. Ensure that you have correctly adjusted the time alignment first before you start the measurement. A detailed description for that can be found on page 32/33.

5. Mute all channels except the both front channels, if you don't want to process an individual equalization for each channel (not recommended). Don't forget to link the both front channels. More information you'll see on page 29.

6. Sit in the driver's seat with the measuring microphone. It is best to place your PC (Notebook/Netbook) on the passenger seat so that you can use it easily. Ensure that the screen is not positioned directly in front of the loudspeaker on the passenger side as this can influence the readings.

7. Start the "pink noise" playback on your car radio (activate the "repeat" function for that track if your car radio offers this feature). Adjust the volume so that any ambient noise is fully masked. Ideally, the vehicle will be in an enclosed garage while you are taking the measurements so that traffic noise does not affect the readings, for example.

8. Begin by measuring the front loudspeaker system, i.e. initially mute the rear loudspeaker and any connected subwoofer (e.g. using the ATF DSP PC-Tool Mute function). Hold the measuring microphone upright and move it slowly in a semicircle between your left and right ears. To start the measurement on your PC, press the F12 function key. "Praxis" will now take the readings for approximately 18 seconds. You can then identify the determined frequency response together with the red reference curve on your PC. See next page (p. 40).

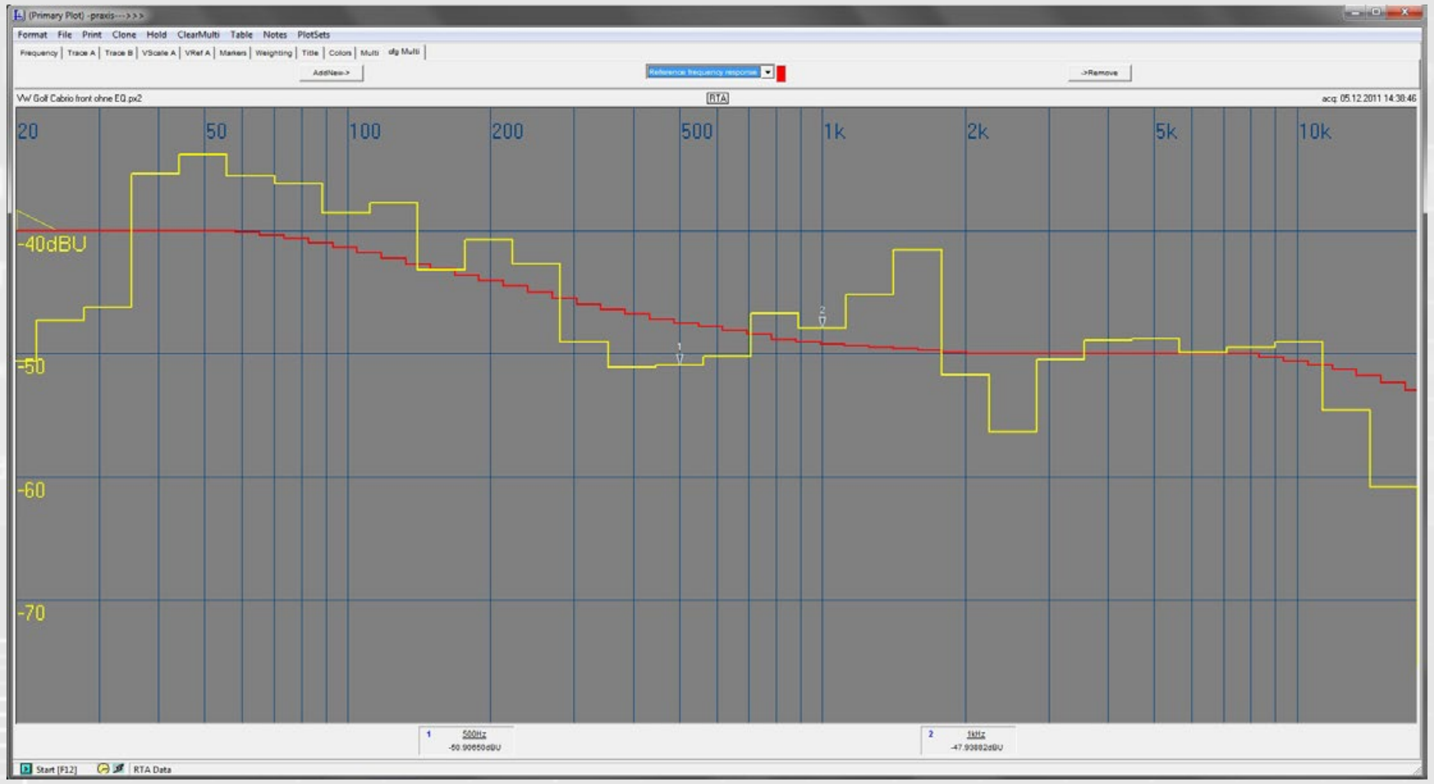
9. First choose appropriate setting for the highpass filters of the front channels. Especially in combination with a separate subwoofer it makes sense to relieve the speakers from low frequency response. This will have a positive impact on distortion, clarity of sound, power handling and maximum SPL. The following settings for the individual speaker sizes have been determined to be feasible:

Speaker	Highpass filter	
6x9" or 20 cm / 8"	40 Hz	-12 dB/oct.
16,5cm / 6.5"	50 Hz	-12 – -24 dB/oct.
13 cm / 5,25"	70 Hz	-18 – -24 dB/oct.
10 cm / 4"	80–100 Hz	-18 – -24 dB/oct.

10. Now use the DSP PC-Tool equalizer to adjust the frequency response so that it matches the reference curve as closely as possible. This is an iterative process, whereby the equalizer is first adjusted before further measurements are taken. However, you will see that the desired result is usually achieved after just a few attempts. Bear in mind that the "Praxis" measurement software is set up so that the frequency response is measured in the same 1/3 octave intervals with the same center frequencies you find in the DSP PC-Tool equalizer.

11. When adjusting the frequency response, avoid large boosts of individual frequency bands; large cuts are no problems, however. Do not attempt to equalize deep narrow band dips in the →

Continue: Adjustment guide



For a proper setup of a DSP a tool for measuring frequency response curves is mandatory. Audiotec Fischer recommends the software „Praxis" from Liberty Instruments, which is available free of charge (see also the expert advice on page 6). The next version of the DSP PC-Tool software will comprehend a proprietary and very easy-to-use measurement tool as well.

(continue)

→ frequency response. Such dips are predominantly caused by out-of-phase signal cancellations and it is not possible to sufficiently compensate for them. The human ear does not usually perceive these signal cancellations as disturbances.

12. Strong narrow peaks in measured frequency responses should, on the other hand, be completely eliminated as the ear is very sensitive to this. If the graphic equalizer is insufficient for this function, please use the additional parametric filter (PP50 DSP) or the fine adjustment function (C-DSP/P-DSP/HELIX DSP/NOX 4DSP/PP 52DSP and PP 82DSP).

13. Once you have corrected the front channels, mute them and activate the rear channels (keep the subwoofer muted as before). Here again don't forget to link the two rear channels if you don't adjust them separately. Now perform the measurement process as described in the same way for the rear loudspeakers. There are different approaches, however:

a) You also measure the rear loudspeakers from the driver's seat. This method is only recommended if the vehicle has just two seats, or if you usually only travel alone. The rear loudspeakers usually sound very unpleasant in the rear seats if they have been optimized for the front listening position.

b) Alternatively, in vehicles with four or more seats, sit in the middle in the rear to take the readings for the rear loudspeakers and adjust the settings to achieve a pleasant sound on the rear seats. Audiotec Fischer uses this procedure to create the vehicle-specific setups for the MATCH amplifiers.

14. Now reactivate all the loudspeakers (subwoofer channel remains muted) and take another frequency response reading. Normally it is necessary to do some fine adjustment of the EQ's when front and rear speakers are playing together. If you detect strong differences in the low frequency response then probably you notice a strong phase shift. It might be necessary to switch the phase of either the front or the rear speakers. In some cases it might even help to slightly adjust the highpass crossover frequency or the highpass slope.

15. The next step is to optimally adjust the subwoofer to the front/rear loudspeakers. So demute the subwoofer first and choose a start value for the lowpass filter. We recommend starting with a crossover frequency of 80 to 90 Hz and a slope of -24 dB/oct. (Butterworth characteristic).

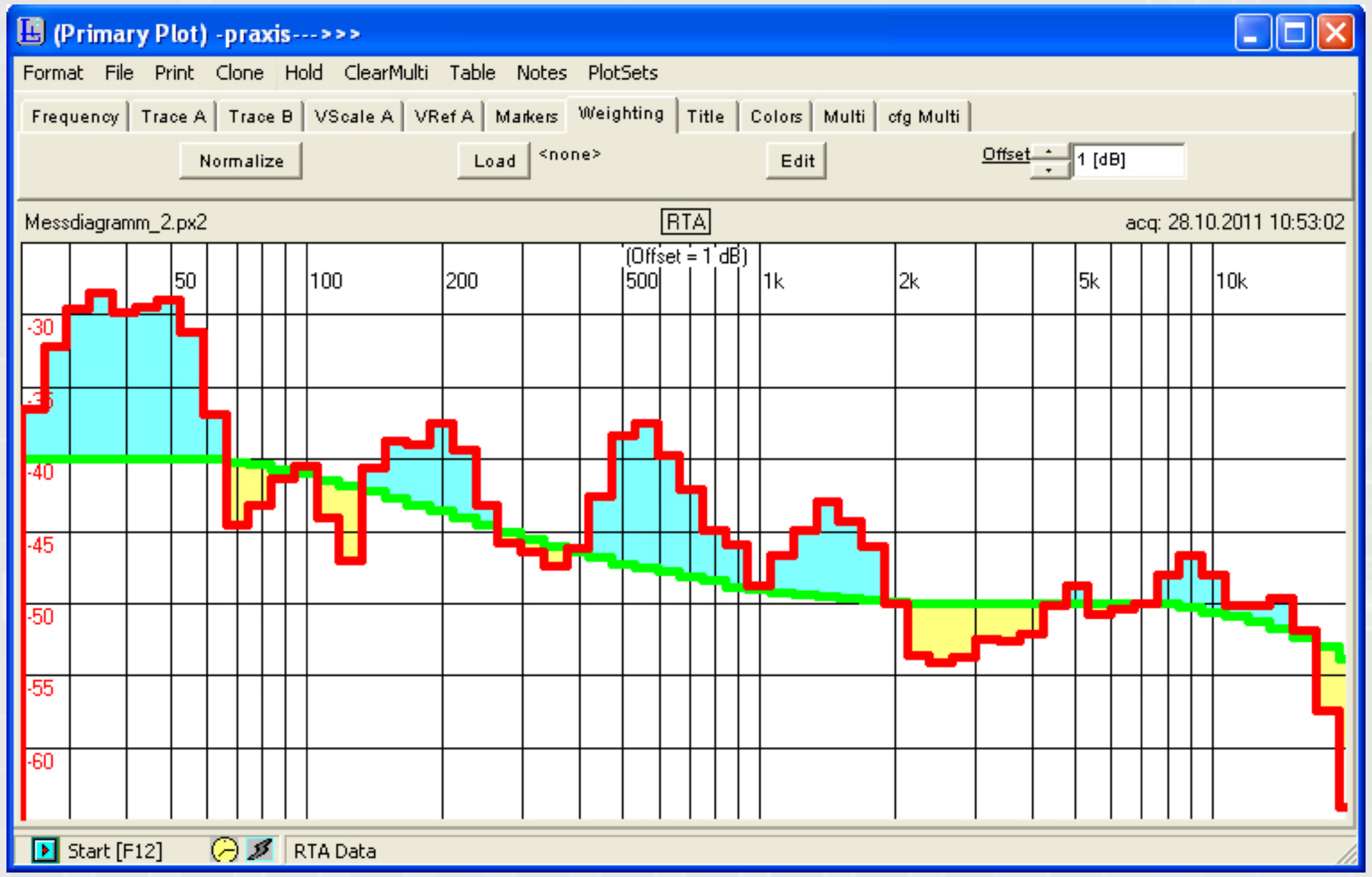
16. If you use a subwoofer from the MATCH series, then make sure that you set the highpass filter of the subwoofer channel accordingly to the parameter shown on page 25.

17. Make another frequency response measurement and vary the phase and output level of the subwoofer channel so that you achieve a smooth transition in the frequency response from subwoofer to the other speakers, even without the use of the EQ. A strong dip in the range of the crossover frequency indicates inadequate phase relationship between the subwoofer and the other channels. Here again it might also help to slightly vary the crossover frequency and/or slope of either the subwoofer or the other speakers. If you don't achieve a flat frequency response then finally try to compensate the flaws using the EQ of the subwoofer channel.

18. Now –after all these measurements and adjustments- do a first acoustical check using music tracks that you know "by heart". Adjusting the DSP in the way as described before usually leads to excellent acoustical results. Nevertheless for the final sound tuning you should rely on your ears.

19. After all settings are done you have to press the button <Store DSP> to transfer all parameters to the amplifier's/processor's internal memory. Note: This step is mandatory – if you skip it, your data will get lost after turning the device off!

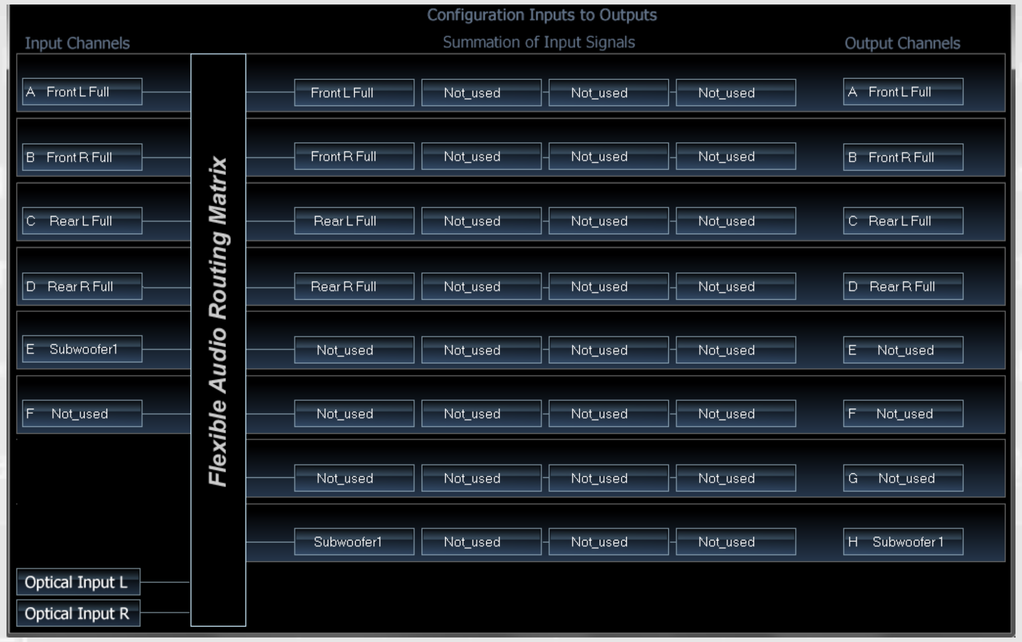
20. Important: Store your setup as an „.afp"-file on your PC as well, using the button <Save> (mouse click left). If you want to share your setup with others who want to upload it to the device via microSD card then first you have to store it as an „ac1" or „ac2" file (right mouse click on button <Save>).



The measurement plot shows the deviation of the speakers' frequency response (red curve) from the desired reference/target response (green curve). The light blue areas need to be cut with the equalizer whereas the yellow areas have to be boosted carefully.

D.1. Adjustment examples of routing channels

a) Example for P-DSP / HELIX DSP



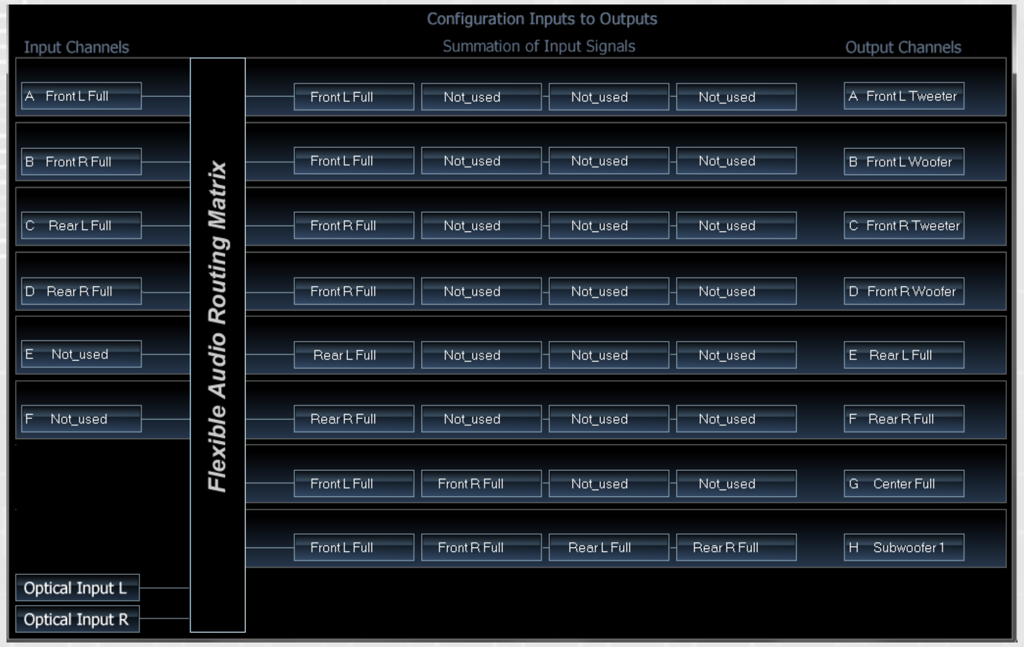
Configuration

Radio
Radio with 5-channel preamp output (Front Left, Front Right, Rear Left, Rear Right, Subwoofer)

Speaker
Speaker configuration: fullrange for "Front Left", "Front Right", "Rear Left" and "Rear Right" plus a separate subwoofer.

The car radio drives the five inputs A-E of the DSP. These inputs will be straightly routed to the outputs A-D + H. This is why you can find in the routing matrix only one edited input field per output channel.

b) Example for P-DSP / HELIX DSP



Configuration

Radio
Car radio with 4-channel preamp output or highlevel speaker output (Front Left, Front Right, Rear Left, Rear Right)

Speaker
Speaker configuration: fully active 2-way system for "Front Left" and "Front Right", fullrange speakers for "Rear Left" and "Rear Right", fullrange center speaker plus a separate subwoofer.

The input A/B for the front speaker system will be routed to two output pairs A/B and C/D whereas the input C/D for the rear speakers will be routed to the outputs E/F. The fullrange center speaker (output G) receives a sum signal from both front input channels A/B and finally the subwoofer will be fed with a sum signal from all four inputs.

c) Example for C-DSP

Configuration

Radio
Signal source with digital optical output Left / Right.

Speaker
Speaker configuration: fully active 3-way system for „Front Left" and "Front Right" plus two separate subwoofers.

The car radio feeds the optical digital input of the device. The stereo signal will be distributed to all six outputs A – F for the front speaker system; the two subwoofers (outputs G+) get the sum signal of the left and right input channels.

d) Example for PP 52DSP

Configuration

Radio
Car radio with 4-channel highlevel speaker output (Front Left, Front Right, Rear Left, Rear Right).

Speaker
An additionally connected power amplifier to „LineOut Sub A" for a separate subwoofer; perhaps in combination with an additional center speaker on "LineOut B".

The PP 52DSP is a 5-channel amplifier with an additional stereo preamp output. The internal output stages are driving the front and rear speakers and one of the dedicated MATCH subwoofers whereas the preamp output can be used for other separate subwoofers or center speakers in combination with another power amplifier.

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