

DSP Special Vol.2 – the Sound Tuning Magazine from Audiotec Fischer
incl. operation and adjustment manual for digital sound processors

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Sound Tuning Magazine
For DSP PC-Tool V3

DSP
Special

Currently supported products



- NOX4 DSP



- DSP PRO
- DSP
- P SIX DSP MK1 & MK2
- V EIGHT DSP



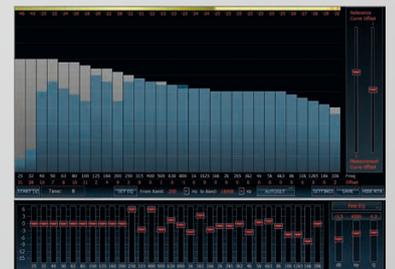
- M 5DSP
- PP 52DSP
- PP 62DSP
- PP 82DSP
- DSP

TOPICS

DSP PC-Tool Version 3



Real-Time Analyzer



Equalizing



The path to a better sound

- Tips and tricks for operating with the DSP PC-Tool Version 3
- The new features of the ATF Real-Time Analyzer
- Including a detailed DSP setup guide

Vol.2

DSP Special Vol.2



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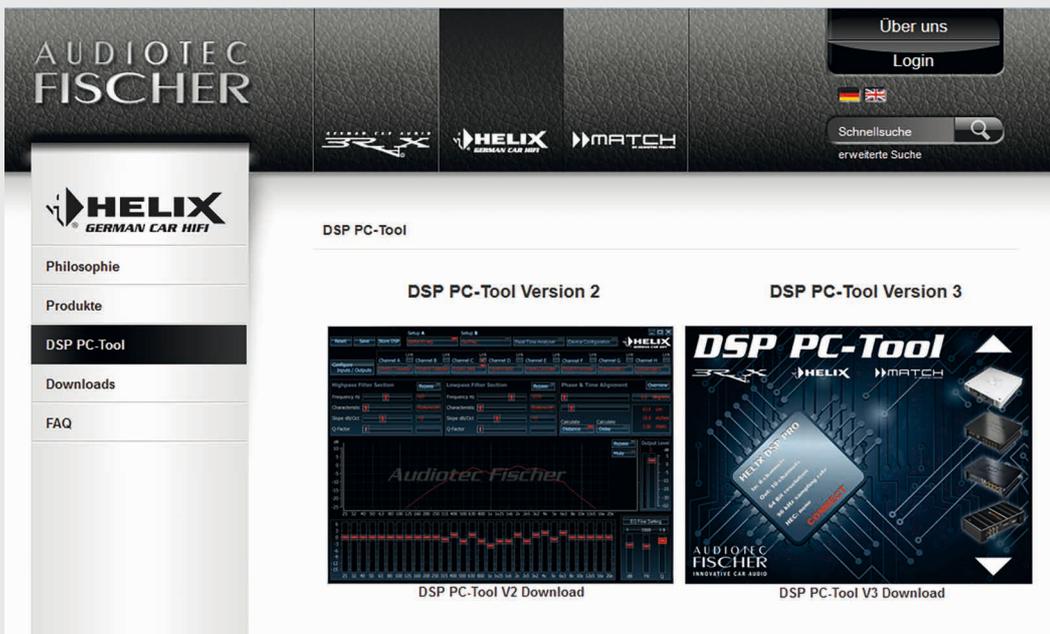
1 General – 1.1. Installing the software

1. General

1.1. Installing the ATF DSP PC-Tool software

The installation of the ATF DSP PC-Tool software is a basic requirement before you can start 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.



The software “ATF DSP PC-Tool Version 3” including the required USB drivers can be downloaded as a complete installation file from the Audiotec Fischer website.

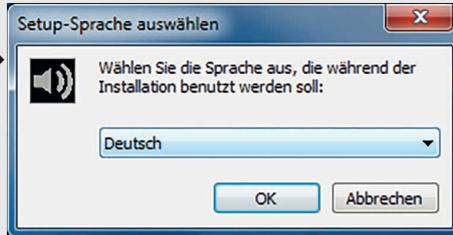
1. Download the installer file “Setup_ATF-DSP-PC-Tool_3.XX.exe” from the Audiotec Fischer website (www.audiotec-fischer.com). This file can be found in the “DSP PC-Tool area”. Save this file on the hard disk of your computer and start the installer file with a double-click. Depending on your operating system you will see the security advice (picture ①). Double-click on the “Execute” button for confirmation. 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.
2. 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 DSP PC-Tool software itself.
3. Now the setup assistant will start – to continue please double-click the respective button (picture ③).
4. Make sure that you carefully read the information in the following window (picture ④). After that you have to confirm this.
5. 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).
6. Now choose a name for the shortcut in your start menu. Here again the installer will make a proposal which you can modify. Otherwise simply confirm to continue (picture ⑥).
7. If you like to have a shortcut icon on your desktop for easiest access of the program, then checkbox the appropriate position (picture ⑦).
8. 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 / ⑨).

1 General – 1.1. Installing the software

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



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



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



3 A click on "Continue" in the "Welcome" window will start the setup assistant.



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



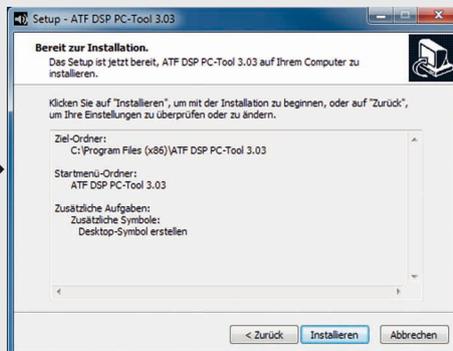
5 The target folder is already predefined; nevertheless it can be modified manually.



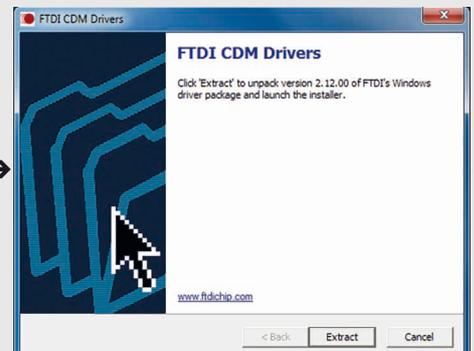
6 Choose your "start menu" folder here.



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



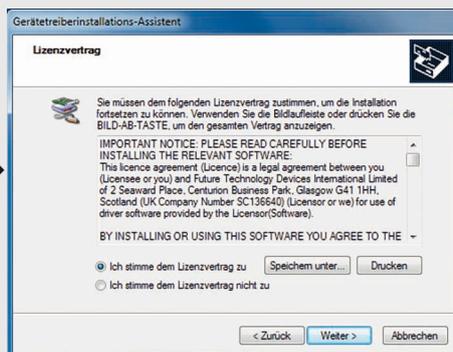
8 Here you can see all chosen parameters at a glance for checking purposes.



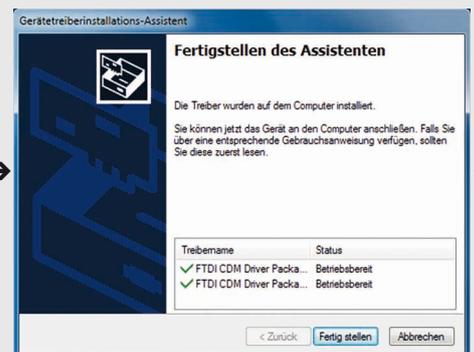
9 To start the setup assistant of the drivers click on the "Extract" button.



10 Click on "Continue" to proceed the installation process of the drivers.



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



12 A click on "Continue" will finalize the driver installation process.

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1 General – 1.1. Installing the software / 1.2. System requirements

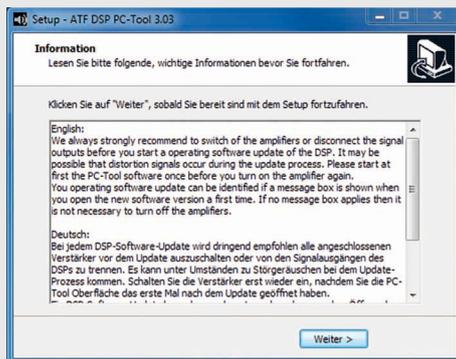
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■ Important

Carefully read the security advices in the following window (picture 13) 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 at a later date then remove the tick in the checkbox (picture 14). Click "finish" for completing the installation procedure.



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



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

1.2. 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 higher

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).

Device system requirements:

- You require the USB cable that was supplied with your DSP device
- The most recent version of the firmware has to be installed in the device. If the firmware is not up-to-date the DSP PC-Tool software will show you a message which allows an automatically update

Technology excursion

Not every DSP is the same – the overall concept makes the difference

To achieve the perfect sound quality is not easy. It is the optimization of every single detail without any compromise. Choosing the right main components inside an amplifier or a signal processor is certainly of high importance but often the tiny technical details differentiate Audiotec Fischer products from the competitors.

- Sophisticated, low-noise power supplies with independent regulators for the analog and digital stages. For example a Helix DSP PRO contains not less than nine regulators to all different internal supply voltages

- All internal voltages operate independently of the on-board voltage within the range of 6 - 17 Volts
- Complex additional filtering of all power and signal lines with EMI-filters
- Low-impedance PCB designs in 4-layer or even 6-layer technology with at least one separate ground layer
- Strict separation of analog and digital signals on separate layers of the circuit board
- Shortest possible signal routing with direct "Input-to-Output" signal flow
- Separate micro controllers for all "non-audio" processes

1.3. Preparing the measuring microphone

Basis for a successful measurement is an accurately installed measuring microphone.

Before you can do the first measurement you have to connect the measuring microphone and its

preamplifier to your computer. In this case the MTK 1 serves as an example

for connecting a measurement microphone to a PC.

1. Connect the XLR plug to the measuring microphone:



2. Connect the XLR jack to the microphone preamplifier:



3. Hook up the miniUSB plug of the included USB cable to the microphone preamplifier:



4. Connect the USB plug to a free USB port of your computer:



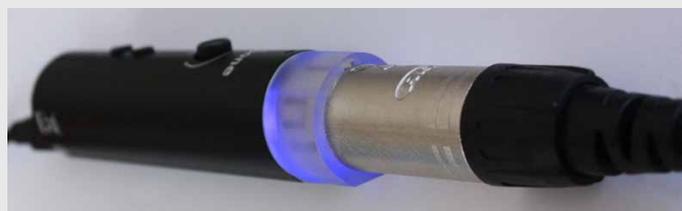
Note

We strongly recommend not to hook up directly the microphone preamplifier to the microphone!

5. Set the gain control on the microphone preamplifier to center position:



6. After connecting the microphone preamplifier to your computer the transparent ring should light up blue:



7. Now activate the phantom power of the microphone amplifier by pressing the tiny push button once (marked with "48V"):

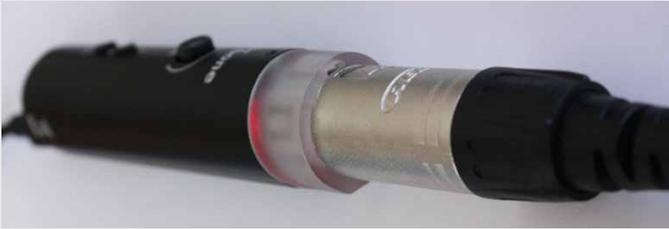


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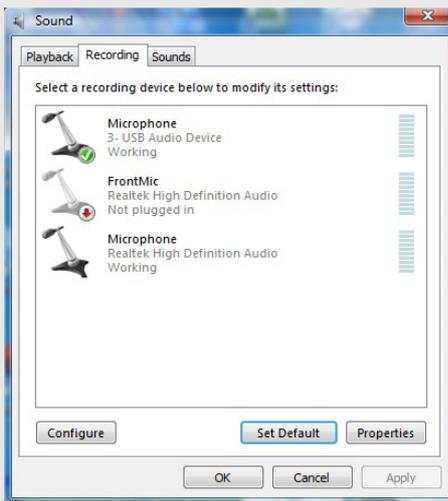
1 General – 1.3. Preparing the microphone

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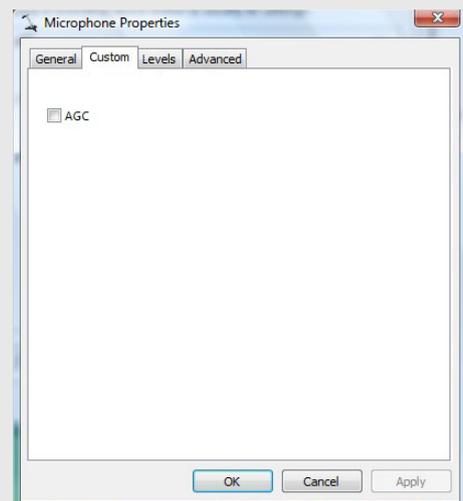
- **8.** The transparent ring should now change its colour from blue to red:



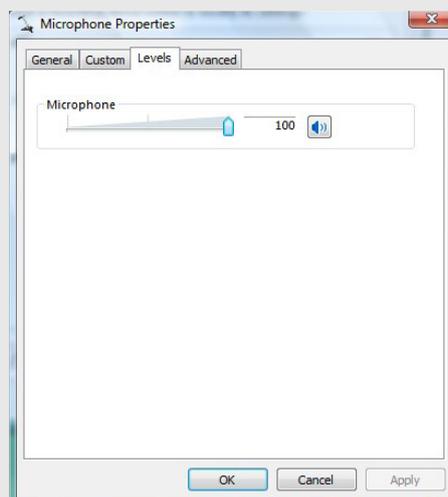
- 9.** Please also check the audio settings in the control panel of your computer whether the microphone has been properly detected.



- 10.** It is important that the automatic gain control "AGC" is deactivated.



- 11.** Make sure that the microphone volume is set to 100%.



Important

It is mandatory that the microphone is connected to the computer before you start the audio analyzer in the DSP PC-Tool. The microphone will not be detected if you connect it after starting the software!

1.4. The test signal

The measurement of frequency response curves with the ATF audio analyzer requires an appropriate test signal. In this case a so-called "pink noise" signal is used which has to be reproduced through your car audio system. The included USB memory stick incorporates such a test signal

on the one hand as an uncompressed "WAV" file (preferred) and on the other as an compressed "MP3" file. As long as your car audio system offers a USB connector you can directly hook up the USB stick for playback. If your car stereo

is not equipped with a USB port then you have to burn the "WAV" file via your computer to an audio CD first before you can use it in the CD drive of your head unit.



2 First steps – Starting the software

2. First operating steps with the software

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

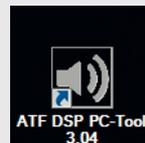
A simple USB cable (included in delivery) is typically all you need to connect any of our DSP devices to your personal computer.

Insert the USB cable into a free port of your computer. Now switch on the device via the radio respectively the remote input or via the highlevel input (except NOX 4DSP).

Assumptions 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.

In order to start the software, please execute the respective icon on your desktop:



Now the following window will appear on your screen:



The software automatically detects within a couple of seconds what device is connected:



By pressing the red "CONNECT" button, the device will be connected with the DSP PC-Tool software.



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



Expert advice:

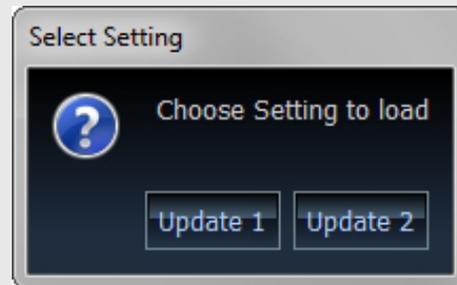
Some equipment for a correct in-vehicle measurement of the frequency response is a "must". Keep in mind that a proper optimization procedure without any tools is hardly possible. That's why Audiotec Fischer is offering "Measurement Tool Kits" that include a high-quality measuring microphone, a preamplifier with USB interface and the necessary connecting cables as well including test signals.

2 First steps - Starting the software

You will then be asked which setting from the internal memory shall be loaded:

If the internal memory is empty or your software version is too old you will see the following hint:

In this case you can choose which memory you like to update – if you don't do anything within 5 sec. the update process of memory 1 will begin automatically:



Important

Updating the firmware completely erases an older setting from the chosen internal memory position!

Simply press "OK" and the update process is started immediately. The update itself will take roughly 10 – 20 sec. Keep in mind that only the chosen storage place will be updated, means if you update "Setting 1", the other setting can still be an older version.

DSP PC-Tool V3 Demo Mode

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



This could happen as well if the USB driver hasn't been installed correctly or the USB cable has not been inserted properly. You can still start the software in demo mode. However, 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.

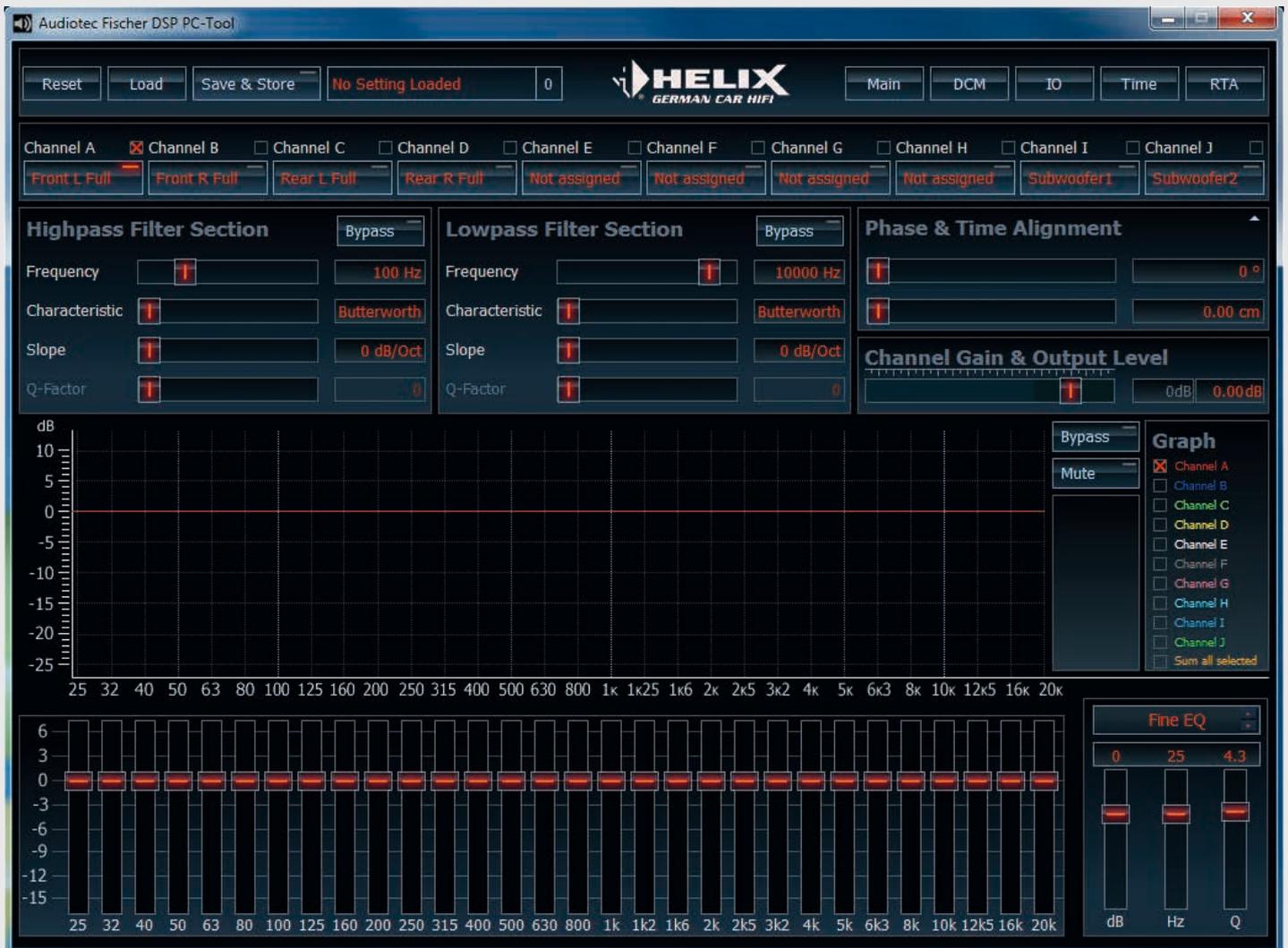
Important

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 DSP 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.

2 First steps – 2.1. The user interface

2.1. 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: channel selection – crossover – time alignment – gain – equalizer.

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 adjustments have been finalized (item 1 to 4) you have to store the complete setup to your DSP device and your hard disk.

2.1.1. 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 a setup with all individual settings to your connected DSP device and the hard drive of your PC.

“Reset” button



The Reset button is used to reset specific or all the adjustments made in the software. Pressing this button will open the “reset menu” where you can choose what parameters shall be reset.



Important

If you press the “Reset” button without having saved your previous setup, you will lose all your data!

If you want to start with a completely new setup based on an existing one, we recommend to checkbox “Reset All” followed by pressing the “Reset” button.

If you don’t want to reset anything, you can leave the submenu by pressing the “Close” button.

“Load” button

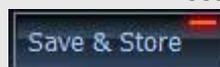


This button allows to open and load an existing setup from the hard disk of your PC. All setups created by the DSP PC-Tool software V3 have the extension “*.afpx”.

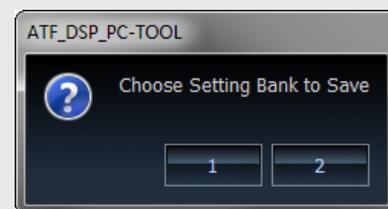
is possible to choose an arbitrary file name by yourself as long as you don’t change the extension “*.afpx”.

After having stored the setup on your computer you will be asked to which setting bank of your DSP device you like to store the setup:

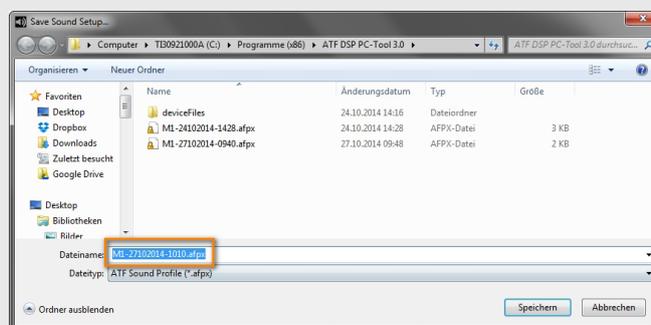
“Save & Store” button



Hitting this button allows to save a setup to a specified location on your PC and to your DSP device. The red light in the button indicates that something has been changed in the setup since the last store operation. If you press the button, the following window will open where you can choose the storage position on your computer and the file name:

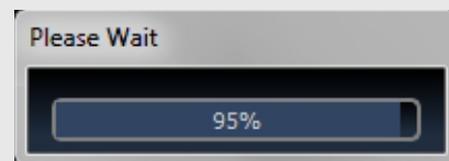


If you don’t press any of the buttons the setup will be automatically stored to bank 1 after a while.



The download process to the DSP device takes a couple of seconds indicated by a bar counting from 100% to zero:

The DSP PC-Tool automatically creates a file name with the system date and system time of your PC. Of course it



Note

Both setting banks 1 and 2 are only available if those have been initialized before during the start-up of the software. If one of the banks contains an

older software version that does not comply with the current DSP PC-Tool software, it has to be updated first.

2 First steps – 2.1. The user interface

Setup Name Indicator



The name of the currently loaded setup is displayed in combination with the setting bank position.

Menu Selector

Here you can select which menu shall be displayed on the screen:



Main: Channel selection, individual settings per channel for highpass/lowpass, phase/time alignment, gain and equalizer

DCM: Device Configuration Menu (for details see chapter 3.1.)

IO: Switches to the "Input & Output Configuration" menu (for details see chapter 5.2.)

Time: Switches to the "Phase & Time Alignment Configuration" menu (for details see chapter 5.3.)

RTA: Opens the "Real-Time Analyzer" overlay (for details see chapter 4.)

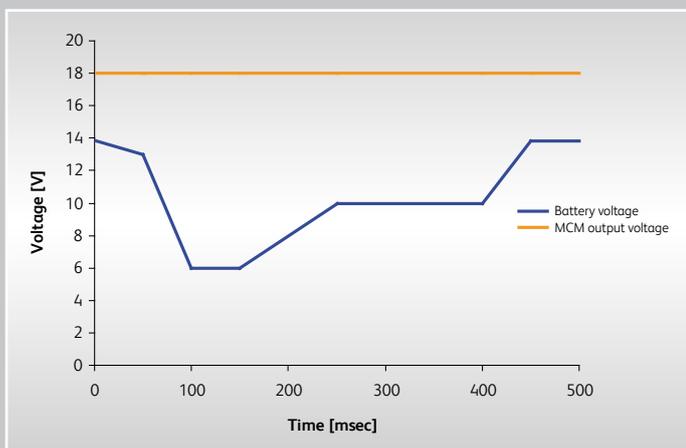
Technology excursion

Extremely compact, efficient and unconditionally stable:

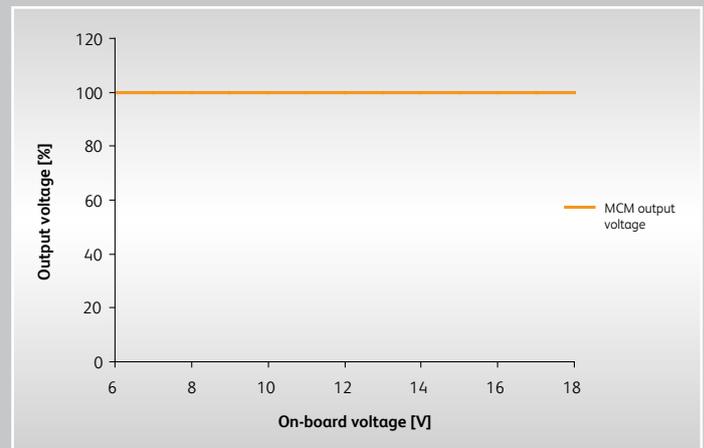
The basis for an optimum sound is a stable power supply. Therefore Audiotec Fischer uses state-of-the-art DC-/DC-converters, which provide stable output voltage under all conceivable operating conditions. But that's not all - our several different power supply concepts all achieve sensational efficiencies with minimum space requirements.

Ready for start / stop

Even if the input voltage drops to 6 Volts the power supply assures a constant internal supply voltage. An ideal precondition for using our amps in cars with start / stop technology.



Internal supply voltage during engine start



Internal voltage in dependency on on-board supply voltage

Advantages of our sophisticated power supply technologies:

- 100 % constant, internal power supply independent of the on-board voltage and the load
- Extremely high efficiency of approx. 96 % – thus the power supply components don't require any cooling
- Switching frequency (350 - 450 kHz) considerably further away from the audio frequency band in comparison with conventional switched power supplies (30 - 40 kHz)
- Full start/stop capability (down to 6 Volts supply voltage)
- Drastically reduced footprint on the PCB enables particularly compact designs

3. Operating with the DSP PC-Tool V3

The different features of the DSP PC-Tool will be explained in this chapter.

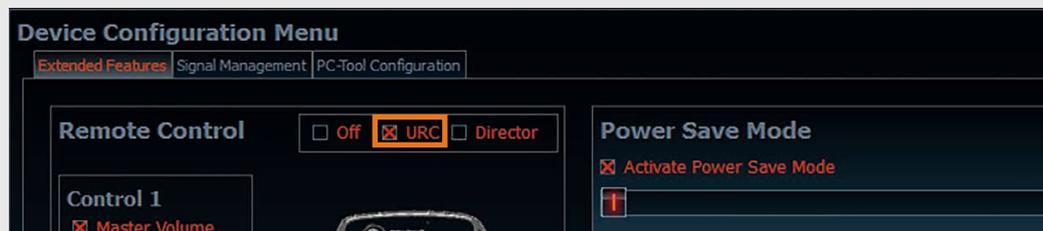
3.1. The device configuration menu (DCM)

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 remote controls. The content of the DCM menu varies with the product. Subsequently every single function within the DCM will be explained in detail using the example of the HELIX DSP PRO.

3.1.1. Extended Features

If you have an optional remote control (e.g. URC 2A) hooked up, you need to address this in your device first. Simply check which type of remote control you like to use:



3.1.1.1. Remote Control “URC 2A”

This cable remote control offers two rotary knobs and a pushbutton – you can define their functionality by yourself. Below you find a short description of the individual possibilities.

“Master Volume”

This 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.

“HEC/AUX Volume”

This allows controlling the volume of a connected Aux source or a HEC input module (e.g. the HEC BT - Bluetooth input module), means that it is no longer necessary to adjust the volume on the source itself. All other signal inputs remain unaffected.

“Digital Volume”

This allows controlling the volume of a connected source via the optical or coaxial SPDIF input. This function makes sense if the digital source doesn't have an own volume control which is typically the case.

“Subwoofer Volume”

Allows adjusting the volume of all channels that have been defined as subwoofer channels in the “IO” matrix.

“Off”

Choose this if you don't need this specific control option.

“Setup Switch”

This allows toggling between two different setups. The function is only given if two different setups are stored in the internal memory. If only one single setup is stored, the “Setup Switch” will have no effect.

“Digital In”

This allows activating the digital input of your device manually. When choosing this, the signal detection circuit of the digital input will be automatically deactivated.

“HEC/AUX In”

This allows activating the HEC module or Aux input of your device manually. When choosing this, the signal detection circuit of the HEC/Aux input will be automatically deactivated.



3.1.1.2. Remote Control “DIRECTOR”

The functionality of the display remote control “Director” (available from March 2015 onwards) will not be configured in the DSP PC-Tool software but via its touch screen directly. So if you have this remote connected to your DSP device, just checkbox the “Director” sign as shown below and proceed with the setup on the remote itself.



The “Director” offers more remote features than the cable version “URC 2A”.

3.1.1.3. Power Save Mode

The Power Save Mode allows to significantly reduce 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 PRO 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 PRO 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 minutes, means that the HELIX DSP PRO

will be turned on for the same time! Current draw without Power Save Mode: 0.4 A (HELIX DSP PRO) + 2 x 2 A (connected amps) x 0.5 hours = 2.2 Ah.

Current draw with Power Save Mode: 0.4 A (HELIX DSP PRO) + 0.0 A (amps turned off) x 0.5 hours = 0.2 Ah.

That’s because the HELIX DSP PRO 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 undesired additional load. The slider control allows to vary the time period until the device activates the “Power Save Mode” from 30 seconds to 600 seconds in increments of 1 sec. (default value: 30 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 remove the tick on “Activate 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.

3.1.2. Signal Management

This sub menu defines the behaviour of the signal detections of the digital and HEC/Aux inputs.

If you have chosen the manual activation of the source via remote control then the respective sliders and priority settings will have no effect.



3.1.2.1. Sensitivity of the signal detection



The respective input will be activated automatically if an input signal is applied (pre-condition: the source selector is set to

“Automatically via Signal Detection”). The slider control “Sensitivity” allows you to adjust whether even very weak signals will activate the respective input (position “-60 dB” or less) or if higher input level is required. The latter may be especially necessary for the

HEC/Aux input if electrical disturbances of other components inside the vehicle will activate the input though no music signal is present. The sensitivity of the signal detection can be varied from -80 dB to 0 dB in steps of 1 dB (default value: -60 dB)

3.1.2.2. Release time of the signal detection

The slider control “Release” defines the time until the amplifier will automatically switch back to the previous source when no more signal is

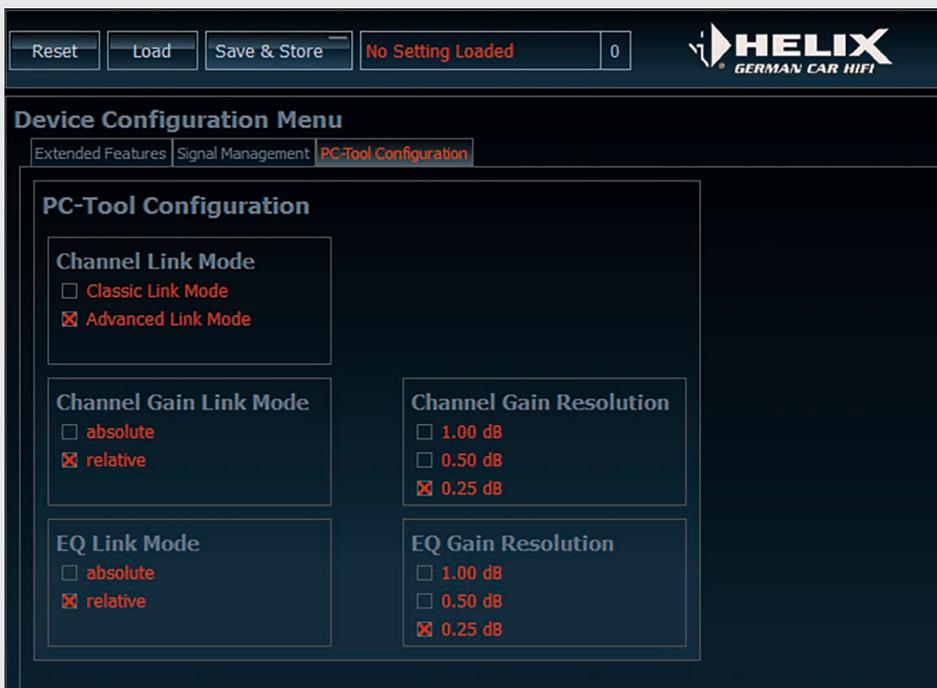
present on the respective input.

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

release time 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 input will automatically switch back to the previous source for a short amount of time. The release time can be adjusted from 0 to 300 sec. in increments of 1 sec. (default value: 3 sec.)

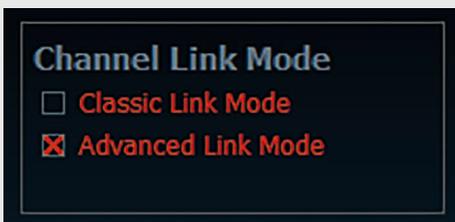


3.1.3. PC-Tool Configuration



This submenu allows general settings of the DSP PC-Tool software.

3.1.3.1. Channel Link Mode



These two different modes are explained in detail in chapter 5.4. “Selecting and linking channels for adjustment”. The “Advanced Link Mode” is the standard and our recommended setup. For those who prefer the link

behaviour that has been used in previous versions of the DSP PC-Tool software should choose the “Classic Link Mode”.

3.1.3.2. Channel Gain Link Mode

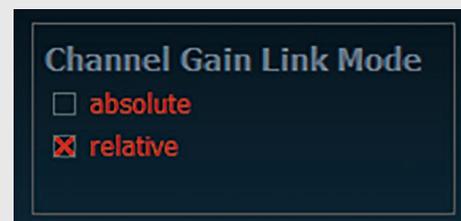
These two different modes are explained as well in chapter 5.4. “Selecting and linking channels for adjustment”.

“**Absolute**” means that the adjusted dB values of the output gain will be identical for all linked channels.

“**Relative**” means that the difference of the output gain will be kept identical for all linked channel.

Here’s an example:

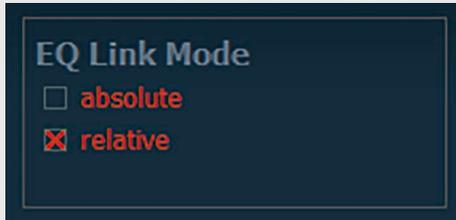
You have set the output gain of channel A to + 2.0 dB, whereas you have chosen a gain of – 1.0 dB for channel B. Now you link the two channels A and B. If you now change the output gain of one of the channels, the difference of 3.0 dB will be kept, means if you set A to 1.0 dB then channel B will automatically be adjusted to –2.0 dB.



Important

Channel gains for both modes will only be correctly set if you link the respective channels **before** you make any adjustment!

3.1.3.3. EQ Link Mode



These two different modes are explained as well in chapter 5.4. "Selecting and linking channels for adjustment".

"**Absolute**" means that the adjusted dB value of a specific EQ band will be identical for all linked channels.

"**Relative**" means that the difference of the dB value of a specific EQ band will be kept identical for all linked channel.



Important

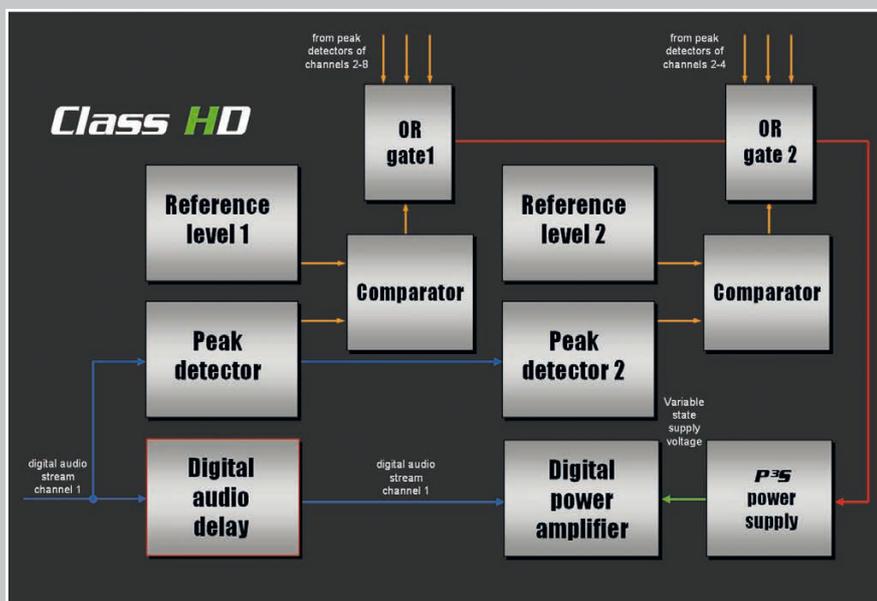
EQ gains for both modes will only be correctly set if you link the respective channels **before** you make any adjustment!

3.1.3.4. Channel and EQ Gain Resolution



In this submenu it is possible to adjust the resolution of the channel gain slider and the EQ gain sliders in the "Main". The standard value is "0.25 dB" but can be changed separately to "0.5 dB" or "1 dB" steps. Actually in practice a "1 dB" step is the best choice for most use cases.

Technology excursion



Class HD technology

The Class HD concept combines the advantages of a Class H technology with the principle of a class D amplifier. The result is an unsurpassed efficiency, which easily outperforms any conventional class D amplifier.

Audiotec Fischers proprietary Class HD concept is implemented, for example, in our PP 82DSP amplifier.

4. The ATF Real-Time Analyzer

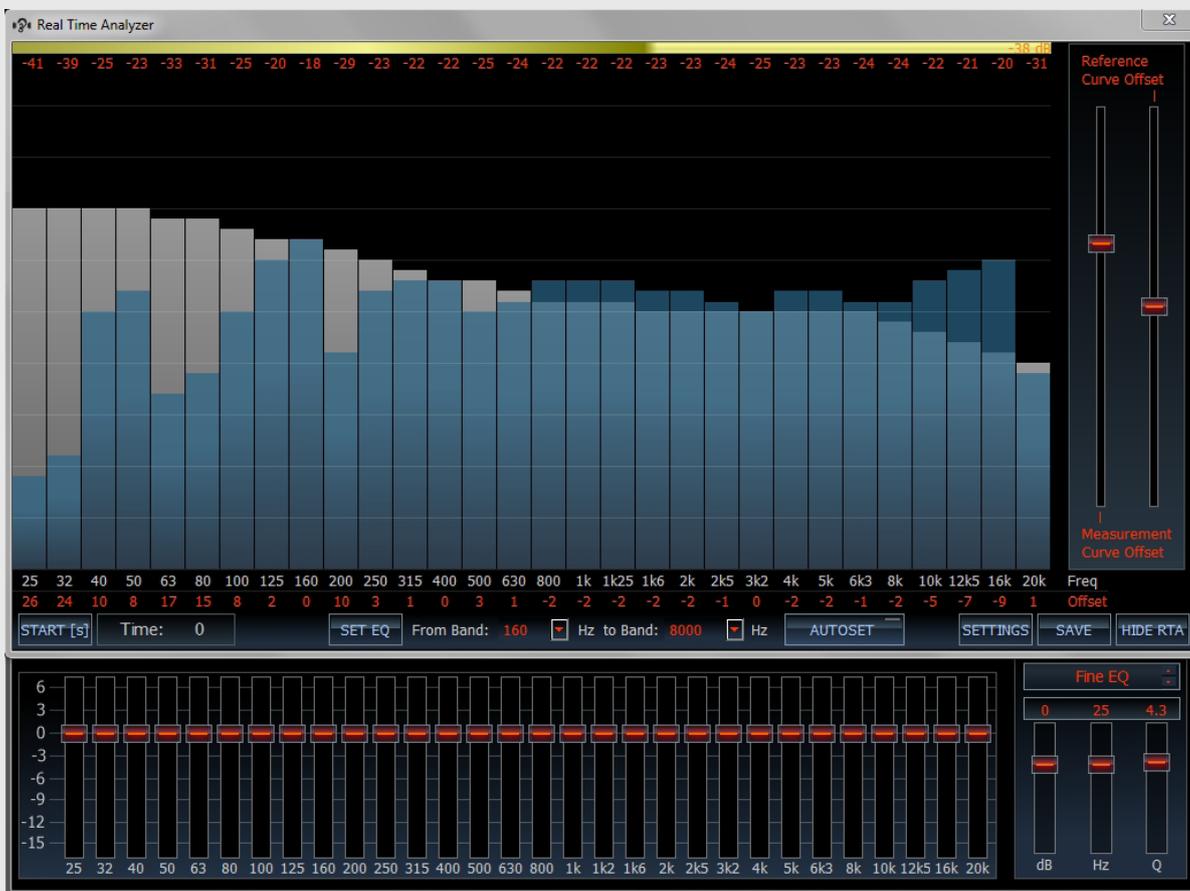
Like in the previous versions you will find in the DSP PC-Tool V3 the Real-Time Analyzer as an indispensable working instrument for a proper DSP setup. Nevertheless this analyzer has been significantly improved.

4.1. Launching the ATF Real-Time Analyzer

At first open the Audiotec Fischer in demo mode, the audio analyzer is DSP PC-Tool and click on the red marked button "RTA". If you use the Audiotec Fischer DSP PC-Tool



without functionality. As soon as one of our DSP devices is properly hooked up to your computer, the audio analyzer will offer its full functionality.



The first thing you will recognize is that the controls for the equalizer are still visible though the RTA has been opened. They are now directly arranged below the analyzer section for easiest access.

The software "ATF DSP PC-Tool" including the Real-Time Analyzer can be downloaded as a complete installation file from the Audiotec Fischer website.

Note

If you use the Audiotec Fischer DSP PC-Tool in demo mode without a connection to any of our DSP devices, then the audio analyzer is not working. In this case the analyzer will be displayed as shown on the right:



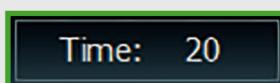
4.2. Explanation of the different functions

To ensure a good workflow it is advisable to get familiar with the various functionalities of the ATF Real-Time Analyzer. The different functions are explained below:



"START" button

To start a measurement click on the blue marked button "START" or simply hit the "s" on your keyboard.



"Time" button

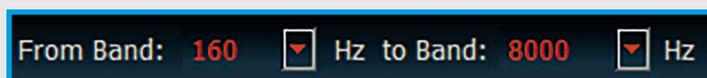
The green marked icon "TIME" displays the duration of the measurement. The default value of 20 seconds cannot be modified. After starting the measurement it will count down from "20" to "0". Nevertheless it is possible to stop a measurement earlier by clicking once more the "START" button.

We strongly recommend to exploit the full duration of 20 sec.



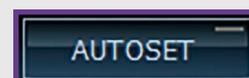
"SET EQ" button

The "SET EQ" button provokes that the "deviation" values of all defined EQ bands will be automatically transferred to the EQ setting. We explicitly recommend to read the details in chapter 5.7.3. before using the "SET EQ" function.



"Frequency range" drop-down menu

Here you define a reasonable frequency range for the auto-correction of the equalizer. This is important for a correct operation of the "AUTOSET" function.



"AUTOSET" button

The "AUTOSET" button starts an automatic loop of "measuring" and "EQ adjustment". We explicitly recommend to read the details in chapter 5.7.3. before using the "AUTOSET" function.



"SETTINGS" button

Hit the orange marked button "SETTINGS" to enter the settings menu.



"SAVE" button

The yellow marked button "SAVE" allows saving the measured frequency response curve as a "png" file to the hard drive of your computer.



"HIDE RTA" button

If you want to close the RTA display, just press the "HIDE RTA" button.

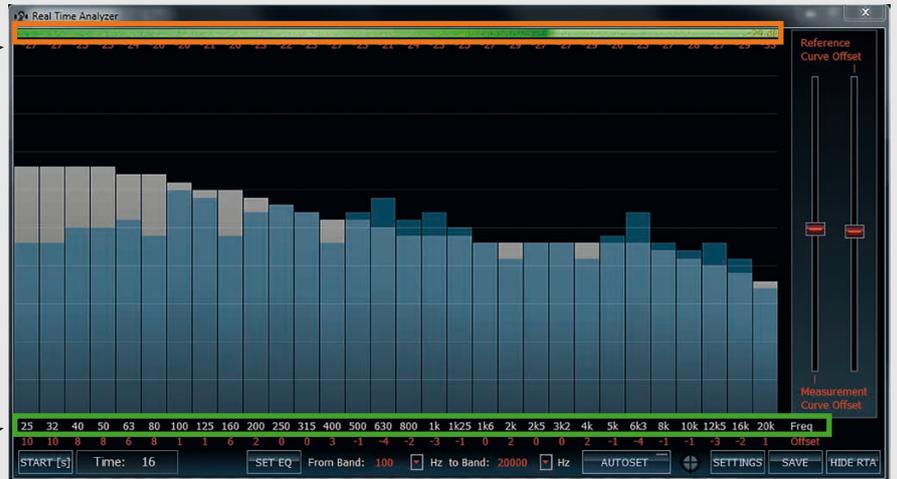
4 Real-Time Analyzer – 4.2. The different functions

Sound pressure level in dB

The values below the coloured bar show the measured sound pressure level for each frequency band in decibels [dB].

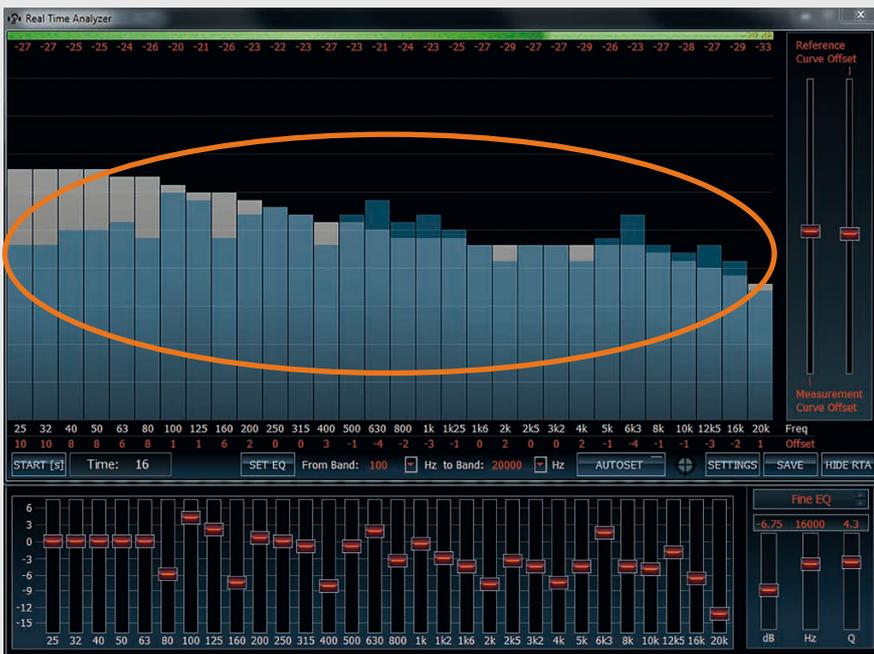
Frequency band in Hz

Here are the center frequencies of each single frequency band displayed in Hertz [Hz].



Note

All sound pressure dB values are no absolute but relative values only!



Frequency response curve

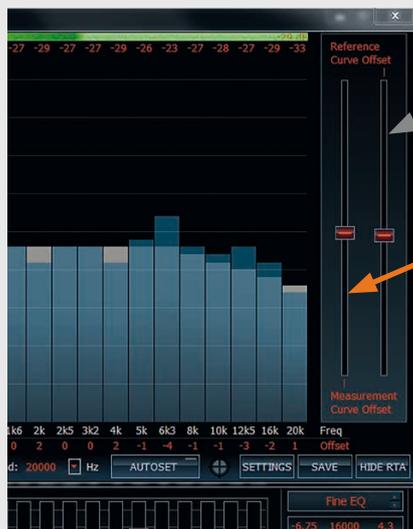
The thirty (30) blue bars show the measured frequency response curve in 1/3rd octave steps from 25 Hz to 20,000 Hz.

Note

The equalizer of the DSP PC-Tool has exactly the same frequency band resolution as the analyzer as long as you don't make any changes in the DSP PC Tool using the "Fine EQ" feature!

Reference curve

The thirty grey bars behind the orange bars indicate the chosen reference curve in 1/3rd octave steps.



Slider

You can move the reference curve vertically up and down using the right slider on the right hand side.

Or you move the measured curve using the left slider on the right hand side.

Expert advice:

We recommend to move the reference curve so that the deviation of the reference and the measured curve will show minor dips and stronger peaks! Keep in mind that equalizing peaks are no problems at all, but "filling up" dips can lead to distortion and increased noise level!



Offset

This line (yellow marked) shows the offset of each measured frequency band from the reference curve in [dB]. Positive values correspond to boosts of specific bands, negative values mean that you have to cut certain bands accordingly.

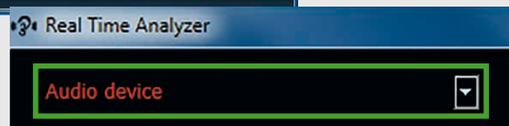
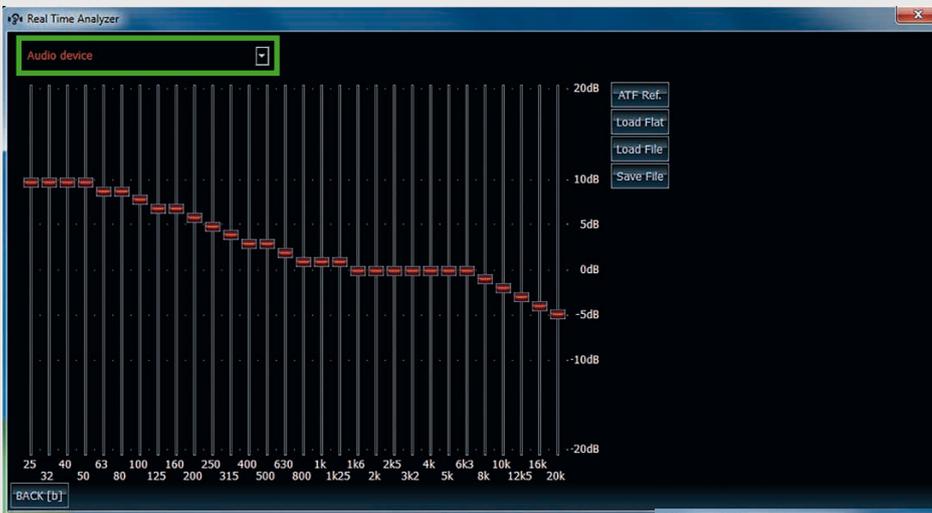
Take great care with strong boosts (> 4dB) as this can easily lead to over-modulation of the DSP and therefore dramatic increase of distortion. Deep cuts (-10 dB and more) are no problem at all.

4.3. Setting up the Real-Time Analyzer

For configuring the Real-Time Analyzer, please click on the orange marked button "SETTINGS".



Then you will see the following menu:



Important

Check first whether your connected microphone is actually the selected device (see green marked frame). It may happen that the internal microphone of your computer is chosen instead which would lead to totally wrong measuring results.

4.3.1. Configuring the reference curve

Choose the desired reference curve which you finally will achieve with the adjustment of the DSP. The "ATF Reference" is selected as default as this curve is identical to today's industry standard.

Alternatively you can choose a linear (flat) reference curve. To load the linear curve just click

on the red marked button "Load Flat". This curve should only be used for home applications, not for car audio measurements, it definitely will not lead to satisfying results when applied for car audio purposes.

If the ATF Reference does not meet your personal taste it is also possible to define your own reference curve. Each single frequency band of the curve can be adapted separately.

After all settings in this menu have been done, you can switch back to the analyzer display by clicking on the "BACK" button or by hitting the "b" on

your computer's keyboard.

Expert advice:

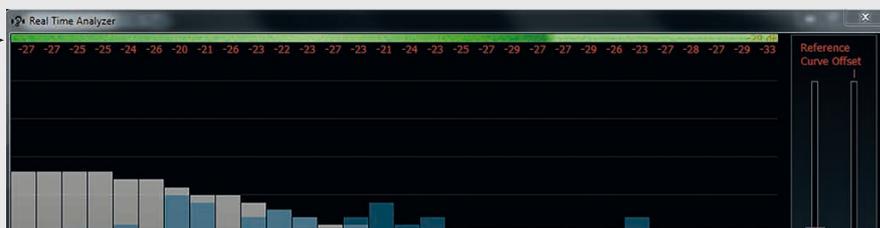
Save a self-defined reference curve via "Save File" (blue marked button) to your hard drive that you can recall it later on via the "Load File" button (yellow marked button).

4.4. Sound level indicator

To attain an optimal measurement result it is advisable to adjust the correct sound level. The ATF Real-Time Analyzer displays by a coloured bar in the upper segment if the sound level of your audio equipment is sufficient for the measurement or not.

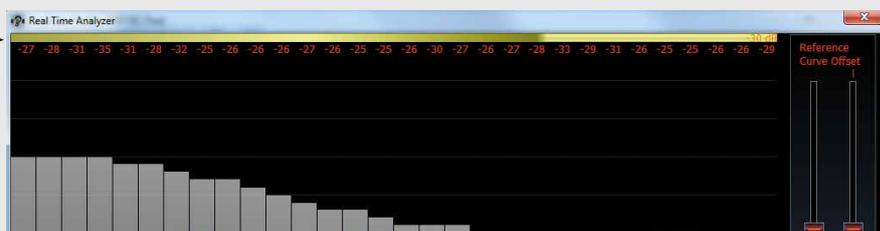
Green bar

Sound pressure level is correct!



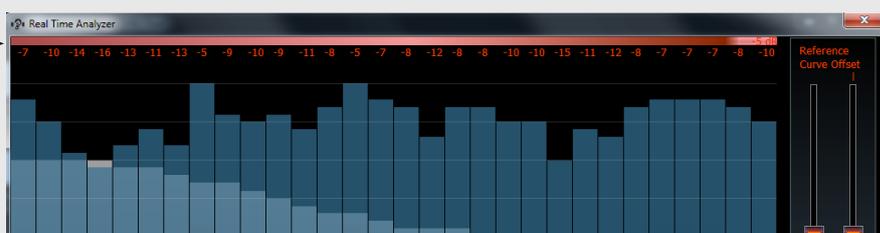
Yellow bar

Sound level too low; Crank up the volume of your car radio or check the sensitivity adjustment of the microphone in the control panel of your computer.



Red bar

Sound level too high; Reduce the volume level on your car radio or check the sensitivity adjustment of the microphone in the control panel of your computer.



4.4.1. Adjusting the correct sound level

1. Take place on the seat in the car in which the measurement will be done.
2. Start the "pink noise" playback on your car radio. Adjust the volume that any ambient noise is fully masked.
3. Hold the measuring microphone upright. Click on the "Start" button (or simply enter "s") and move the microphone slowly in a semicircle between your left and right ear.
4. Now the coloured bar indicates if the sound level is sufficient for the measurement or not.
5. This is an iterative process which may require several runs until you reach the correct sound level.



Note

You have to do the adjustment of the correct sound level for each vehicle and different measurement positions.

5. Configuring a sound setup

Step by step to your own sound setup

The creation of a sound setup consists of four work steps. At first you have to do the channel routing followed by the time alignment and the filter settings. At last do the equalizing.

Note

You achieve the best result when the car is parked in the closed garage that no ambient noise can affect the measurement.

5.1. Preparations

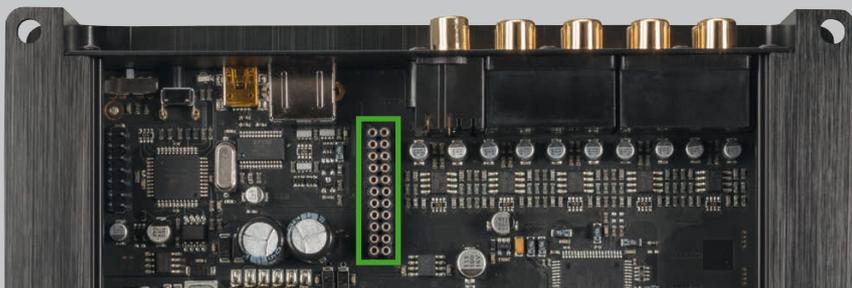
- Sit in the driver's seat and make sure that all sound controls of your car radio are set either to linear or center position. Deactivate any existing loudness functions. The balance and fade controllers must be adjusted to "center position" as well.
- Connect your amplifier / processor to your PC and start the ATF DSP PC-Tool software.



Technology excursion

HEC slot

The HELIX Extension Card Slot, simply called "HEC", makes a device extremely flexible and versatile. It allows, by using optional HEC modules, to extend the functionality of the device - for example by a Bluetooth® audio streaming module, an additional optical digital input or an optical digital output.



HELIX Extension Card slot

5.2. Channel routing – Configure the Inputs & Outputs (IO)

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

- Start with the setup of the channel routing matrix first. You can skip this step in case of the PP 52DSP when you do not use the RCA line outputs. In the appendix you will find a couple of examples for typical applications.

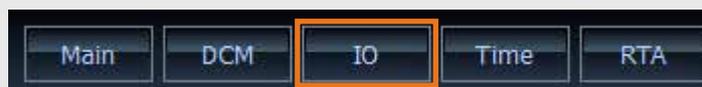


The HELIX DSP PRO provides four sets of stereo preamplifier inputs (RCA/Cinch; A - G) and an optical or coaxial stereo digital input. In addition the inputs A - H offer highlevel inputs which can be directly connected to the speaker outputs of a car radio.



The “IO” menu

Clicking on the “IO” button opens the channel routing overview where you can configure any of the inputs and outputs.



This allows you to assign names to each of the individual channels, to mix input signals and allocate a specific input signal to each output.

To protect the connected loudspeakers, default parameters for each of your settings can be loaded automatically, which may, of course, be modified as required.



The HELIX DSP PRO offers ten output channels in total – therefore you can find five output pairs (RCA/Cinch; A – J) on its back panel. The routing matrices define which input channel will be allocated to which output channel. Even mixing of several input channels to one output channel is possible.

Note

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

5 Sound setup – 5.2. Channel routing

5.2.1. Naming of the inputs

On the left hand side of the “IO” menu you can allocate a name to each of the input channels. Double-click on the respective channel to open a selection menu where you can choose from a list of names. It is not possible to enter different names than the ones offered in the pull-down menu:

This naming does not provoke any function so far, but it helps you to better differentiate between the several channels. The following names can be chosen:

- Front L Full(range)
- Front R Full(range)
- Front L High
- Front R High



- Rear R Full(range)
- Rear L High
- Rear R High
- Rear L Mid
- Rear R Mid
- Rear L Low
- Rear R Low
- R(ear) Center Full(range)
- R(ear) Center High
- R(ear) Center Low
- R(ear) Sum
- Subwoofer 1 - 4
- Line In 1- 8
- Digital In 1
- Digital In 2
- Opt In 3 – 8
- Opt Sub 1 - 8
- Front L Mid
- Front R Mid
- Front L Low
- Front R Low
- F(ront) Center Full (range)
- F(ront) Center High
- F(ront) Center Low
- F(ront) Sum
- Rear L Full(range)

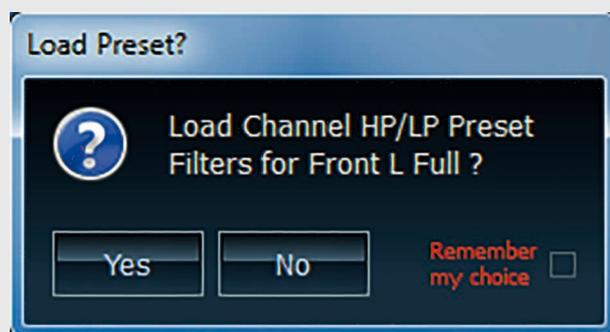
5.2.2. Naming of the outputs

Just as with the inputs, you are able to allocate names to the outputs on the right hand side of the input matrix.

Click on the orange marked arrow key to open a pull-down menu and choose a name:

As soon as you choose one of the options you will then be asked if you want to load the respective preset filter adjustment for this channel:

Any unused output channel should ideally be set to “Not assigned” so that it will be muted automatically.



We strongly recommend answering this question with “Yes”. This prevents, for example, that a tweeter connected to an output accidentally receives a fullrange signal which may damage it in seconds.

Default settings of individual channel types

Output channel	Default Highpass	Default Lowpass
Not assigned	Channel output is muted	
Front L / R Full	-	-
Front L / R High	3237 Hz, 12 dB, Butterworth	-
Front L / R Mid	511 Hz, 12 dB, Butterworth	3237 Hz, 12 dB, Butterworth
Front L / R Low	-	3237 Hz, 12 dB, Butterworth
F Center Full	-	-
F Center High	3237 Hz, 12 dB, Butterworth	-
F Center Low	-	3237 Hz, 12 dB, Butterworth
F Sum	-	-
Rear L / R Full	-	-
Rear L / R High	3237 Hz, 12 dB, Butterworth	-
Rear L / R Mid	511 Hz, 12 dB, Butterworth	3237 Hz, 12 dB, Butterworth
Rear L / R Low	-	3237 Hz, 12 dB, Butterworth
Rear R Fullrange	-	-
R Center Full	-	-
R Center High	3237 Hz, 12 dB, Butterworth	-
R Center Low	-	3237 Hz, 12 dB, Butterworth
R Sum	-	-
Subwoofer 1 - 4	-	80 Hz, 24 dB, Butterworth
Line Out 1 - 8	-	-

Note

All default filter settings of course can be modified in the "Main" menu. They are only used to protect the connected loudspeakers and are no recommendation for a correct setting. Keep in mind that it is only possible to change the output names in the "Main Routing" matrix – these names will automatically be used for the other matrices as well!

5.2.3. Allocating inputs to outputs

In the middle of the "IO" menu, you can allocate any of the input signals to each of the outputs.

This is done by a simple "drag & drop" function. Just click on the input of your choice, hold the left mouse button and drag this input to the selected output channel. It is mandatory

to allocate at least one input to every output that you like to use. Any output channel can be the sum of up to four input channels, which for instance makes sense if you like to generate a

sum signal for the subwoofer from all four front and rear input channels like in the example below:



Removing a chosen input channel from the routing matrix is easy as well – just do a right mouse click on the respective element in the matrix and it will disappear immediately.

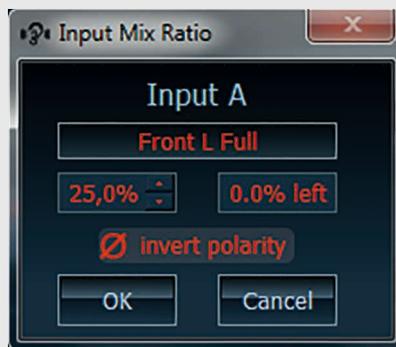
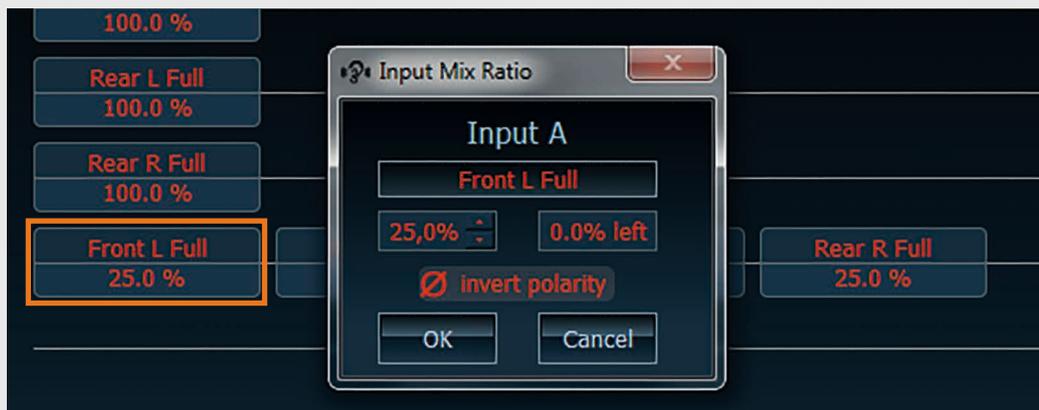
5 Sound setup – 5.2. Channel routing

If you add two or more input channels to one output, each input channel is weighted equally. You will notice that the “percentage” value will automatically be adapted if you add input channels. But it is possible to change this ratio as well. Double-click on the element in the matrix and a new window will open, allowing you to change the percentage.

This can be done by using the up and down arrows or by directly editing the value.

It is even possible to reverse the polarity of a single input channel in the matrix – just set a tick in the “Invert” box.

This could make sense if you e.g. like to generate a differential signal out of the two rear channels for a so-called “rear fill” speaker.



Note

Note that the total sum of all input channels for a dedicated output channel can not exceed 100%!

5.2.4. Other routing matrices

Depending on the device you have, there won't be only a “Main Routing” matrix but other matrices as well.

The HELIX DSP PRO actually offers three different matrices, allowing you to have a separate routing for Aux sources and digital sources as well. As long as you only have one single source (for instance your car radio head unit), you only need the “Main routing”, but as soon as you like to feed your device with more sources simultaneously, one single matrix won't be sufficient.

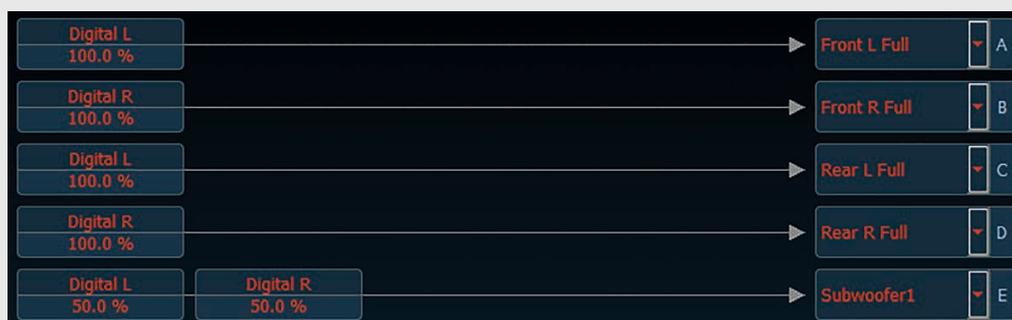
Here's an example why a separate routing for the individual sources might be necessary:

Normally a car radio delivers four channels (Front L, Front R, Rear L and Rear R) or sometimes five channels (FL, FR, RL, RR plus subwoofer) which have to be allocated to the outputs like in the paradigm shown in chapter 5.2.3.

But a source like a DVD player connected via the optical digital input or an MP3-Player via Aux input or HEC Bluetooth module will only deliver two input channels which require a different routing.

The example below shows a typical allocation of two digital input channels to the output channels in a five-channel system.

You can proceed in the same way with the “AUX Routing” matrix, if you have an AUX source connected.



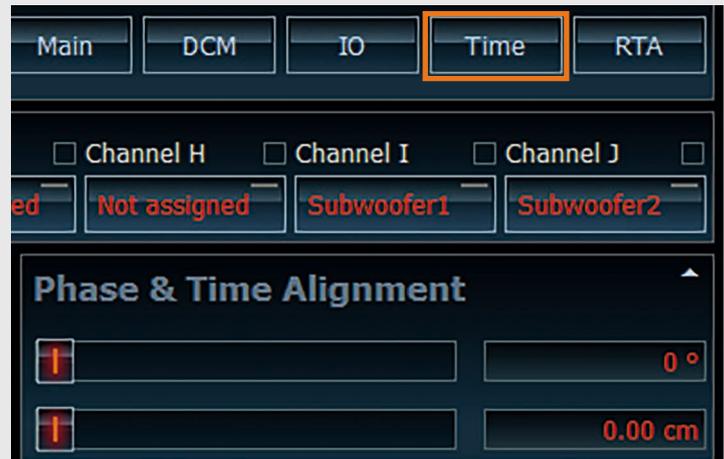
5.3. Phase and time alignment (TIME)

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

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.

The adjustment of phase and time alignment can be done in the main menu, but we strongly recommend to use the "Phase & Time Alignment Configuration" menu instead display-

ing all values at a glance. So you don't need to permanently switch back and forth between the individual channels when adjusting the time alignment and phase.



The phase and time alignment can either be adjusted for each channel individually in the main menu or, more comfortably, for all channels by clicking on the "Time" button.

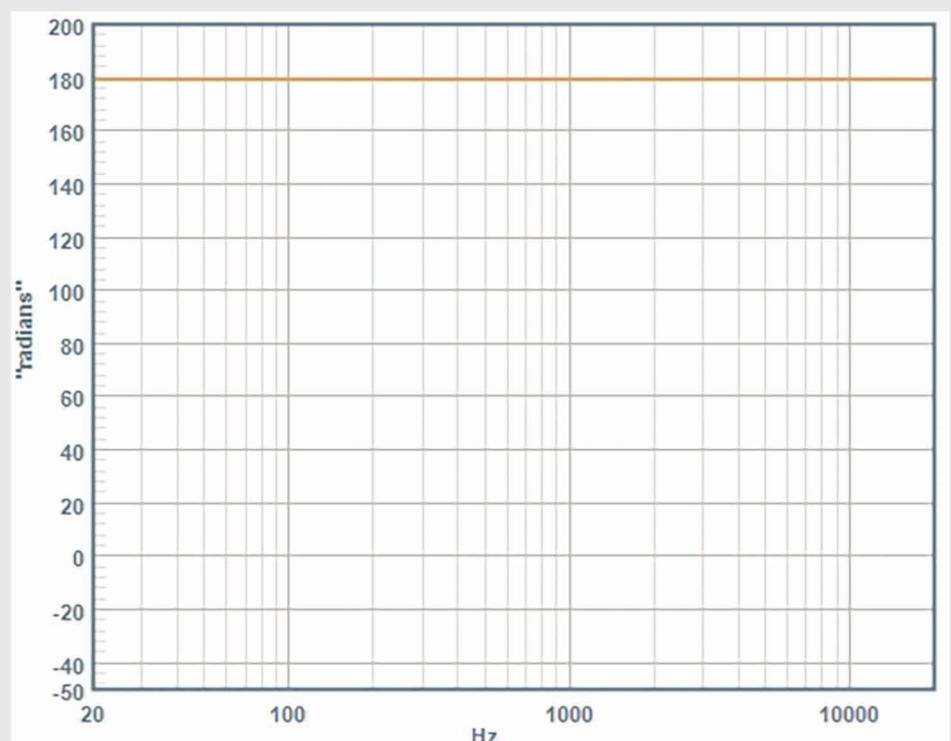
5.3.1. Adjusting the phase

The functionality of the phase slider depends on the definition of the respective channel in the "IO" menu.

Channels that are defined as full-range channels only allow to switch phase between 0° and 180°, whereas channels that are defined as subwoofer channels or mid / high channels in fully active systems can be precisely adjusted from 0° to 348.75° in 11.25° increments. Both ways of phase adjustment behave significantly different – so please make yourself familiar with these options.

Phase switch:

The phase switch 0° / 180° simply inverts the phase / polarity of the signal, causing no additional delay. The reversed phase is frequency independent:



5 Sound setup – 5.3. Phase and time

Phase control:

In several use cases it is important to have the possibility of a more subtle phase adjustment.

In the following example the high-pass filter is set to 1,008 Hz and the phase to 90°:

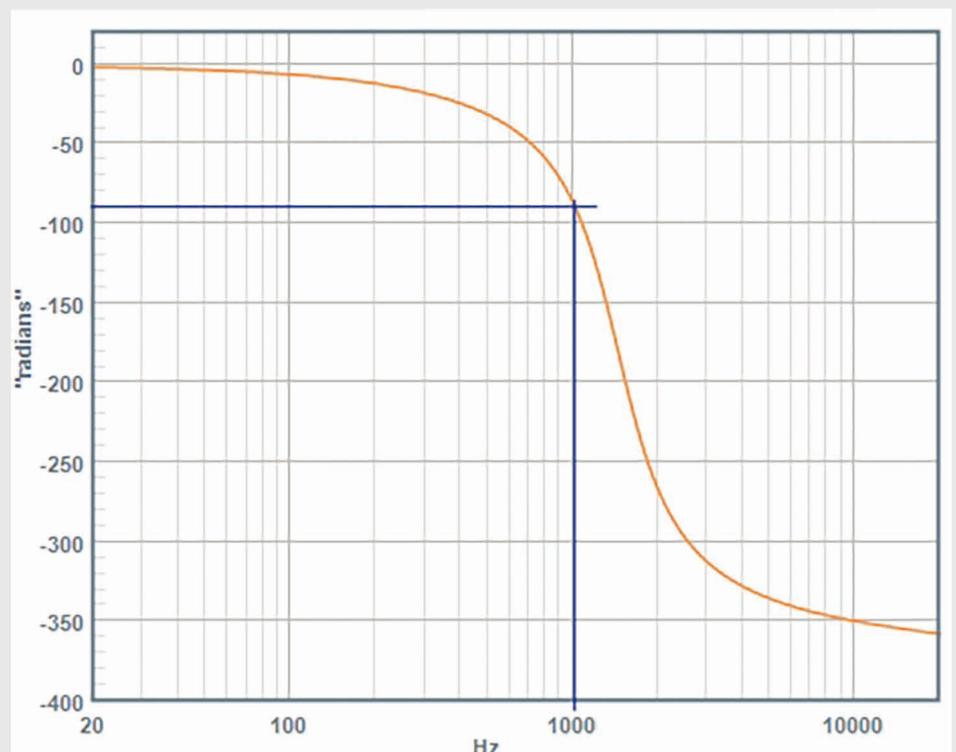


Especially when you want to precisely adapt the subwoofer to the front and rear speakers, the “traditional” way of switching the phase between 0° and 180° often does not lead to satisfying results. Here a phase control with much finer phase steps can do a great job. The same necessity may occur when you try to match a midrange and a tweeter at a given crossover frequency in a fully active sound system. Here as well both standard values of 0° or 180° can cause undesired peaks or dips in the frequency response at the crossover point due to an improper phase relation.

Though the phase control in our product may look quite simple, it is mandatory for a proper operation to understand what it actually does. The phase control is realized as a 2nd order allpass filter with a variable corner frequency. Allpass filters do not have an effect on the amplitude of the signal, but add a frequency dependent phase shift.

When using the phase slider you do not have direct access to the corner frequency of the allpass filter. The DSP PC-Tool automatically calculates the appropriate corner frequency depending on the setting of your high-pass filter. Only in case of the subwoofer channel the chosen lowpass filter cut-off frequency will be the reference for calculating the phase shift.

The graph below shows the frequency-dependent phase response with a 90° phase shift at 1,008 Hz:

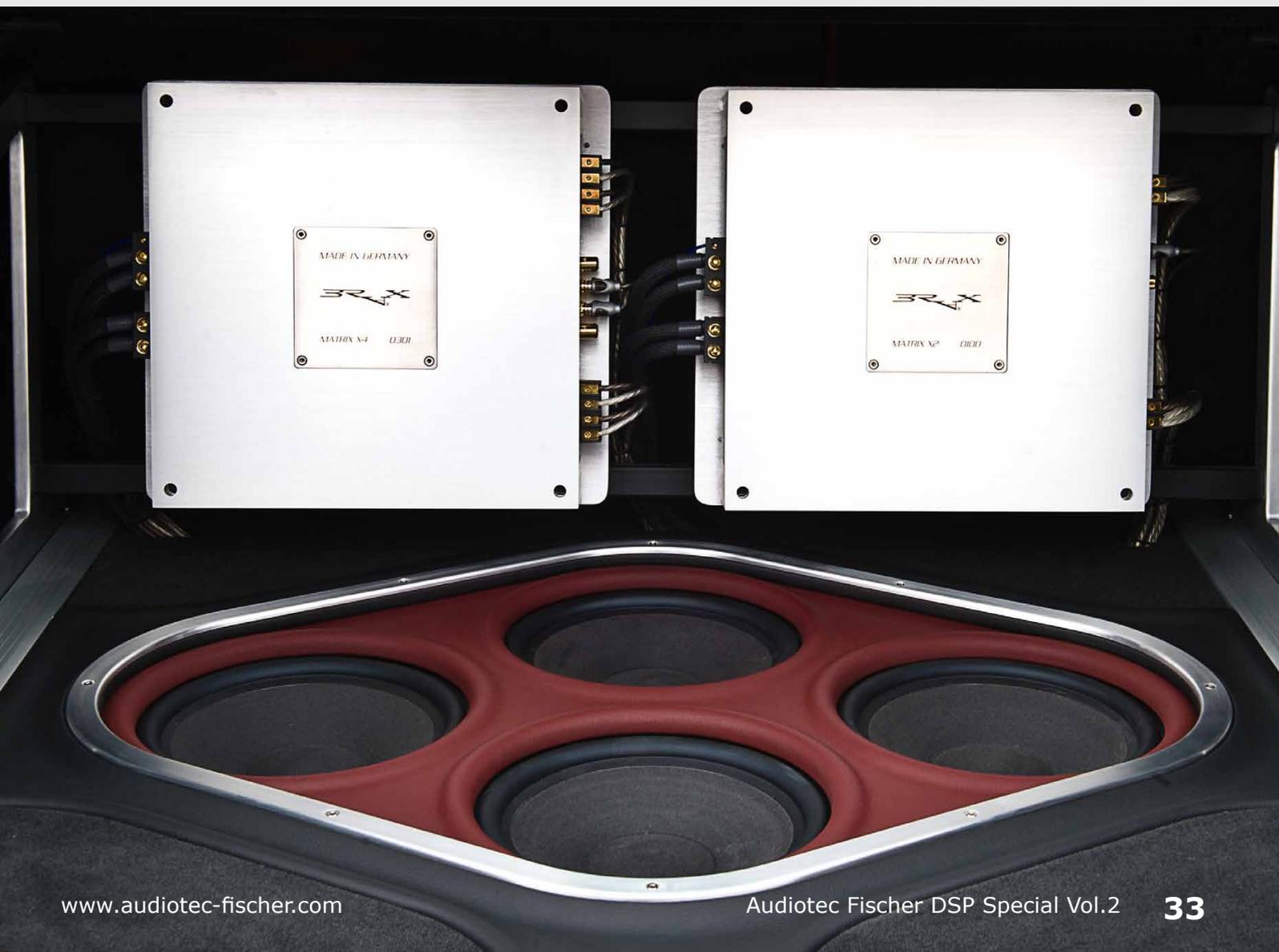


The actual calculated corner frequency of the allpass filter in this case is 1,450 Hz, but this is not displayed anywhere as this info is of no value for the user.

Expert advice:

If you want to setup a fully active sound system consisting of subwoofer, woofer, midrange and tweeter, then always start with adjusting the phase relationship between subwoofer and woofer at the crossover point, followed by phase relation between woofer and midrange and finally with the phase adjustment between midrange and tweeter. Never try the same in reversed order if you don't want to go mad!

Channels defined as:	Steps:	Phase reference:
Front L/R Full	0° / 180°	n/a
Front L/R High	11.25° steps	cut-off frequency of highpass filter
Front L/R Mid	11.25° steps	cut-off frequency of highpass filter
Front L/R Low	0° / 180°	n/a
Front Center Full	0° / 180°	n/a
Front Center High	0° / 180°	n/a
Front Center Low	0° / 180°	n/a
Rear L/R Full	0° / 180°	n/a
Rear L/R High	11.25° steps	cut-off frequency of highpass filter
Rear L/R Mid	11.25° steps	cut-off frequency of highpass filter
Rear L/R Low	0° / 180°	n/a
Rear Center Full	0° / 180°	n/a
Rear Center High	0° / 180°	n/a
Rear Center Low	0° / 180°	n/a
Subwoofer 1/2/3/4	11.25° steps	cut-off frequency of lowpass filter



5.3.2. Adjusting the time alignment

It is only possible to achieve a correct “front staging” of the music if the time alignment is set correctly.

It is only possible to achieve a correct “front staging” of the music if the time alignment is adjusted properly.

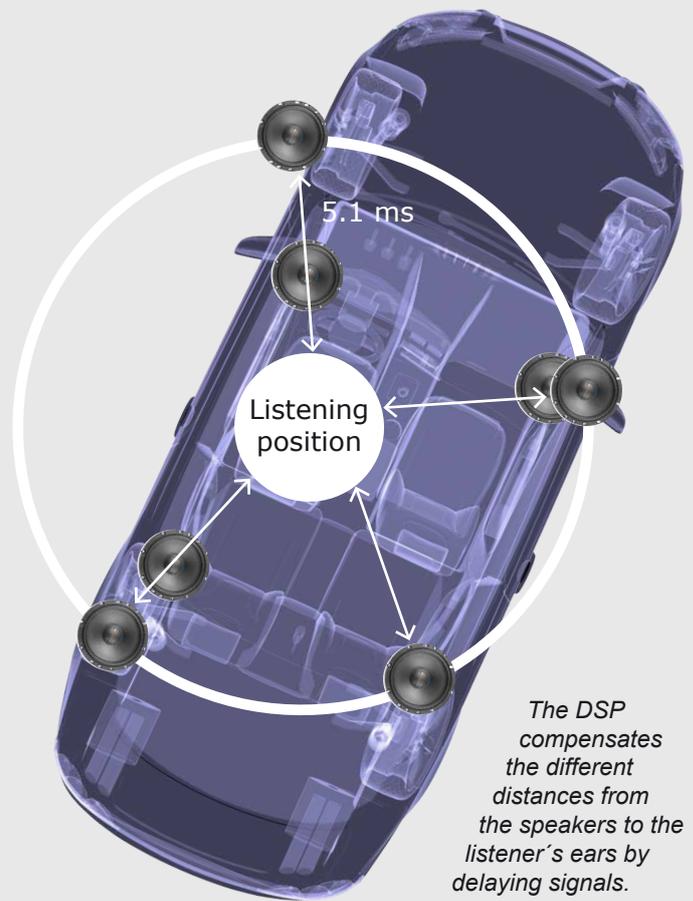
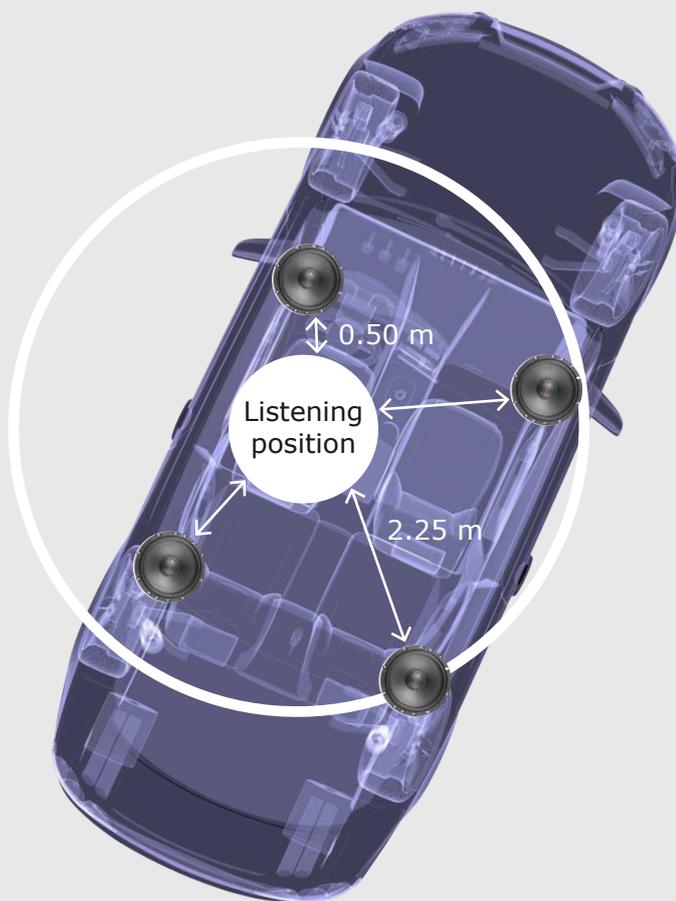
Therefore the software allows sensitive adjustments in increments of 7 mm / 0.021 msec for all devices

with 48 kHz sample rate and even finer 3.5 mm / 0.0105 msec steps for those running at 96 kHz.

The DSP PC-Tool provides now two or three different ways of adjusting the



time alignment (depending on the set-up in the “IO” menu)



The DSP compensates the different distances from the speakers to the listener's ears by delaying signals.

■ Important

The time alignment function may only be used to achieve the optimum stereo reproduction at one single listening position in the vehicle. However, spatial reproduction at the other positions are usually bad with time alignment. Always switch on the time alignment before adjusting

the frequency response with the equalizer. When you adjust the time alignment of the subwoofer use either the “tape measure method” or the “Phase” slider. Other more precise methods are not necessary.

“Delay”

This mode requires the direct input of the required time delay values for each channel.



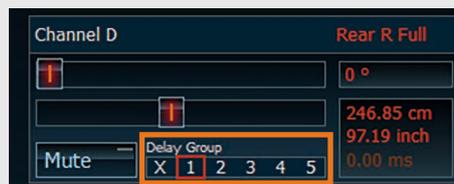
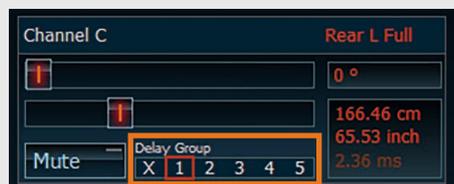
Here the DSP PC-Tool software does not assist you in any way, means that you need to calculate by yourself what time delay is required.

“Delay Groups”

Sometimes it makes sense to combine selected channels during the time alignment adjustment.

In many cases you achieve a perfect front staging only if you add an extra delay to the rear speakers (means more delay is required than the software has calculated by itself due to the distance values you entered).

In this case simply join the two rear channels to a “delay group” (here group 1).



“Distance”



Simply use a tape measure to determine the space in cm or inches (the program always displays both values). The program uses this to calculate the necessary time delay for each channel. Have a look at the time delay value in the blue box while you change the distance value of a different channel – you will notice that it will be automatically adapted. Whatever distance you enter in one of the channels – the one with the furthest distance will be set to “zero” time delay and is the reference for all other channels.

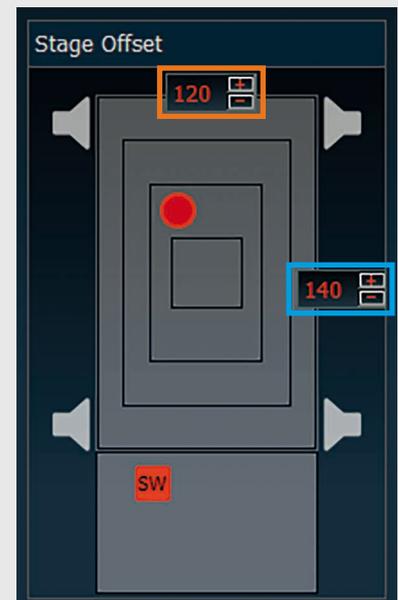
Now you can move one of the sliders of the two rear channels and the time delay will be adapted on **both** channel in parallel. You can define up to 5 different delay groups depending on your application.

“Graph”

The “Graph” method is probably the easiest way to set up the time alignment, but this will only properly work in typical 5-channel speaker configurations, consisting of two front speakers, two rear speakers and a subwoofer in the trunk. That’s why this method is only available in some specific cases, depending on how you have defined your channel configuration in the “IO” menu.

All you have to do is measuring the distance between the two front speakers and the distance between a front speaker and a rear speaker. Enter these values in “cm” in the orange and blue marked boxes in the graph.

Now you can move the red dot with



the cursor of your PC until you achieve a nice front staging. The subwoofer (squared “SW” dot) can be moved separately for further bass improvement.

Note that this method is damned fast and easy to use, but in most cases the best results will be achieved with the “distance” method.

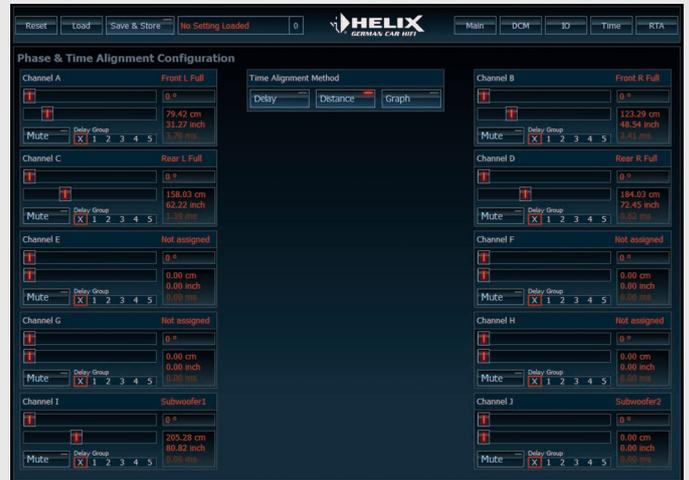
5 Sound setup – 5.3. Phase and time

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

The optimum setting can easily be found by using the “Distance” mode as follows:

1. Open the time alignment overview and choose the “Distance” mode.

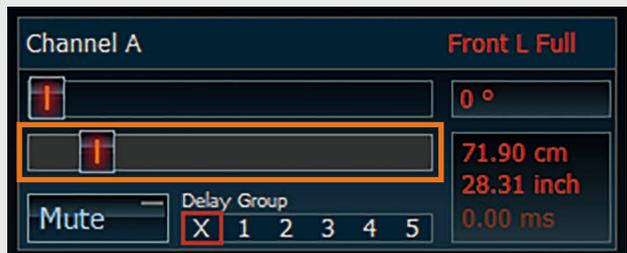
The time alignment overview displays the distance and delay values of all channels at a glance.



2. Use a tape measure to measure in centimeters / inches the distance between the front left loudspeaker and your head in your listening position.



3. Adjust this distance value using the slide control for the respective channel.
4. 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”. It is not necessary to measure / adjust the values as accurately as possible as they merely serve as initial values for the fine adjustment.



5. Now mute the rear channels and the subwoofer using the "Mute" function.



6. Select a music track which preferably consists only of vocals without instruments or chorus. Adjust the time alignment of each front speaker (Front left ... or Front right ...) in small increments until you can hear the voice either directly in front of you or slightly to the right, the so-called 1 o'clock position. In order to achieve the desired result you can adjust the time alignment by a few centimeter/ msec.

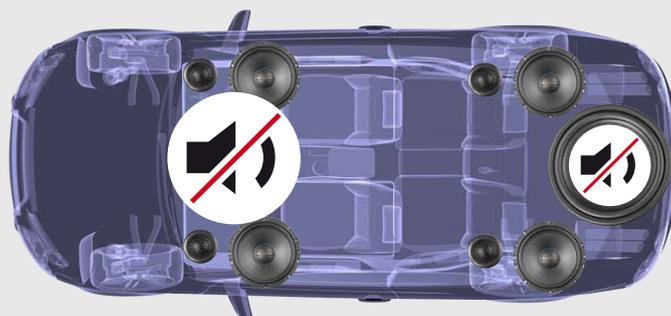


Advice:

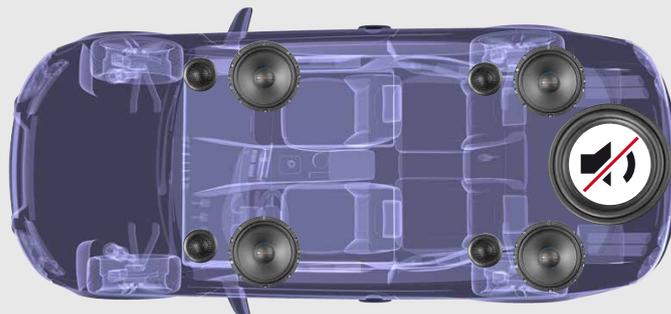
Use the arrow keys of the keyboard for easier fine adjustment

7. Now mute the front channels using the "Mute" function and reactivate the rear channels.

8. Adjust the time alignment of the rear speakers in the same way (use the same vocal track) until you hear the voice directly from behind or slightly from the right behind (the so-called 5 o'clock position).



9. Unmute / reactivate the front speakers. Keep the subwoofer channel muted as before.

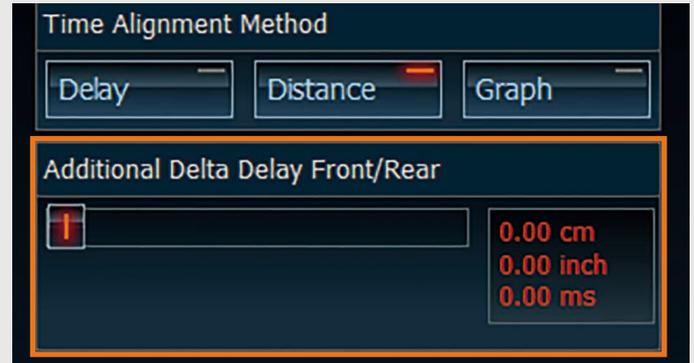


(continued on the next page) →

5 Sound setup – 5.3. Phase and time

(continued from page 37)

→ **10.** Some of our products offer the so-called "Additional Delta Delay Front / Rear" slide control to "electronically" shift the rear speakers backwards until it is almost impossible to notice them because then they cannot have any negative impact on a perfect front staging.



11. If you have a device without this feature we recommend using the "Delay Group" feature and link the rear channels together. Then you can influence the time delay of all rear channels at the same time by moving one single slider of one of the rear channels. According to experience values of approx. 100 cm (corresponds to 3 msec. delay) as an "extra delay" for the rears are sufficient .

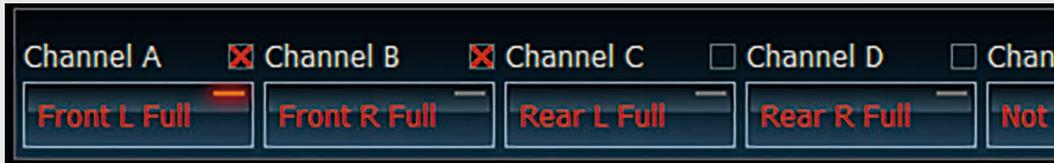


12. If you have done the time alignment for the front and rear channels correctly, then demute all the other channels. This completes the adjustment of the time alignment.



5.4. Selecting and linking channels for adjustment

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 because different values for right and left channel should be avoided. Even with the equalizer a separate adjustment



of channels only makes sense if sound optimization is required for a single listening position only.

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

channels).

There are several modes for linking channels available (selection can be done in the Device Configuration Menu "DCM", chapter 3.1.).

Important

When linking two or more channels the previous adjustments are not automatically transferred to the other channel/s. Only those adjustments which are made **after** linking will be applied to the linked channels.

Therefore please consider first if two or more channels should be adjusted simultaneously.

Expert advice:

When creating a vehicle-specific setup, Audiotec Fischer adjusts the gain, filter and EQ settings of the two front channels identically. The rear channels will be proceeded in the same way.

The "Advanced Link Mode"

This mode allows you to permanently link several channels.

Here's an example how it works:

You want to permanently link the channels A and B and the channels C and D.

First select channel A:



Now set a tick on channel B to link these two channels:



Every change of any highpass-/lowpass filter, gain, phase or EQ setting you do on channel A **or** B will now be applied to both channels A **and** B!

Then choose channel C:



Set a tick on channel D to link these two channels:



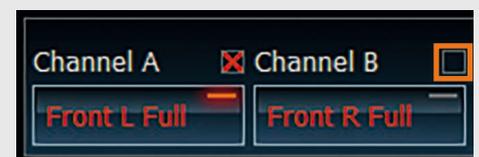
Every change of any highpass- / lowpass filter, gain, phase or EQ setting you do on channel C **or** D will now be applied to both channels C **and** D!

You can proceed in the same way with all the channels you like to link. If you want to "unlink" linked channels, please do the following:

Choose the channel you like to unlink, here in this example channel A:



Then remove the tick on channel B and then both channels can be adjusted individually again:



5 Sound setup – 5.4. Selecting and linking channels

The “Classic Link Mode”

This mode allows you to temporarily link several channels. The linking of channels is quite similar to the “Advanced Link Mode”.

Here’s an example how it works:

First select channel A:



Now set a tick on channel B to link these two channels:



Every change of any highpass / lowpass filter, gain, phase or EQ setting you do on channel A will now be applied to both channels A **and** B!

Important

The temporarily linkage of the two channels A and B will be cancelled automatically as soon as you select any other channel!

Note

Even with the “relative linking” function it is not possible to exceed the gain limits of -15 to +6 dB for each individual EQ band.

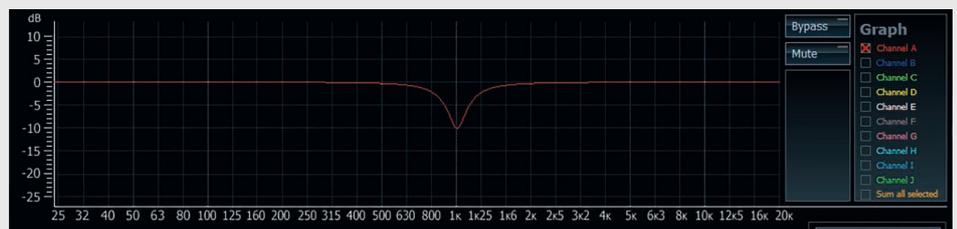
Absolute and relative linking of channels

The DCM allows you to select whether you want to carry over EQ and gain values of one channel to another channel in an absolute or relative manner.

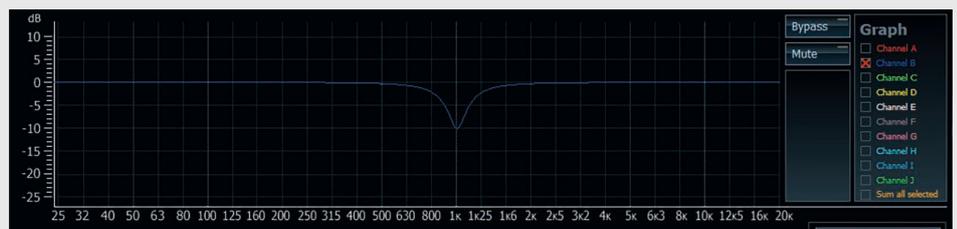
Absolute linking

All values of the EQ and gain settings of a specific band applied after the linking will be identical for the linked channels. In the example below you can see that the EQ setting of the 1,000Hz band for channel A is automatically copied to the linked channel B:

Channel A:



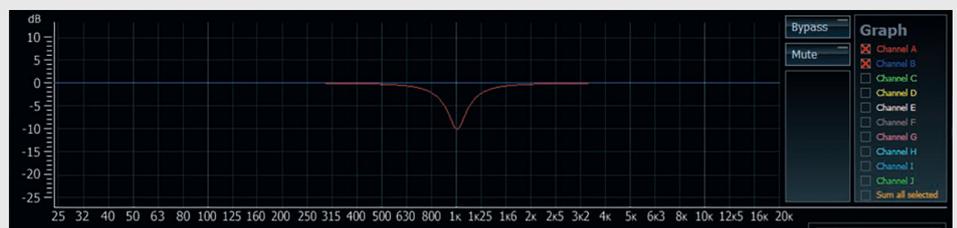
Channel B:



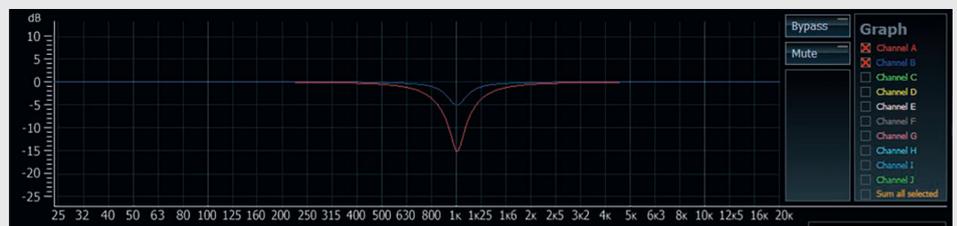
Relative linking

All values of the EQ and gain settings for a specific band applied after the linking will be added to the values of the linked channels that have been made before the linking. Here’s an example – channels A and B have different EQ settings before they were linked. Channel A has an EQ filter set to 1,000 Hz -10 dB whereas channel B is flat:

Channel A:



Now channels A and B will be linked and an EQ filter on channel B is set to 1,000 Hz -5 dB. So these additional “-5 dB” at 1,000 Hz of channel B will be added to channel A:



5.5. High- and lowpass filter settings

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

5.5.1. Adjustment of high- and lowpass filter

This section enables the configuration of almost any type of highpass filter (left side) and lowpass filter (right side) for the selected channel.

The following parameters have to be

adjusted separately:

- Crossover frequency
- Filter characteristics
- Slope
- Q-factor

(The "Q-factor"-slide control is only activated if you select the "Self-Define" characteristics).

The functions of the individual parameters are described below.



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).

5.5.2. Adjustment of crossover frequency

Use the slide control to select the required crossover frequency for the highpass or lowpass filter.

The setting accuracy depends on the device type and is shown in the chart.

	Channel	Filter	Slope	Frequency range	Frequency resolution
HELIX DSP PRO NOX 4DSP	Fullrange	High-/Lowpass	0 - 42 dB/Oct.	20 - 20,480 Hz	1/24 Oct.
	Subwoofer	High-/Lowpass	0 - 42 dB/Oct.	20 - 300 Hz	1 Hz
HELIX DSP MATCH DSP PP 82DSP PP 52DSP	Fullrange	High-/Lowpass	0 - 30 dB/Oct.	20 - 20,480 Hz	1/24 Oct.
	Subwoofer	High-/Lowpass	0 - 30 dB/Oct.	20 - 300 Hz	1 Hz

5 Sound setup – 5.5. Filter settings

5.5.3. Adjustment of filter characteristics

Butterworth (Q = 0.71)

- 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 = 0.58)

- 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 = 0.50)

- 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. 0.90)

- 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 highpass required for tweeters which are operating close to their resonance frequency

Self-Define

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

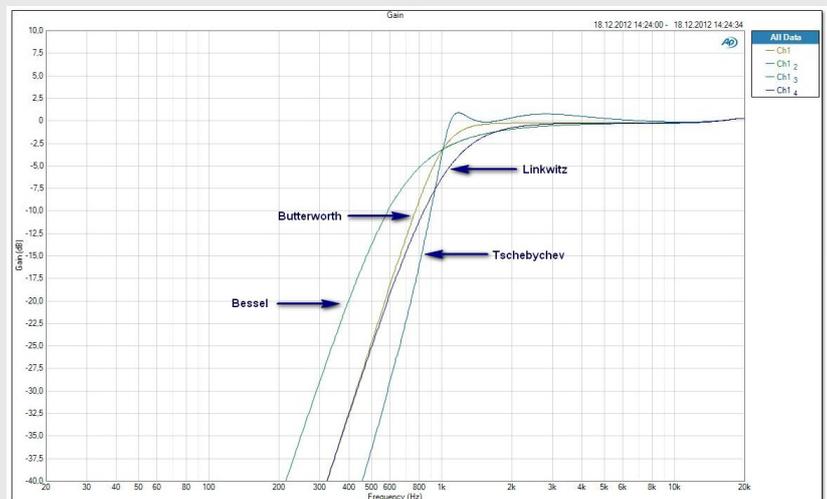
There are four different filter characteristics available for selection and the option of setting your own highpass and lowpass filter using "Self-Define". The table on the left presents an overview of the typical properties of the different filter characteristics.

If in doubt, choose preferably "Butterworth". This filter function is predominantly used in commercial crossovers.

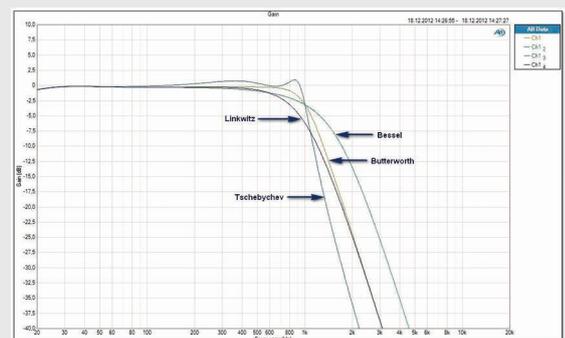
Note

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 HELIX DSP PRO and NOX4 DSP).

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



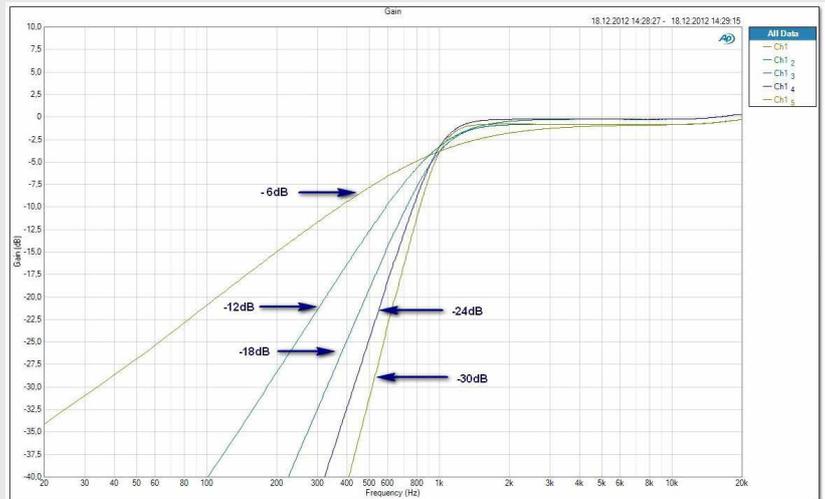
Examples of the different filter characteristics are presented for a 1,000 Hz highpass filter (above) and lowpass filter (right) with a slope of 24 dB per octave.



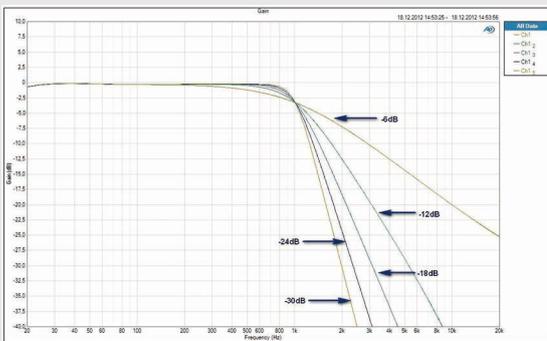
5.5.4. Adjustment of slope

Use this slider 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 HELIX DSP PRO / NOX4 DSP and a maximum of 30 dB per octave on HELIX DSP, MATCH DSP, PP 52DSP and PP 82DSP.

The effect of the slope on the frequency characteristics of a filter is illustrated in the graph below showing an example for a highpass filter and a lowpass filter with 1,000 Hz cut-off frequency and Butterworth characteristic.



The effect of the slope on the frequency characteristics of a filter is illustrated using the example of a 1,000 Hz highpass filter (above) and a lowpass filter (left), both with Butterworth characteristic.



Tips for adjustment

If you have any doubts choose preferably "Butterworth". This filter function is predominantly used in commercial crossovers.

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

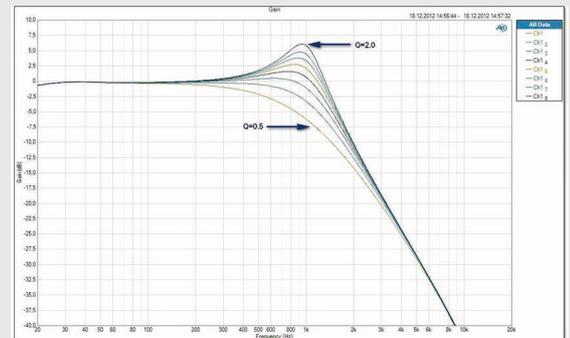
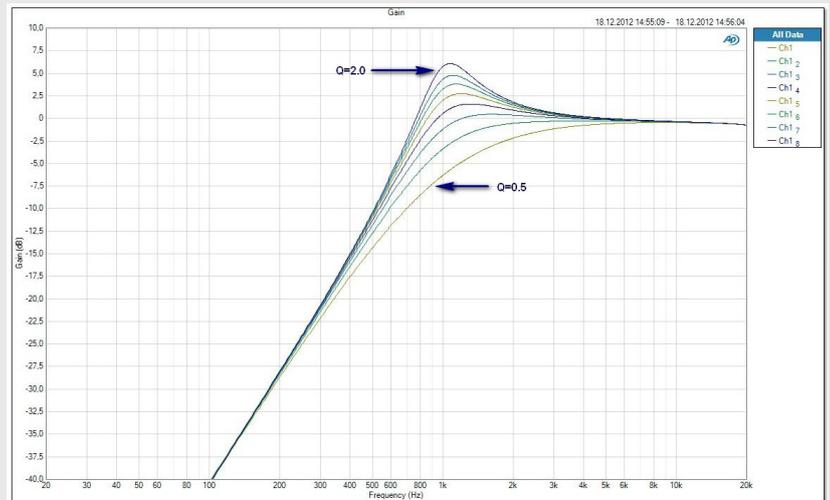
- The steeper the slope, the worse the filter impulse response.
- The preferred slope for the subwoofer channel lowpass filter is "-24 dB".
- A standard value for the crossover network between the woofer and the tweeter in fully active systems is "-12 dB".
- If the frequency response of a midwoofer shows 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 operating in a fully active system at up to 3,000 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 highpass 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.

5 Sound setup – 5.5. Filter settings

5.5.5. 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 on the right side using the example of a high- and lowpass filter with a crossover frequency of 1,000 Hz.



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 1,000 Hz.

5.5.6. The bypass function of the high- and lowpass filters

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".



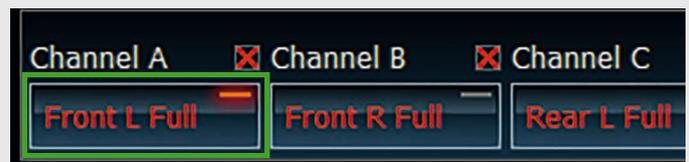
Important

In fully active systems, do not simply bypass the tweeter highpass filter. Without any frequency crossover, irreparable damage may be caused even at low vol-

umes! Bypassing a lowpass filter, on the other hand, is generally uncritical and will not result in any damage of the loudspeaker.

5.5.7. Adjusting the filters

1. 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 (green marked in the picture).



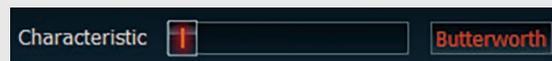
2. **Crossover frequency**

Use the slide control to select the required crossover frequency for the highpass or lowpass filter. The setting accuracy depends on the device type.



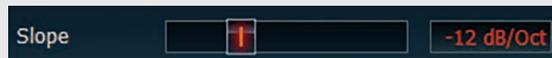
3. **Filter characteristics**

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 in the appendix presents an overview of the typical properties of the different filter characteristics.



4. **Slope dB/Oct**

With this slider it is possible to modify the transition steepness of the filter. At a value of "0 dB" the filter is not active. The adjustment ensued in 6 dB increments up to the maximum which depends on the connected device.



5. **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.

Expert advice:

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

5 Sound setup – 5.5. Filter settings

Examples of crossover frequencies

Especially in combination with a separate subwoofer it makes sense to relieve the speakers from low frequencies 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:

Passive systems				
Speaker size	Filter type	Filter slope	Characteristic	Minimum crossover frequency
6x9" or 20 cm / 8"	highpass	-12 dB	Butterworth	40 Hz
16,5 cm / 6.5"	highpass	-12 dB to -24 dB	Butterworth	50 Hz
13 cm / 5.25"	highpass	-18 dB to -24 dB	Butterworth	70 Hz
10 cm / 4"	highpass	-18 dB to -24 dB	Butterworth	80 -100 Hz

For fully active speaker configurations you will find some reasonable filter settings below. Please note that these values are good for starting with a new setting but most likely need some fine tuning during the setup process depending on your specific application!

Active systems				
Speaker size	Filter type	Filter slope	Characteristic	Minimum crossover frequency
Tweeter 20 mm / 0.75"	highpass	-24 dB	Butterworth	3,000 Hz
Tweeter 20 mm / 0.75"	highpass	-12 dB	Butterworth	4,000 Hz
Tweeter 20 mm / 0.75"	highpass	-6 dB	Butterworth	6,000 Hz
Tweeter 25 mm / 1"	highpass	-24 dB	Butterworth	2,000 Hz
Tweeter 25 mm / 1"	highpass	-12 dB	Butterworth	3,000 Hz
Tweeter 25 mm / 1"	highpass	-6 dB	Butterworth	4,000 Hz
Midrange 50 mm / 2"	highpass	-24 dB	Butterworth	800 Hz
Midrange 50 mm / 2"	highpass	-12 dB	Butterworth	1,000 Hz
Midrange 50 mm / 2"	highpass	-6 dB	Butterworth	1,500 Hz
Midrange 50 mm / 2"	lowpass	-12 dB	Butterworth	3,000 - 5,000 Hz
Midrange 85 mm / 3.5"	highpass	-24 dB	Butterworth	500 Hz
Midrange 85 mm / 3.5"	highpass	-12 dB	Butterworth	800 Hz
Midrange 85 mm / 3.5"	highpass	-6 dB	Butterworth	1,000 Hz
Midrange 85 mm / 3.5"	lowpass	-12 dB	Butterworth	2,500 - 4,000 Hz
Woofers 13 cm / 5.25"	highpass	-18 dB to -24 dB	Butterworth	70 Hz
Woofers 13 cm / 5.25"	lowpass	-12 dB	Butterworth	3,000 - 5,000 Hz
Woofers 16,5 cm / 6.5"	highpass	-12 dB to -24 dB	Butterworth	50 Hz
Woofers 16,5 cm / 6.5"	lowpass	-12 dB	Butterworth	2,500 - 3,500 Hz
Woofers 20 cm / 8"	highpass	-12 dB	Butterworth	40 Hz
Woofers 20 cm / 8"	lowpass	-12 dB	Butterworth	2,000 Hz
Subwoofer	lowpass	-24 dB or higher	Butterworth or Tchebychev	60 - 100 Hz

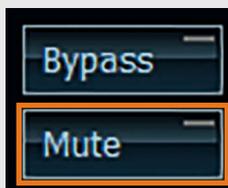
5.6. Channel Gain & Output Level

This small part of the “main window” seems to be self-explaining, but a wrong adjustment cannot only ruin your sound quality but your equipment as well!

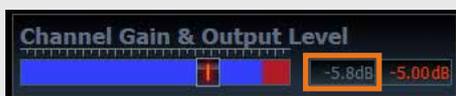


Provided you have properly set up the overall gain of your device chain (head unit - DSP – amplifier) then you only need the gain slider to match the different sensitivities of your speakers in your car. In practice this slider should only be moved by a couple of dB's. If you need larger values for adjustment, check your configuration for any failures / misconnections.

Don't use the Channel Gain for muting the signal of a specific channel – that will not work as the maximum attenuation is -30 dB only. For that case each channel has a “Mute” button which you find on the right side of the frequency graph.



If you apply a signal to a certain input channel you will notice a blue bar in the gain slider which represents the output scale. The actual value in dB is additionally displayed in the marked box.



The other box indicates the setting of the gain slider, here -5 dB.



Don't get confused by the red area next to the blue bar – this does not indicate that the output signal is already overdriven, but already warns you that

you are quite close to maximum level (0 dBFS).

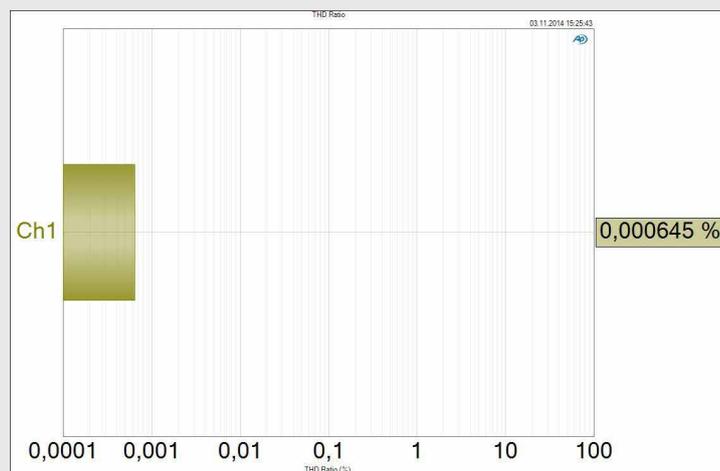


It is getting dangerous if you see a completely red bar in the gain slider with gain values greater 0 dBFS).



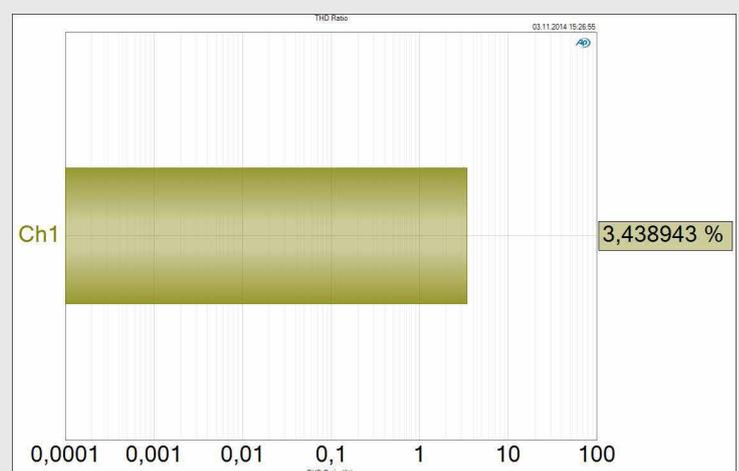
That is really an alarm signal – your output is now overdriven, creating lots of distortion.

A THD measurement shows quite clearly the effect of overdriving a digital system. Typically the best signal-to-noise and THD values are achieved when you operate the device close to full scale (0 dBFS), but as soon as you exceed this “magic limit” of 0 dBFS, you might kiss your tweeters goodbye.



THD of a DSP device at full scale (0 dBFS)

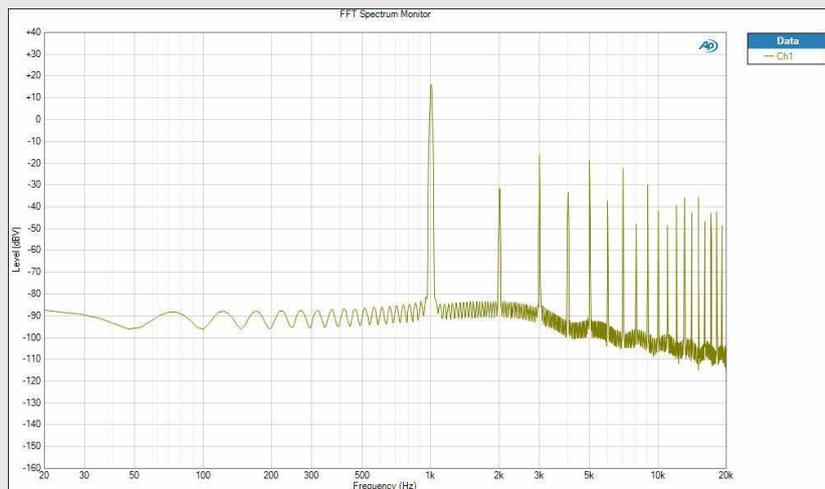
The same device but here the output level has been overdriven by 1 dB only!



5 Sound setup – 5.6. Channel Gain & Output Level

Though the THD value of more than 3 % might not sound too awful, a closer look at the frequency spectrum reveals the drama. In the graph below you see an overdriven 1 kHz output signal with all its undesired distortion components (“harmonics”) at 2 kHz, 3 kHz, 4 kHz, 5 kHz and so on.

This enormous amount of additional energy has to be handled by the tweeters, but unfortunately they aren’t designed for that!



Over-driving a digital system causes way more harmonics than over-driving an analog amplifier – this is very dangerous especially for tweeters!

5.6.1. The resolution of the gain control

It is possible to adjust the resolution of the gain slider in the Device Configuration Menu (DCM).

The standard value is “0.25 dB” but can be changed to “0.5 dB” or “1 dB” steps. Actually in practice “1 dB” steps are the best choice for most use cases.

Important

Carefully watch the level display when you do a setup, especially when you crank up the volume of your head unit.

Overdriving the system should be avoided at any time if you don’t intend to ruin your equipment!

Technology excursion

Class Ultra D amplifier technology

Unlike conventional Class D-amplifiers the “Class Ultra D” concept incorporates the indispensable output filter into the feedback loop. This offers significant advantages which are responsible for the unique sound quality of this digital amplifier:

a) The frequency response of the amplifier is almost independent of the load while conventional Class D-amplifiers have different frequency responses at 2 Ohms, 4 Ohms or 8 Ohms.

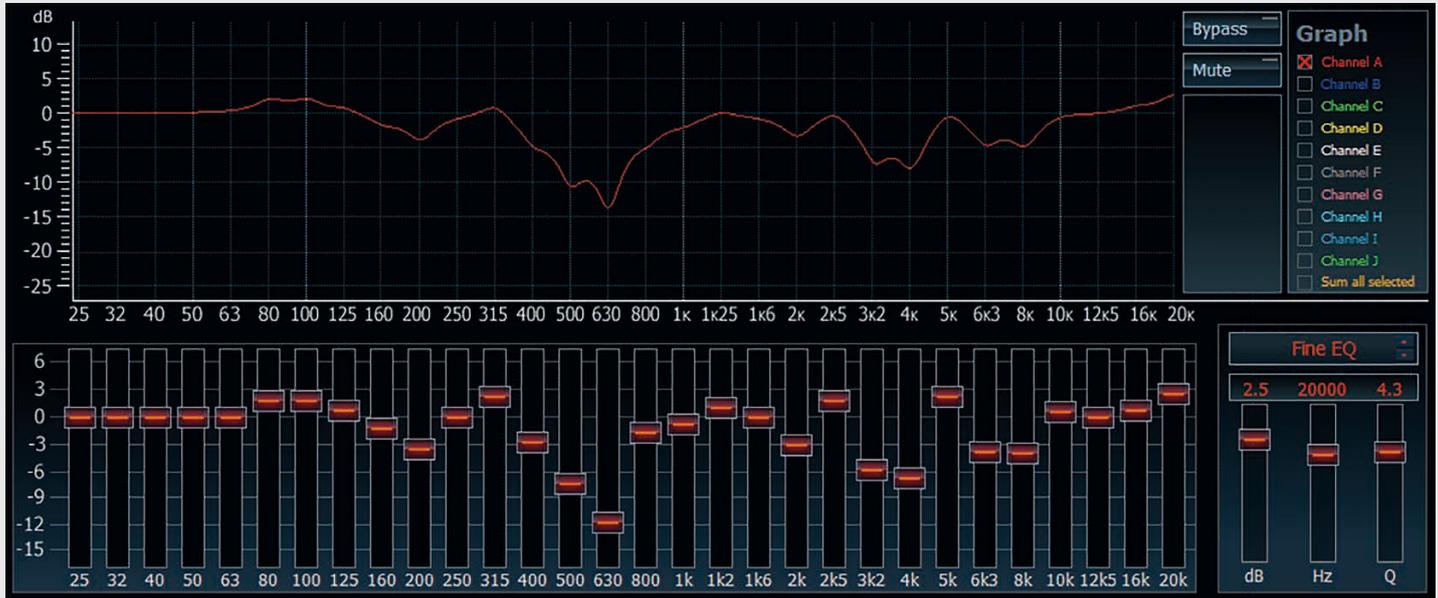
b) The damping factor of the amplifier is not reduced by the resistance of the output

filters. The amplifier is tapping the feedback signal on the speaker terminals whereby the negative impacts of the resistance of the output filters and PCB traces will be eliminated. As a result the amplifier achieves a damping factor of approx. 1,000 over the entire audio range.

c) The distortion caused by the filter components will be eliminated as they are incorporated into the feedback loop.

5.7. 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 indicates the effect of the equalizer on the frequency response.

5.7.1. Functions of the Equalizer

5.7.1.1. 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 each channels. There are two exceptions: the channels F and G of PP 52DSP and E to H of PP 82DSP only 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 shift. This cannot 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.

5.7.1.2. The gain resolution of the equalizer

It is possible to adjust the gain resolution of the equalizer in the Device Configuration Menu (DCM).

The standard value is "0.25 dB" but can be changed to "0.5 dB" or "1 dB" steps. Actually in practice "1dB" steps are the best choice for most use cases.

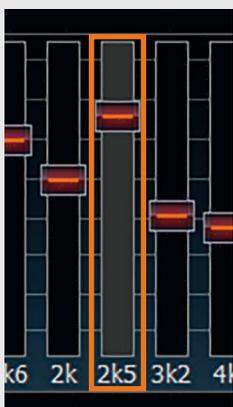
Important

Always choose your EQ gain resolution before you use the EQ. If you change the resolution later on, it might happen that your EQ settings will be modified.

5.7.1.3. The "Fine EQ" feature

Though the equalizer already has a very practicable 1/3rd octave resolution it is possible to fine-tune each single band individually if even an higher resolution is required.

The "Fine EQ" feature allows varying center frequency and the Q-factor in very narrow steps.



To fine-adjust the selected band, click with the mouse on the gain slider of the specific band. This will then be highlighted in grey.

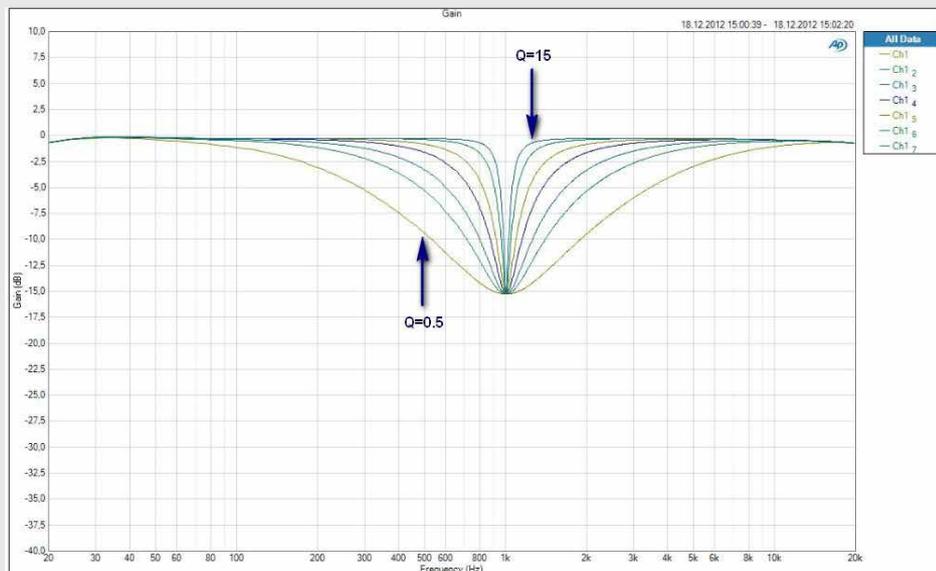
Here's an example:

The center frequency of the 1 kHz band can be raised up to 1,150 Hz. The next "1.25kHz" band can be lowered to 1,175 Hz only. So it is not possible to set two EQ bands to exactly the same center frequency.

The "Q" slide control is used to adjust the bandwidth of the frequency boost or cut. A low Q-factor impacts a wide frequency range while a high Q-value allows eliminating very narrow flaws in the frequency response. The Q-factor can be varied from 0.5 to 15.0 in 0.1 increments - its effect is illustrated in the diagram below.

The center frequency can now either be lowered up to 1/12th octave or increased up to 1/6th octave in 1/24th octave steps.

Anyhow, an overlapping of frequency bands is not possible.



The graph shows the effect of different Q factors of an EQ band. In this case the center frequency (here 1 kHz) and the gain (here -15 dB) remain unchanged.

5.7.1.4. The “Parametric EQ” feature

The “Parametric EQ” feature is quite similar to the “Fine EQ” function but allows an even more comprehensive adjustment.

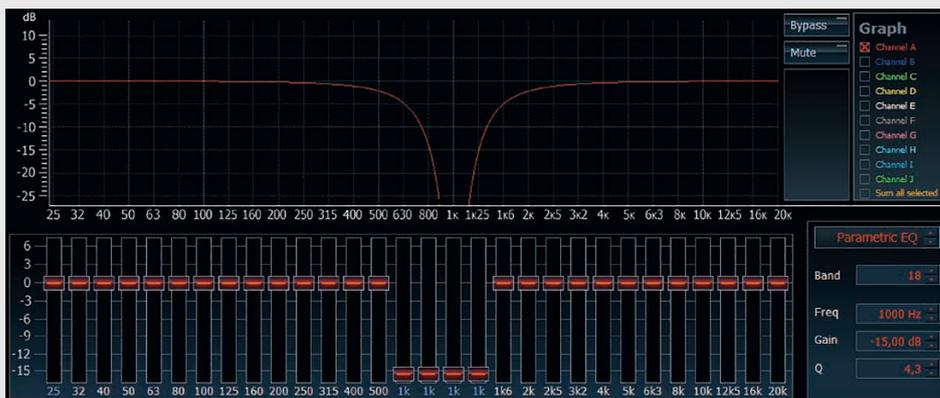
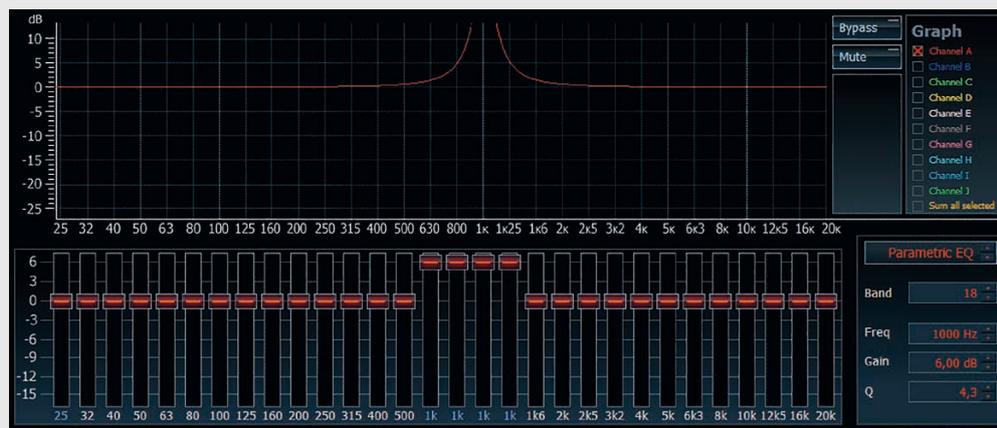
Actually it allows modifying the center frequency in 1 Hz-steps. This can be done either by using the up / down keys on your keyboard or hitting the small up / down arrows next to the center frequency display. Finally you can even enter the value directly – just highlight the frequency figure and type in the required value.



In contrast to the “Fine EQ” here you won’t find any limit for the frequency range of each band as long as you enter numbers between 20 Hz and 20,480 Hz, means that each band can be set to a frequency within the complete frequency range.

Important

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 image on the right side 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!

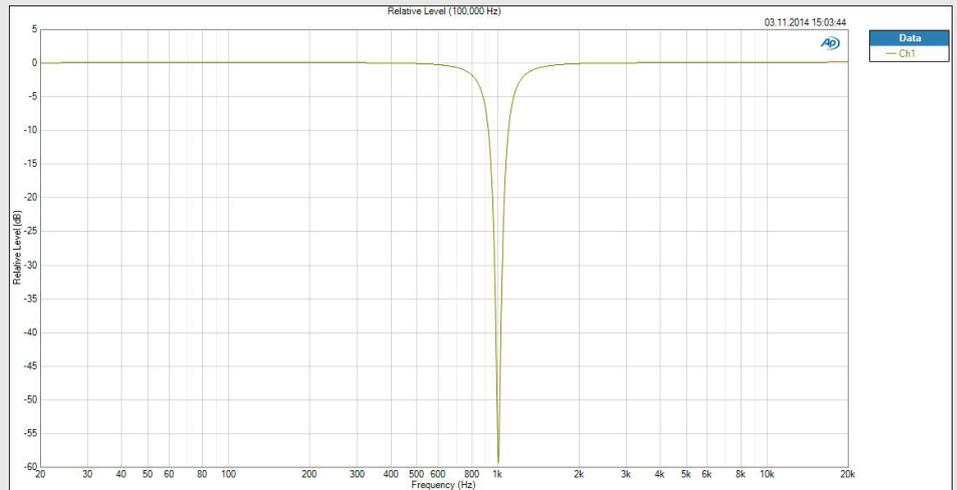


On the other hand it isn’t critical if you choose the same center frequency for several bands to cut a specific frequency range. If multiple bands are set to the same center frequency, it will be possible to realize stronger attenuations than only one EQ band can do as the attenuations will be added.

5 Sound setup – 5.7. Equalizer

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.

On the right side you can see an extreme example where four EQ bands have been set to 1 kHz, $Q = 15$ and Gain = -15 dB. The result is a narrow -60 dB dip in the response curve.



5.7.1.5. The “Shelf EQ” feature

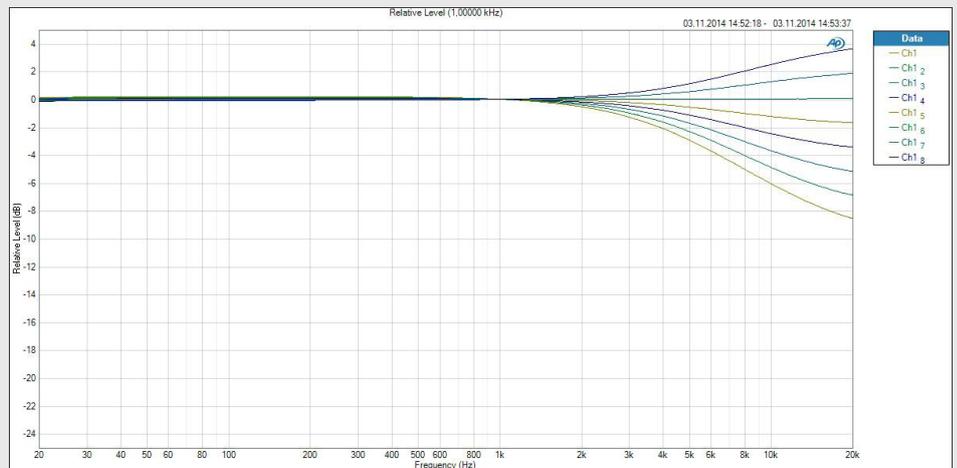


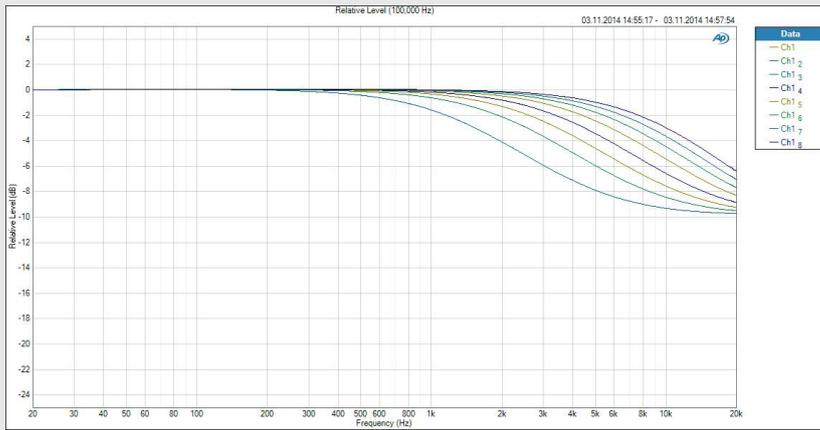
The first EQ band (20 Hz) and the last EQ band (20 kHz) additionally offer a third, the so-called “Shelf EQ” function. The first EQ is a so-called LowShelf-filter whereas the last band act as a HighShelf-Filter.

Shelf EQ's are often used when you like to boost / cut a wide frequency

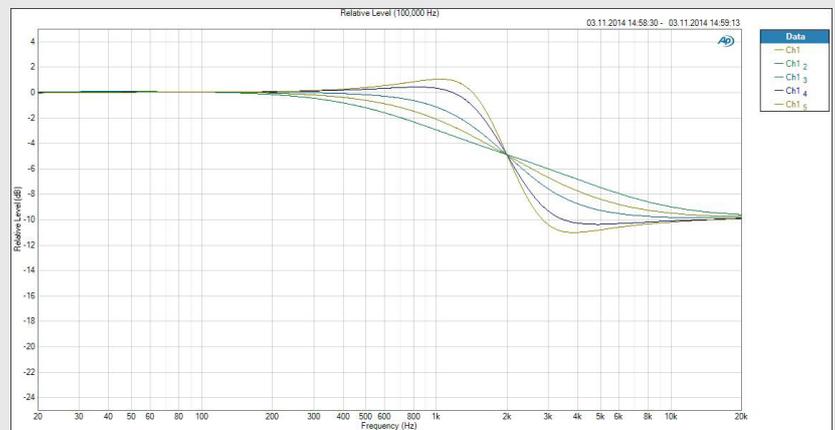
range in the bass or treble region. Tone controls in car radios typically have a very similar behaviour like a shelf EQ.

Below you can see an example for a highshelf filter with a corner frequency of 5,000 Hz, a Q factor of 0.5 and different gain settings:





The graph on the left side shows the same highshelf filter, but with a constant Q factor of 0.5, -10 dB gain and different corner frequency settings. Finally below you see the effect of varying the Q factor, leaving corner frequency (2 kHz) and gain (-10 dB) constant.



Warning

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.

5.7.1.6. 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 only have 15 frequency bands in 2/3rd octave steps instead of 30 bands in 1/3rd increments. Nevertheless each band can

be precisely fine-tuned in exactly the same using the "Fine EQ" or the "Parametric EQ" functionality which already has been described before in detail. The first and the last band can also be used as "Shelf EQ's". So even this "re-

duced" EQ covers the whole frequency range and will be more than sufficient to eliminates all flaws in the acoustical frequency response of your system.

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

5 Sound setup – 5.7. Equalizer

5.7.2. Equalizing

The target of a properly adjusted EQ should be a smooth overall frequency response.

To optimally adjust 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

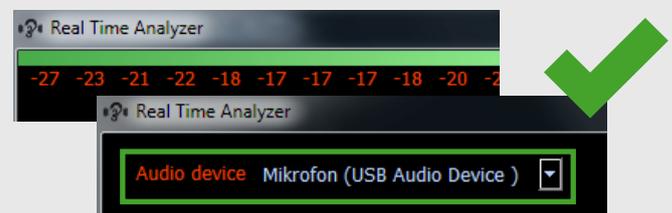
absolute professionals are unable to make perfect adjustments by solely using their ear.

Thankfully, good measuring equipment is affordable and the investment

is definitely worth. The following steps show how to do a frequency response in your vehicle.

5.7.2.1. Adjusting the front speakers

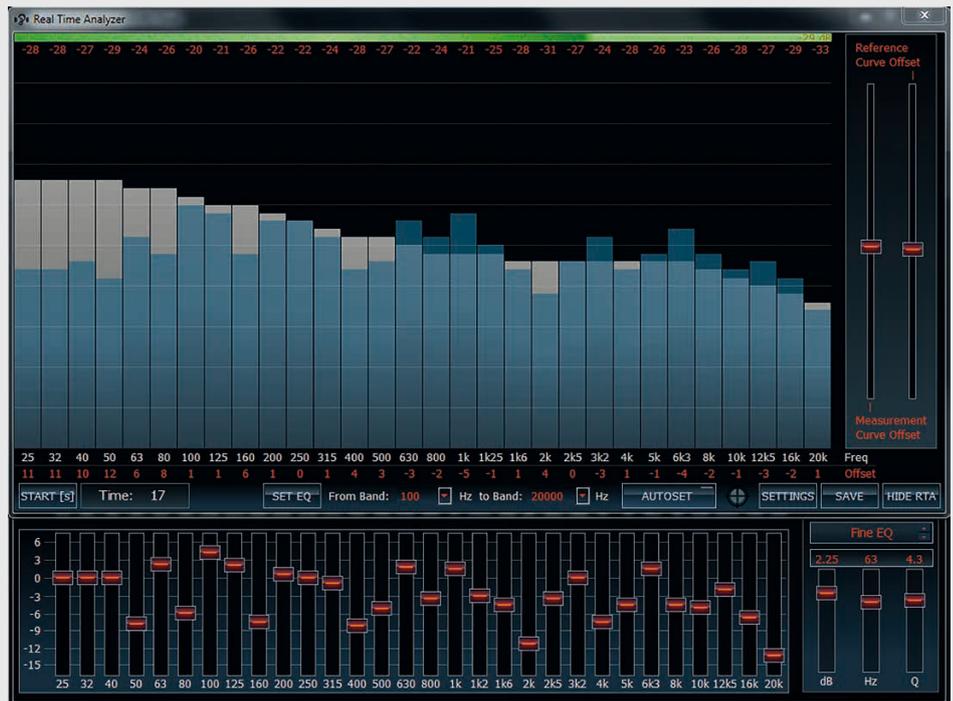
1. Mute all channels except the front channels and link the front channels.
2. Sit in the driver's seat with the measuring microphone and place your notebook / netbook on the passenger seat. Ensure that the screen is not positioned directly in front of any loudspeaker as this will influence the measurement.
3. 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 a closed garage while you are making the measurement that traffic noise does not affect the readings.
4. Check if your microphone is the selected device and if the sound level of your audio equipment is sufficient for the measurement or not (the bar should light up green). See chapter 4.3. / 4.4. for further information.



5. Hold the measuring microphone upright and move it slowly in a semicircle between your left and right ear back and forth. To start the measurement on your PC press the "START" key on the RTA software or simply press "s". The ATF Real-Time Analyzer will now do a first reading which takes 20 sec.

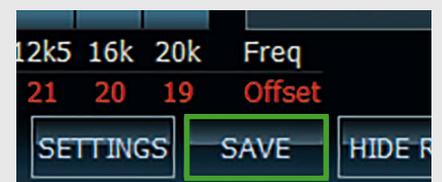


6. You can then identify the determined frequency response of the front speakers together with the reference (target) curve on your PC.



Expert advice:

Save the result of every measured frequency response curve for your own documentation. This allows you to display the result of the adjustments graphically. To save the result as a .png file just push the button "SAVE".



(continued on the next page) →

5 Sound setup – 5.7. Equalizer

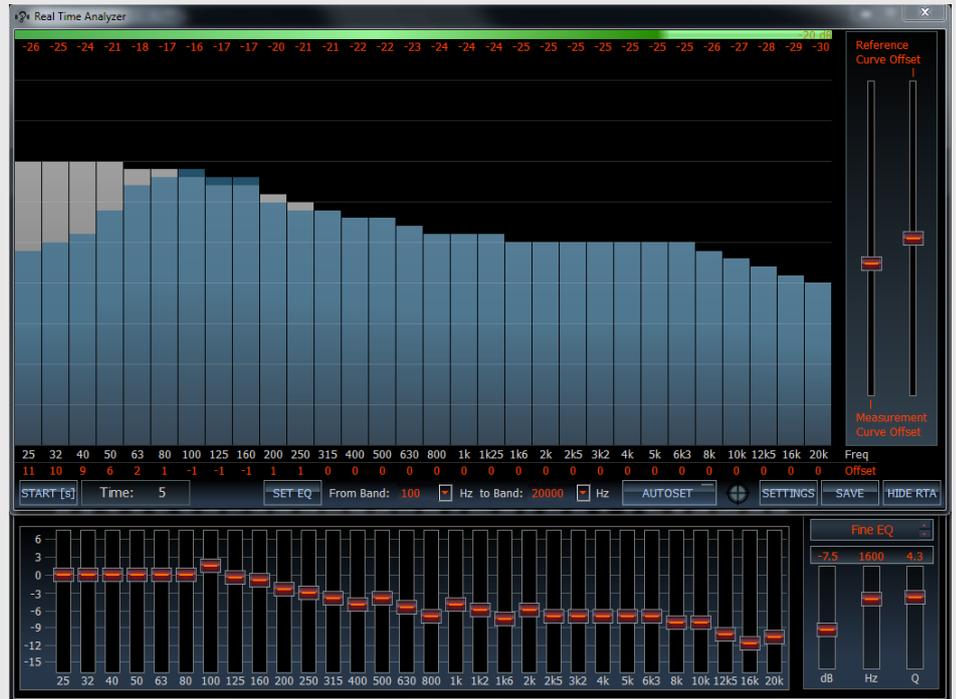
(continued from page 55)

- 7. Now you have the choice – either you use the equalizer manually to adjust the frequency response that it matches the reference curve as close as possible or hit the "SET EQ" key or the "AUTOSET" button.

How to adjust the frequency response by using the "SET EQ" or "AUTOSET" function is described in chapter 5.7.3.

Note

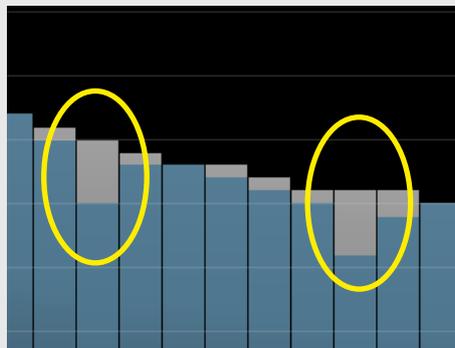
This is an iterative process which may require several measurements / adjustments until you reach the grey target curve.



Deep narrow dips

Don't try to eliminate deep narrow dips in the frequency response (areas marked in yellow)! These dips are mainly a result of cancellation due to phase inversion and cannot be properly removed.

In many cases the human hearing will not recognize such cancellations.



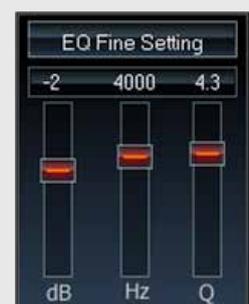
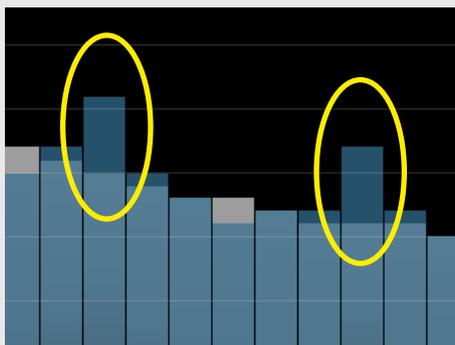
Advice:

If the resolution of the graphic equalizer itself isn't sufficient, make use of the "Fine Setting" function for further optimization.

Strong narrow peaks

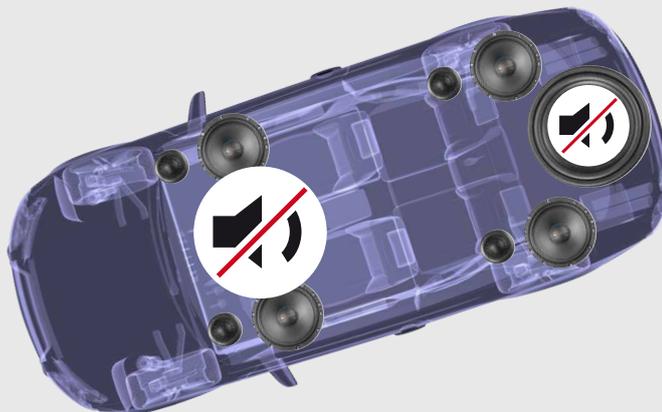
Strong narrow peaks in the frequency response should be definitely eliminated as the human hearing is very sensitive to such acoustical flaws.

You can find further information in chapter 5.7.1.



5.7.2.2. Adjusting the rear speakers

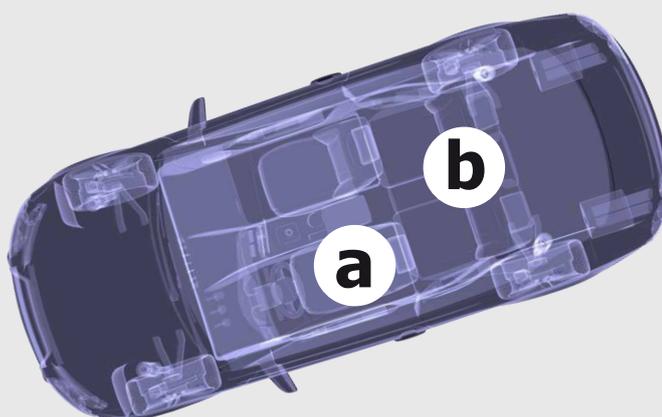
8. After the equalizing for the front channels has been finished, mute them and demute the rear channels. Don't forget to link the two rear channels if you don't adjust them separately. The subwoofer remains muted.



The right microphone position for measuring the rear speaker

There are two different approaches to perform the measurement process:

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 typically only travel alone. The rear loudspeakers usually sound very unpleasant in the rear seats if they have been optimized for the front listening position.



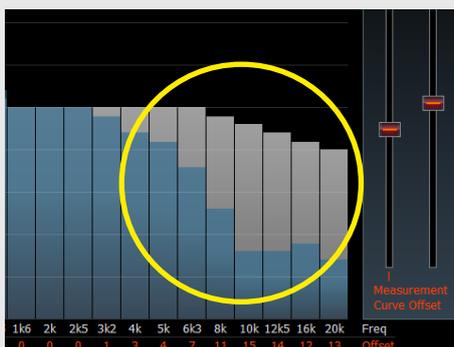
b) Alternatively, in vehicles with four or more seats sit in the center position of the rear seat bench.

Audiotec Fischer uses this procedure to create the vehicle-specific setups for the MATCH amplifiers.

9. Now perform the measurement process as described before (see chapter 5.7.2.1.) in the same way for the rear loudspeakers.

10. Adjust the frequency response. In most cases it is hardly possible to achieve a full approach to the target curve for the rear speakers.

Extreme boost of selected frequency bands with the equalizer should be avoided at all times!



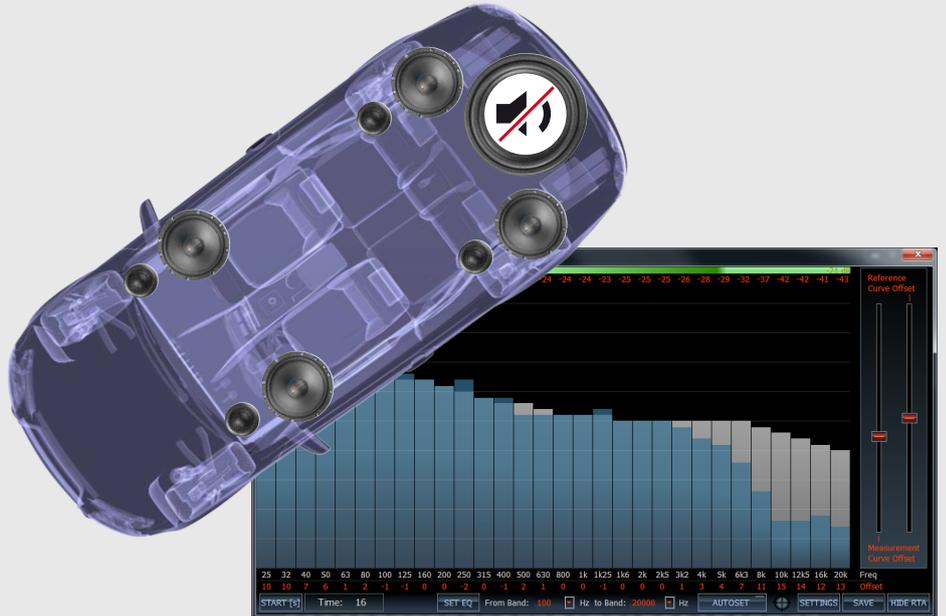
Note

Too much treble response of the rear speakers in most cases will have a negative impact on a proper front staging.

5.7.2.3. Checking the frequency response front and rear playing together

11. Take place on the driver's seat again.
12. Demute front and rear speakers (subwoofer channel remains muted) and do another frequency response measurement.

Normally it is necessary to do some fine adjustment of the EQ's when front and rear speakers are playing together. You can find further information about equalizing in chapter 5.7.1.



Hint

If you detect strong differences in the low frequency response then probably you notice a significant phase shift. It might be necessary to switch the phase of either the front or the rear speakers.

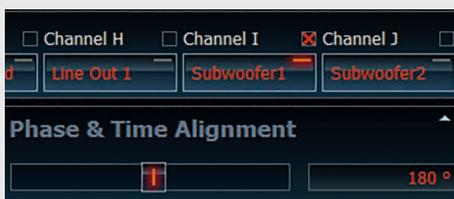


5.7.2.4. Adapting the subwoofer acoustically

13. Finally reactivate the subwoofer.
14. Make another frequency response measurement.

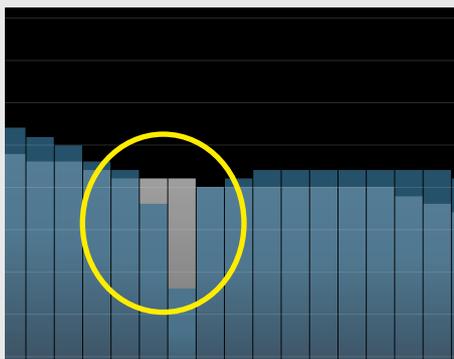


15. Vary the phase and output level of the subwoofer channel that you achieve a smooth transition in the frequency response from subwoofer to the other speakers, even without the use of the EQ.



Strong dip

A strong dip in the range of the crossover frequency indicates inadequate phase relationship between the subwoofer and the other channels. If you don't achieve a flat frequency response using the phase shift function finally try to compensate the flaws using the EQ of the subwoofer channel.



5.7.2.5. The final sound tuning

16. Now – after all these measurements and adjustments – do a first acoustical check by using music tracks 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!



Important

After all settings are done you have to press the button <Save & Store> to save your setup on your PC and 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!

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 & Store>).



5.7.3. The “SET EQ” and the “AUTOSET” function

Both the “SET EQ” and the “AUTOSET” features help you setting up your equalizer within minutes.

These fantastic tools are not a typical “auto-calibration” process which might be known from home theatre AV receivers where you just place the microphone, hit a button and wait for the processor to do the whole set-up of all channels including highpass and lowpass filters automatically.

Frankly spoken, we aren’t happy with all these calibration routines because they can easily lead to very strange acoustical result which might be even dangerous for your speakers. Especially in a car the conditions for such algorithms are even more tricky compared to home applications. Therefore Audiotec Fischer decided to go its own way and define a “semi” self-tuning process which deliberately requires more interaction from you in order to achieve best possible results.

Never fear – it’s definitely quite easy to handle and in the following we will go through this process step by step.

Here are the preconditions before you start “SET EQ” and “AUTOSET”:

- You have made all your adjustments in the “DCM” menu
- You have finished the input-to-output routing in the “IO” menu
- The time-alignment is already done
- You linked all channels that you like to handle equally
- You have found reasonable settings for the highpass and lowpass filters for all channels
- All your measurement equipment (microphone, preamplifier, etc.) has been properly setup as described before

If all this has been done and checked you can start with equalizing the frequency response.

Typically you start with the front speakers, means that the rear speakers and the sub have to be muted.

Now start the initial measuring process by hitting the “START” button or simply by pressing the “S” key on your keyboard. Remember to slowly move the microphone in a semi-circle between the left and right ear during the whole measurement time of 20 sec.

Note

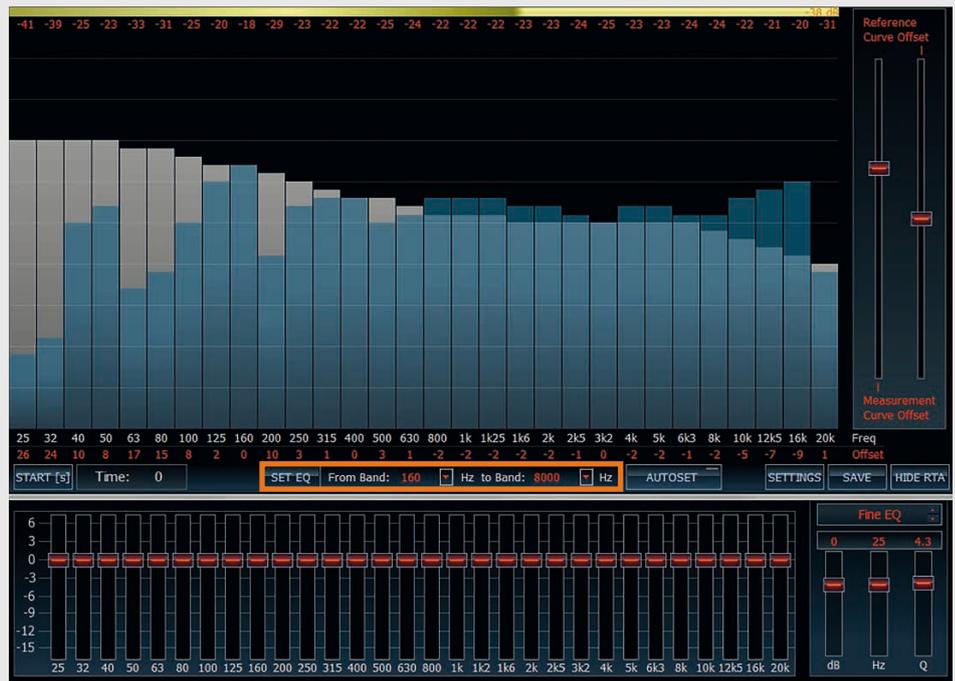
“SET EQ” and “AUTOSET” will only influence the channel that is visible in the “Main” menu! So if you want to adjust channel “A” then choose channel “A” in the main menu. If you like to adjust channel “A” and “B” simultaneously then both channels have to be linked first.



5.7.3.1. Example for an initial measurement

Two things are now important for a correct operation of the "AUTOSET" function – so never intend to skip this first measurement!

- a) Define a reasonable frequency range for the auto-correction of the equalizer. It does not make sense to try compensating the whole audio band from 25 Hz to 20 kHz as this normally will lead to strong boosts and cuts in the EQ settings. In the example above it would be useless to eliminate the significant dips below 250 Hz as well as the strong roll-off in the 20 kHz band. So the EQ adjustment in this specific case has been limited from 250 Hz to 16 kHz.



- b) Move the measured (blue) curve (using the sliders on the right side) so that the gain of most EQ bands has to be reduced therefore avoiding strong boosts as good as possible.

Now you have the choice – either you hit the "SET EQ" key or the "AUTOSET" button.

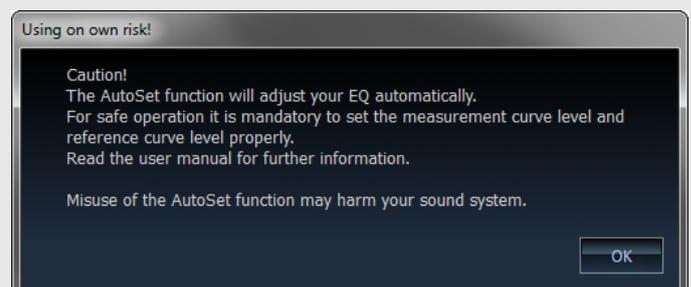
The "SET EQ" provokes that the marked "deviation" values of all defined EQ bands will be automatically transferred to the EQ setting:



You now have to run manually another measurement to check whether the response curve will already meet the target curve. If there is still a deviation, just press "SET EQ" once again and the modified "deviation" values will again be transferred to the EQ. After a few iterations you should be able to achieve an excellent result.

The "AUTOSET" is very similar – the only difference is that it starts an automatic loop of "measuring" and "EQ adjustment".

Once you press the "AUTOSET" button you will notice a warning advice that has to be confirmed before the adjustment itself starts.

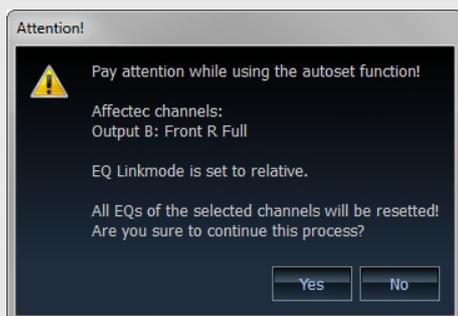


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5 Sound setup – 5.7. Equalizer

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→ Then a second advice pops up, telling you which channels has been chosen for calibration. Carefully check that these definitely are the channels you like to adjust!

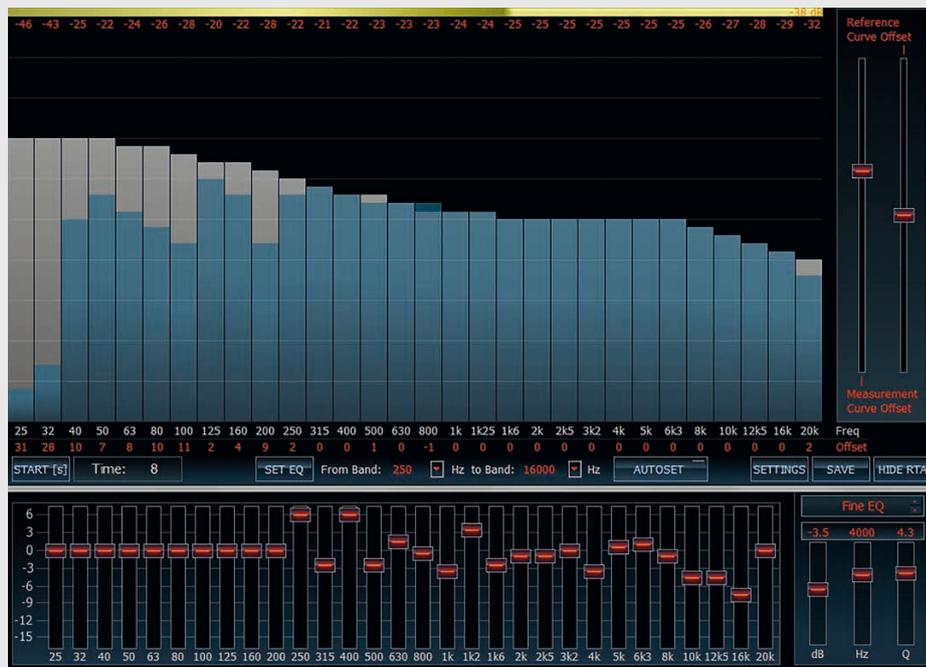


As said before this is an iterative process which means that the EQ bands will be automatically set, followed by a new measurement (limited to 10 sec duration), followed by another re-adjustment of the EQ's, followed by another measurement and so on.

The procedure doesn't stop after a certain time but has to be aborted by you manually if you think that the frequency response curve is close enough to the target curve. Typically this requires 4 -5 iterations.

Keep in mind that you have to move the microphone in a semi-circle during the whole time of the iteration process!

The final result could look like our example below – the response curve in the specified frequency range of 250 – 16,000 Hz is almost 100 % identical to the target curve:



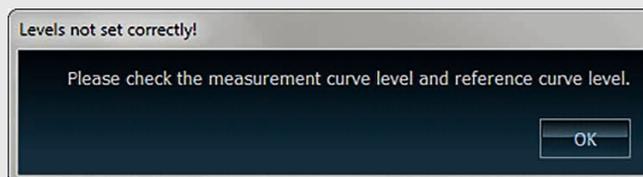
Note

The EQ adjustment algorithm includes a protection feature in order to avoid critical setups as good as possible. So if it is detected that several adjacent bands require full boost (+6 dB) or maximum cut (-15 dB) an error message appears:



Expert advice:

The results of the "AUTOSET" function are a very good start for the acoustical fine tuning which should be done in combination with well-known, high quality music material and the most sensitive measurement tool – your ears!



In this case the process will be aborted. Nevertheless always carefully watch the automatic EQ settings for irrational adjustments!

After the front channels have been adjusted proceed with the all other channels the same way.

6. Supplement

6.1. Special tricks

Today everybody typically uses a mouse or touchpad to make all the adjustments in a software program. That's quite comfortable when you are at your desk, but could be challenging

when sitting with a large notebook in a very narrow compartment like a car.

For that reason we did not only implement the possibility to use the left / right / up / down arrow keys for ad-

justing several features but also defined other keys for direct access of several functions.



You can switch between the different setting sections just by hitting the appropriate key on the keyboard. With the four arrow keys you can do the settings.

Here are the direct key access codes:

Window	Key	Section
Main	h	"Highpass Filter Section"
	l	"Lowpass Filter Section"
	p	"Phase & Time Alignment" section
	t	"Phase & Time Alignment" section
	e	Equalizer
	g	"Channel Gain & Output Level" section
	s	Open "SAVE" window

Here's an example:

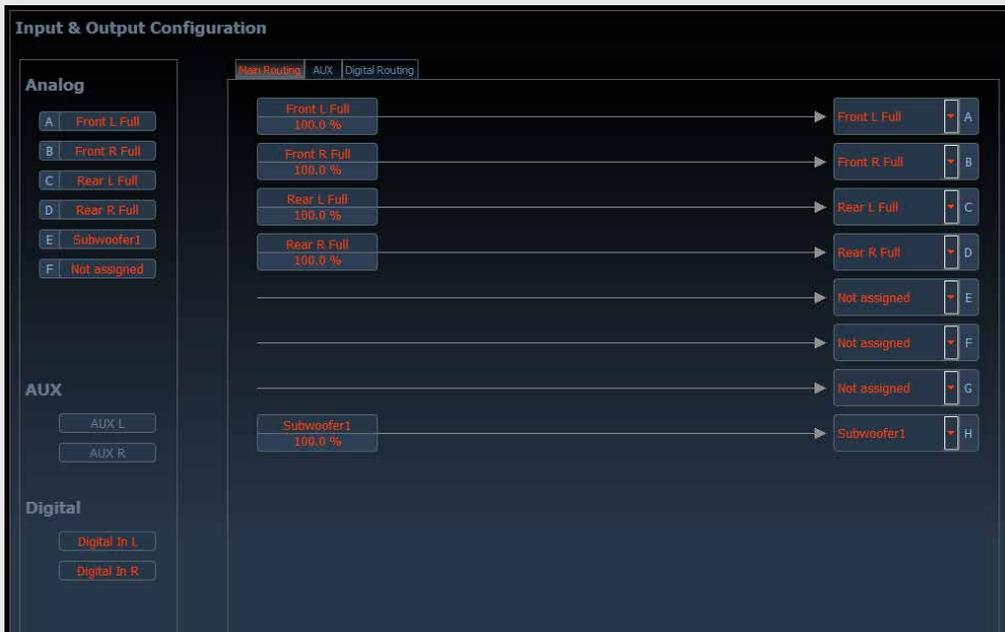
You like to adjust the highpass filter for a specific channel only by using the keyboard.

First press "h" to enter the highpass filter section – notice that the frequency slider will then be highlighted in grey.

Now you can use the four arrow keys to do all settings of the highpass filter without the need to touch your mouse or touchpad. In the same way you can switch to other sections just by hitting the appropriate key on your keyboard.

6.2. Examples for channel routing

a) HELIX DSP – car radio with 5-channel preamp output



Configuration

Radio

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

Speaker

Speaker configuration: full-range 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 rear the outputs A-D + H. This is why you can find in the routing matrix only one edited input field per output channel.

b) HELIX DSP – car radio with 4-channel highlevel output



Configuration

Radio

Car radio with 4-channel pre-amp 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", full-range 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) HELIX DSP – car radio with digital output



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.

Configuration

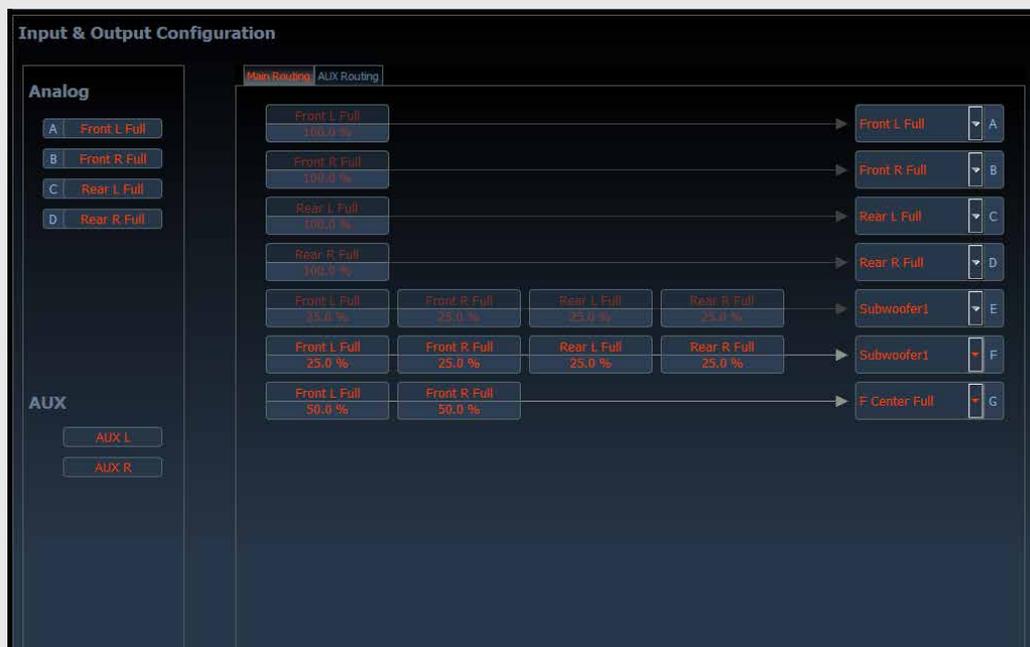
Radio

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.

d) PP 52DSP – adding a separate subwoofer amplifier



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.

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".

6.3. Check list configuring a sound setup

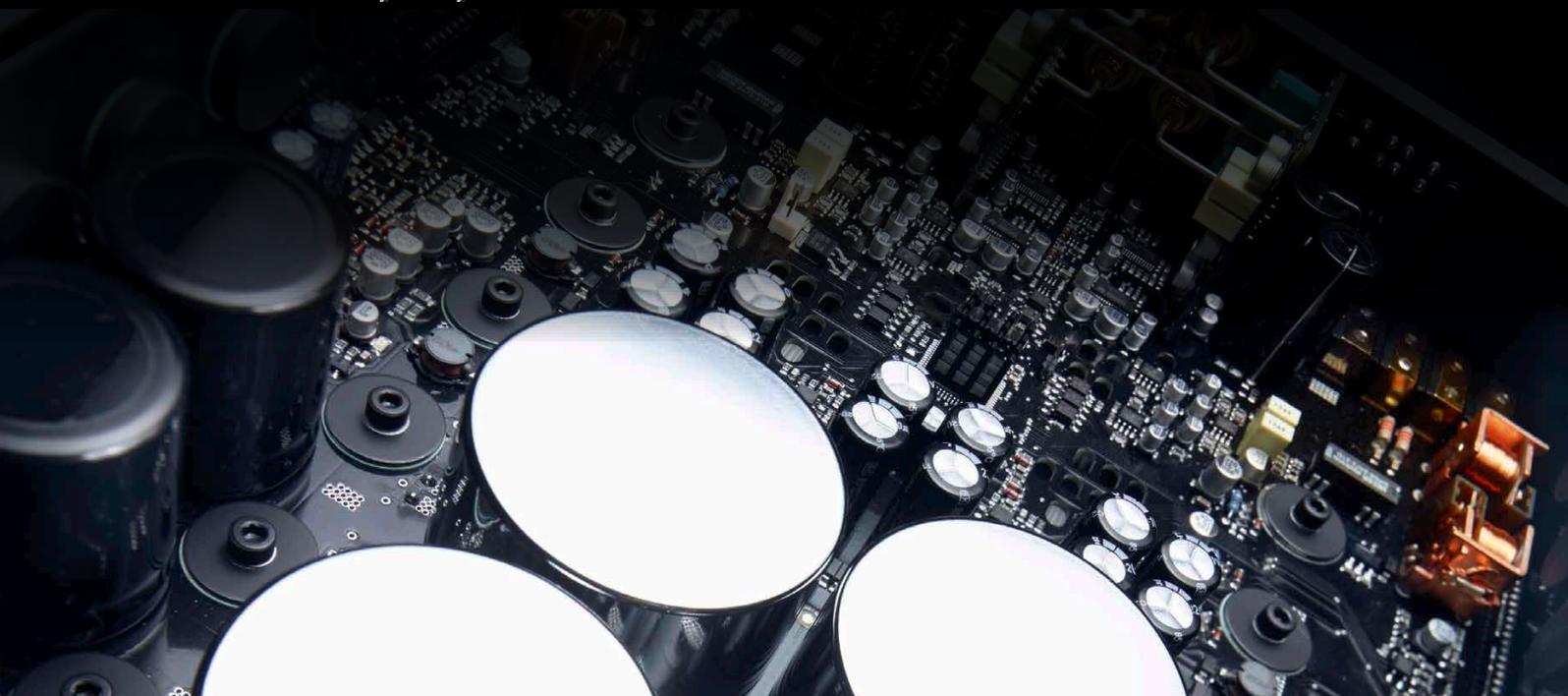
Number	Check item	Done
1.	Place the vehicle in a quiet environment	<input type="checkbox"/>
2.	Connect the measurement microphone to the PC	<input type="checkbox"/>
3.	Prepare the test signal (chapter 1.4. / p. 9)	<input type="checkbox"/>
4.	Set all sound controls of your radio to either "linear" or to the center position	<input type="checkbox"/>
5.	Deactivate any existing loudness functions	<input type="checkbox"/>
6.	Adjust the balance and fader control to the "center position" as well	<input type="checkbox"/>
7.	Connect the DSP device to the PC	<input type="checkbox"/>
8.	Start the DSP PC-Tool	<input type="checkbox"/>
9.	Check if your microphone is the selected device in the ATF Real-Time Analyzer (chapter 4.3. p. 24)	<input type="checkbox"/>
10.	Choose the right reference curve (if you prefer another one as the ATF curve) (chapter 4.3.1. / p. 24)	<input type="checkbox"/>
11.	Make the channel routing (chapter 5.2. / p. 27)	<input type="checkbox"/>
12.	Adjust the time alignment (chapter 5.3. / p. 31)	<input type="checkbox"/>
13.	Check which channels you like to link (chapter 5.4. / p. 39)	<input type="checkbox"/>
14.	Set the high- and lowpass filters (chapter 5.5. / p. 41)	<input type="checkbox"/>
15.	Mute all channels except the front channels and link the front channels	<input type="checkbox"/>
16.	Place your notebook / netbook on the passenger seat. Ensure that the screen is not positioned directly in front of any loudspeaker (chapter 5.7.2. / p. 54)	<input type="checkbox"/>
17.	Start the "pink noise" playback	<input type="checkbox"/>
18.	Check if the sound level of your audio equipment is sufficient (chapter 4.4. / p. 25)	<input type="checkbox"/>
19.	Measure the frequency response of the front speakers (chapter 5.7.2.1. / p. 54)	<input type="checkbox"/>
20.	Use the equalizer to adjust the frequency response that it matches the reference curve as close as possible (chapter 5.7.2.1. / p. 56)	<input type="checkbox"/>
21.	Repeat check item 15 and 16 until you reach the grey target curve	<input type="checkbox"/>
22.	Mute the front channels and demute the rear channels, the subwoofer remains muted. Link the two rear channels if you don't adjust them separately	<input type="checkbox"/>

Number	Check item	Done
23.	If necessary change your position for measuring the rear speakers (chapter 5.7.2.2. / p. 57)	<input type="checkbox"/>
24.	Measure the frequency response of the rear speakers (chapter 5.7.2.2. / p. 57)	<input type="checkbox"/>
25.	Use the equalizer to adjust the frequency response that it matches the reference curve as close as possible (chapter 5.7.2.2. / p. 57)	<input type="checkbox"/>
26.	Repeat check item 20 and 21 until you reach the grey target curve as close as possible	<input type="checkbox"/>
27.	Take place on the driver 's seat again	<input type="checkbox"/>
28.	Demute front and rear speakers, the subwoofer remains muted	<input type="checkbox"/>
29.	Make a measurement of front and rear speaker playing together (chapter 5.7.2.3. / p. 58)	<input type="checkbox"/>
30.	Do a fine adjustment of the EQ 's	<input type="checkbox"/>
31.	Reactivate the subwoofer	<input type="checkbox"/>
32.	Make another frequency response measurement	<input type="checkbox"/>
33.	Achieve a smooth transition in the frequency response from subwoofer to the other speakers (chapter 5.7.2.4. / p. 58)	<input type="checkbox"/>
34.	Final sound tuning by listening to your favourite music or one of our recommended sound tracks (chapter 5.7.2.5. / p. 59)	<input type="checkbox"/>
35.	Transfer all parameters to the amplifier 's / processor 's internal memory	<input type="checkbox"/>
36.	Store your setup on the harddrive of your PC	<input type="checkbox"/>

BRAX

HELIX

MATCH



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