

 ENGLISH



CAR AUDIO SYSTEMS



ESX TOOLKIT

USER MANUAL

VERSION 1.0

Table of Contents	Page
Introduction	3
Overview ESX DSP Device	3
ESX Toolkit Software	4
Info, Status, Device Identification	4
Demo Mode	5
User Interface	5
Load and Export	6
Reset	6
Presets	7
Language	7
About	7
Master Volume	8
I/O Chanel Assignment (Routing)	10
Output Channels	10
Alias Names	11
Input Channels	12
Routing	13
Expert Mode (Overview)	14
Main Page (Main)	16
Output Equalizer	16
Output Volume	18
High-Pass and Low-Pass Filters	18
Link Mode	19
Phase Switch	20
Time Alignment (Delay)	20
Expert Mode (Continued)	20
Staging Layer	21
Staging Link Mode	21
Dynamic Loudness Adjustment (DLA)	21
Input, Output and Staging EQ	23
Time Alignment (Delay)	24
Delay Adjustment	25
Distance Measurement	25
Delay Groups (Link Mode)	25
Meter	26
Level Indicators	26

Congratulations on Purchasing Your New ESX DSP

Dear Customer,

We are thrilled that you chose our product and would like to sincerely thank you for your trust. Your ESX DSP (Digital Signal Processor) has been designed to provide you with an exceptional user experience and top-tier performance. With its advanced technology and comprehensive features, you can be confident that you are getting the best possible audio quality and adaptability for your needs.

This manual serves as a comprehensive guide for setting up and using your new DSP. It offers detailed step-by-step instructions and important information to help you get started and ensure that you can make the most of your product.

Before you begin, we recommend thoroughly reading this user manual. This will allow you to familiarize yourself with the features and capabilities and ensure you use your DSP to its full potential.

We wish you great enjoyment with your new product!

Which Devices Does This Manual Apply To?

This user manual has been specifically created for the setup and use of the following ESX DSP models:

- **D66SP**
- **QM66SP**
- **D68SP**
- **QL810SP**
- **QL812SP**
- **QE812SP**
- **VE900.7SP**
- **VE1000.6SP**
- **VE1300.11SP**

It contains all the essential information to help you unlock the full potential of your device. Regardless of the model you have chosen, you will find all relevant instructions and tips here.

ESX Toolkit DSP Software

Your ESX DSP can be easily configured using the accompanying DSP software. The software provides a clear and user-friendly interface that allows you to customize all functions according to your preferences.

Before connecting your DSP device to your computer or laptop for the first time, visit our website and download the latest version of the ESX Toolkit. We also recommend regularly checking for updates to keep your DSP up to date.

You can find the ESX Toolkit and the corresponding manual on www.esxaudio.de/english/

Using the DSP Software

Installing the ESX Toolkit software is essential before you can use and configure your DSP device. Ensure that your DSP device is connected to your laptop before starting the software. The included USB cable is generally sufficient to connect any of our DSP devices to your laptop— simply plug it into an available USB port.

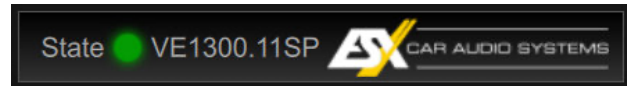
Additionally, your DSP must be powered on for the software to recognize the device and enable communication between them. Your DSP will automatically power on via the car radio, either through the Remote Input (REM IN) or using the Auto-TURN ON feature.

To start the DSP software, click on the corresponding icon on your desktop. The user interface of your DSP will then open.



Info, Status, and Device Name

The green LED indicates that your DSP device is successfully connected. The software automatically detects which DSP device is connected.



Demo Mode

If no DSP device is connected to the software or if the device is not powered on, the user interface will open, but the status LED will not light up. This behavior can also occur if the USB drivers are not correctly installed.

The ESX Toolkit software can be run in demo mode. However, in this mode, no changes can be made to the DSP device, and settings cannot be saved to the device. To make changes, you must restart the software once the DSP device is properly connected and powered on. As mentioned earlier, the green status LED indicates that the connection between the software and DSP has been established.

The User Interface

In the top bar, you will find all the key functions for controlling and adjusting your DSP.



This include::

- **Load:** Load previously saved settings from a file.
- **Export:** Export current settings to a file.
- **Reset:** Reset all settings or individual options to factory defaults.
- **Preset:** Select preset configurations.
- **Language:** Change the language of the software.
- **I/O:** Open the settings for signal paths and routing.
- **Main:** Access the main functions.
- **Time:** Set the time alignment.
- **Meter:** Monitor signal levels with the level meter.
- **About:** Get information about the software and device.

These buttons provide direct access to the main areas and functions required for configuring and using your DSP.

Load and Export

The **Load** and **Export** buttons are located at the top left of the main navigation. These functions allow you to load saved tuning parameters from your laptop or export them to back up your settings or use them on other devices.

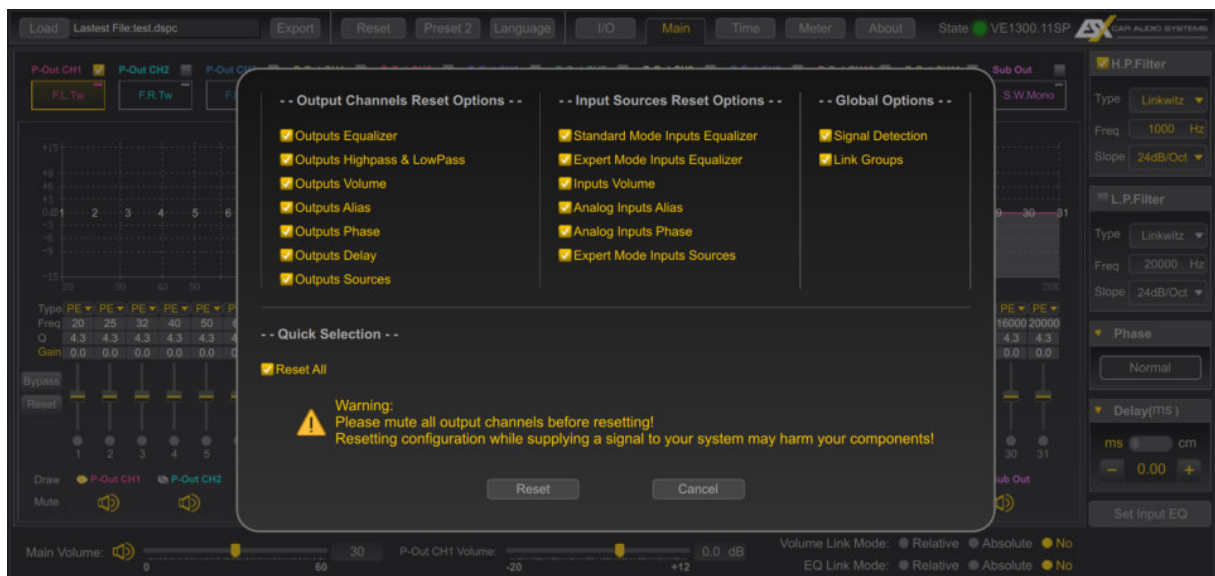
Load: The previously saved **.dspc file** with tuning parameters on the PC can be loaded and edited again.

Export: When exporting the set parameters, you must set a password to protect the exported file. Be sure to remember the password, as the tuning parameter file cannot be reloaded without it.

Reset

If you want to reset the settings of your DSP device to factory defaults, you can do so using the Reset function. This feature helps resolve errors or start a new configuration from scratch. You can choose whether to reset **all settings** or **only specific parts** to the factory default.

To reset, simply click on **Reset** to apply the changes. The reset only applies to the user interface and all set parameters. The presets stored on the device will remain unaffected.



Note: A reset to factory defaults will either reset all custom settings or only specific parameters, restoring the original default values. Be aware that any unsaved changes will be lost! Make sure to save your settings beforehand if you wish to restore them later.

Presets

With the Preset function, you can save, load, or delete different configurations of your DSP as presets. This allows you to access your preferred settings at any time. You can later switch between the saved presets via the DSP remote control or the Toolkit app.

Save: Save the tuned settings directly on the device. The name can be up to 10 characters long and supports only letters and numbers, with no spaces.

Load: Select a preset already saved on the device to use or further edit.

Delete: Remove the selected preset from your DSP.

After each DSP tuning, it is recommended to save the set parameters as a preset to ensure that the settings are not lost due to a sudden power outage.

Language (Language Selection)

The DSP Toolkit software supports multiple languages and allows you to switch between **English**, **German**, **French**, and **Italian**. You can easily select the desired language via the corresponding option in the main navigation.

About

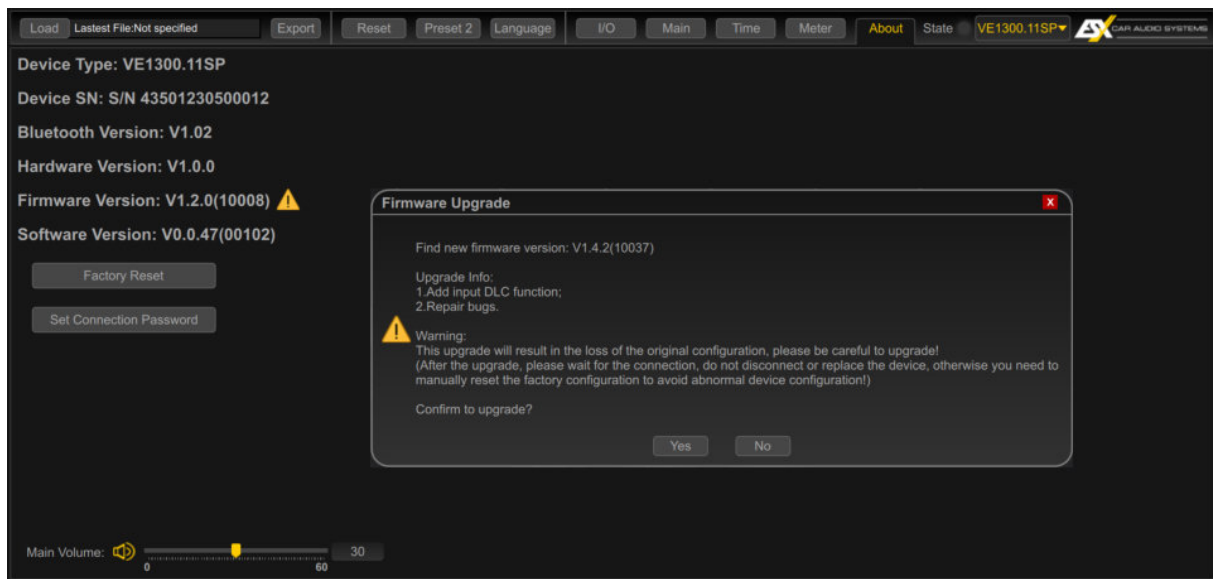
In the top right menu tab **About**, you will find important information about your DSP device, such as:

- **Device type**
- **Serial number (SN)**
- **Bluetooth version**
- **Hardware version**
- **Firmware version**
- **Software version**

Additionally, the functions for **firmware update**, **factory reset**, and **password** are available.

Firmware Update

When a new firmware version for your DSP is available, a notification will appear next to the current firmware version. Click on it to start the firmware update.



Note: A firmware update will result in the loss of the original configuration, including all settings. Please be cautious when updating!

After the firmware update, wait until the connection is fully established. Do not disconnect or replace the device during the process. Otherwise, you may need to manually reset the device to factory settings to avoid potential issues with the device configuration.

Main Volume

At the bottom left of all menu pages, you will find the main volume control. Here, you can adjust the volume of your DSP or temporarily mute it by clicking on the speaker icon.



Note: We recommend reducing the volume to zero or muting it when first starting the software to avoid accidental overloads and potential damage to the connected speakers.

The I/O, Main, Time, and Meter Tabs

In the central section of the main navigation bar, you will find four tabs containing the most important functions for configuring your DSP.



You may have noticed that the user interface automatically opens on the **I/O** tab after launching the software. Next to it, you'll find additional buttons that take you to the **main** page, **time alignment**, and the signal **meters**.

I/O – Configuring Inputs and Outputs

Channel assignment (routing) is managed under the I/O menu. Clicking this button opens the window where you can configure each input and output channel of your DSP. Here, you can:

- Name channels
- Sum input signals
- Assign up to four input signals to each output channel

(For more details, see the section I/O Channel Assignment on page 9.)

Main Page for Central Settings

In the Main tab, you will find channel selection, high-pass and low-pass filters, the equalizer, and other core functions. (For more details, refer to the Main Page section on page 14).

Time Alignment

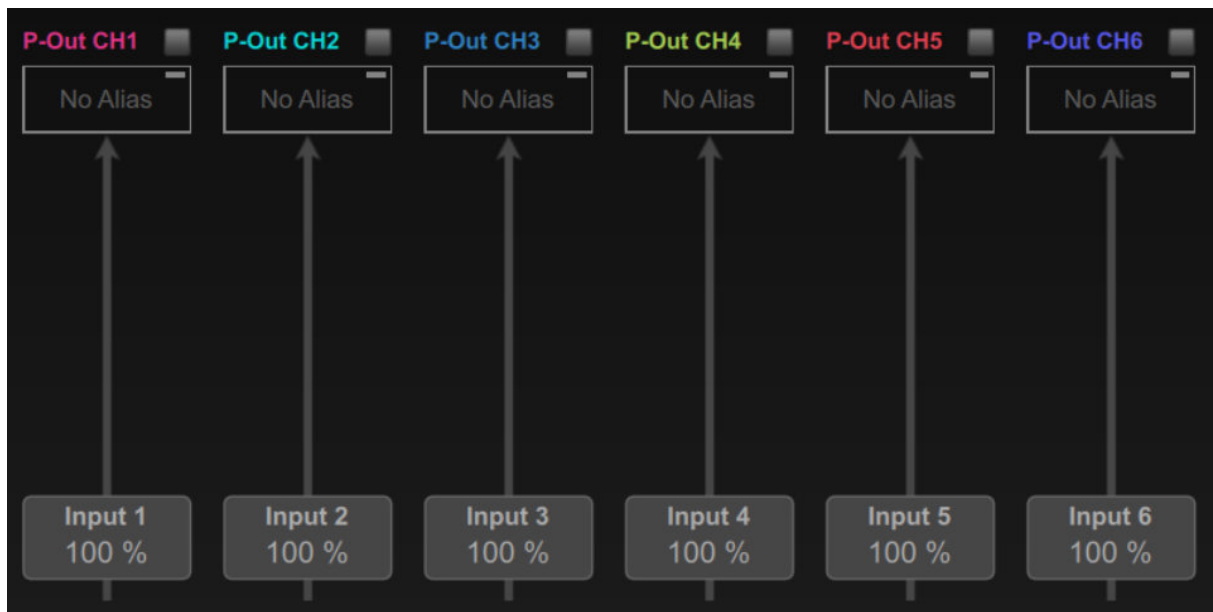
Under the Time menu, you can optimize the timing of your speakers inside the vehicle to create a precise sound stage. (For more details, see the section Time Alignment on page 21).

Meter – Level Monitoring for System Control

Clicking the “Meter” button opens the level meters. Here, you can ensure that your system levels are correctly adjusted. (For more details, see the section Meter on page 23).

I/O Channel Assignment (Routing)

Clicking the **I/O** button opens the channel assignment (routing) section, where you can configure each input and output of your DSP individually. You have the option to assign alias names to channels, merge multiple input signals, and assign up to four input signals to each output.



Output Channels

At the top, you'll see the labels of all available output channels, each displayed in a different color. In our example, these are channels CH1 through CH6. Depending on your DSP model, output channels may be labeled alphabetically (e.g., A-F) instead of numerically. Below each channel label, there is a field where you can assign an alias name to the respective output channel.

Note: The scope and functionality of the channel assignment, as well as the number of available input and output channels, vary depending on the DSP device. The I/O page only displays the channels available for your specific unit.

Click on any output channel to select and edit it. The selected channel is highlighted in color so you can always see which output channel is currently active and being edited.

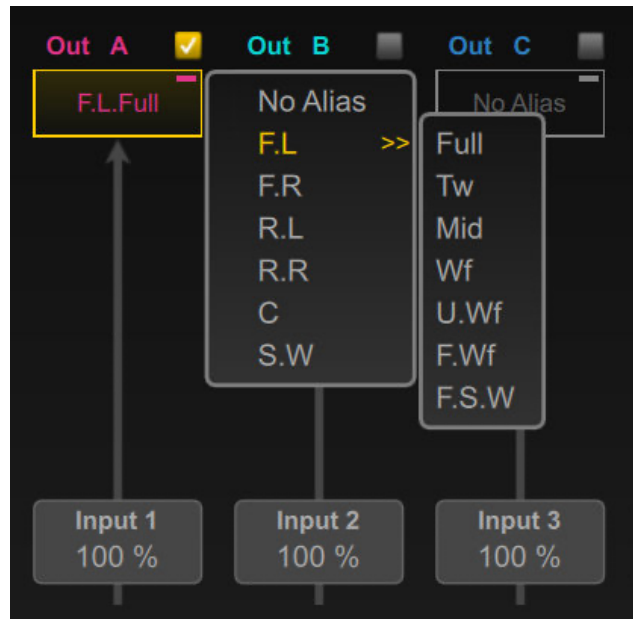
Assigning Alias Names

You can assign an **alias** to each output channel. Right-clicking the corresponding field below the channel label opens a selection menu where you can choose from a predefined list of alias names. Alternatively, you can use the field on the far right of the I/O page to assign an alias to an output channel. It is not possible to enter custom names. Assigning an alias has no functional impact—it simply helps you keep track of your output channels more easily by making them easier to distinguish.

The predefined alias names are to be understood as follows:

- **F.L** Front Left
- **F.R** Front Right
- **R.L** Rear Left
- **R.R** Rear Right
- **C** Center
- **F.L** Subwoofer

- **Full** Full
- **Tw** Tweeter
- **Mid** Midrange
- **Wf** Woofer
- **U.Wf** Underseat Woofer
- **F.Wf** Footwell Woofer
- **F.S.W** Front Subwoofer



Note: Depending on the selected language, the alias names and abbreviations may vary. This allows for more flexible adaptation to different language preferences and usage contexts.

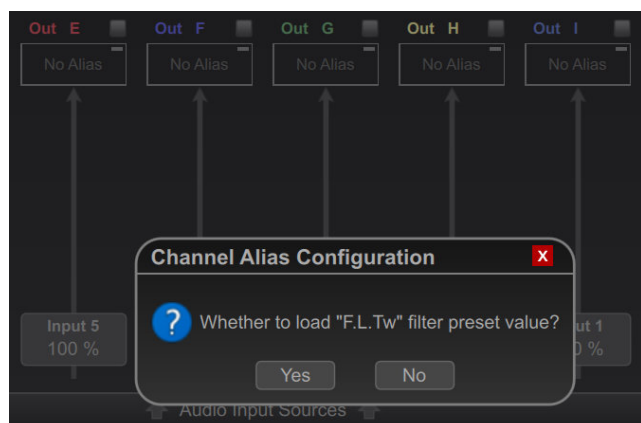
Loading Predefined Filter Settings

After assigning an alias to an output channel, you will be prompted to load the predefined filter settings.

If you select **"Yes"**, the filter settings will be loaded automatically. These include default values tailored to different frequency ranges or speaker types.

For example:

- **Tweeter** 3 kHz – 20 kHz
- **Midrange** 5 kHz – 20 kHz
- **Woofer** 20 Hz – 500 Hz
- **Subwoofer** 20 Hz – 80 Hz
- **Fullrange** 0 – 20 kHz (kein Filter)



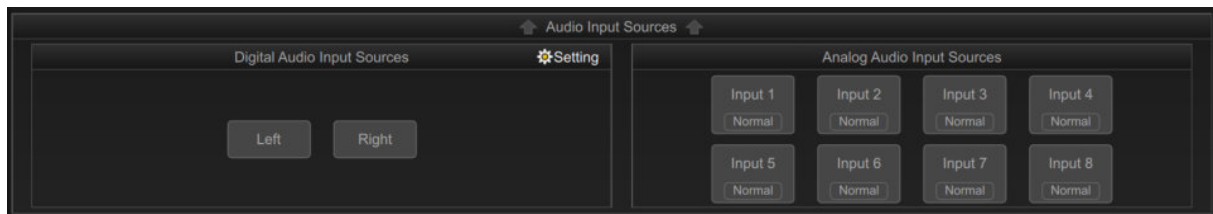
If you select **"No"**, only the alias will be assigned to the output channel, and no changes will be made to the filter parameters. The existing filter settings will remain unchanged.

The predefined filters are intended to protect your speakers by preventing frequencies from being sent to them that lie outside the recommended range for the respective speaker type. This helps to avoid overloading and potential damage.

If you wish to adjust the filter settings later to match your individual requirements, you can do so in the Main Menu, where you can further refine the filter parameters, including filter type and slope.

Input Channels

At the bottom of the **I/O menu** page, you'll find the input channels, which are divided into digital and analog channels. Just like with the output channels, you can assign alias names to the analog input channels to improve clarity.



Right-clicking on an output channel opens a selection window where you can choose from a list of options.

Note: Digital input channels such as Bluetooth or the optical input transmit a stereo signal. This signal is simply split into left and right channels, and alias names cannot be assigned.

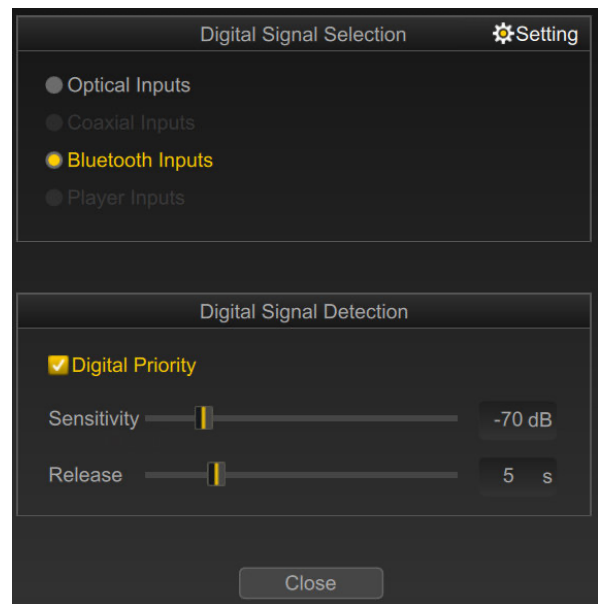
Depending on the DSP and input source used, additional routing windows are available alongside the main routing. This allows you to manage analog and digital inputs separately. The corresponding tabs can be found on the left above the output channels. If you use only a single input source—usually your car's head unit—the main routing is sufficient. Once multiple input sources are present, separate routings become useful. For example, you can set up one routing for the analog high-level input and another for digital Bluetooth streaming to configure the routing accordingly.

Selecting Digital Input Sources

In the digital input settings, you can define and prioritize your input sources. This means the system will automatically switch to a digital input as soon as a signal is detected.

Within this menu, you can choose between different digital input sources. You also have the option to adjust the input sensitivity to balance volume levels and, if necessary, set a delay to ensure precise synchronization with other sources.

If you don't want to miss important vehicle sounds, the software offers an option to mix them in via the routing settings. This allows you to combine digital and analog sources flexibly and create a well-balanced mix of signals.



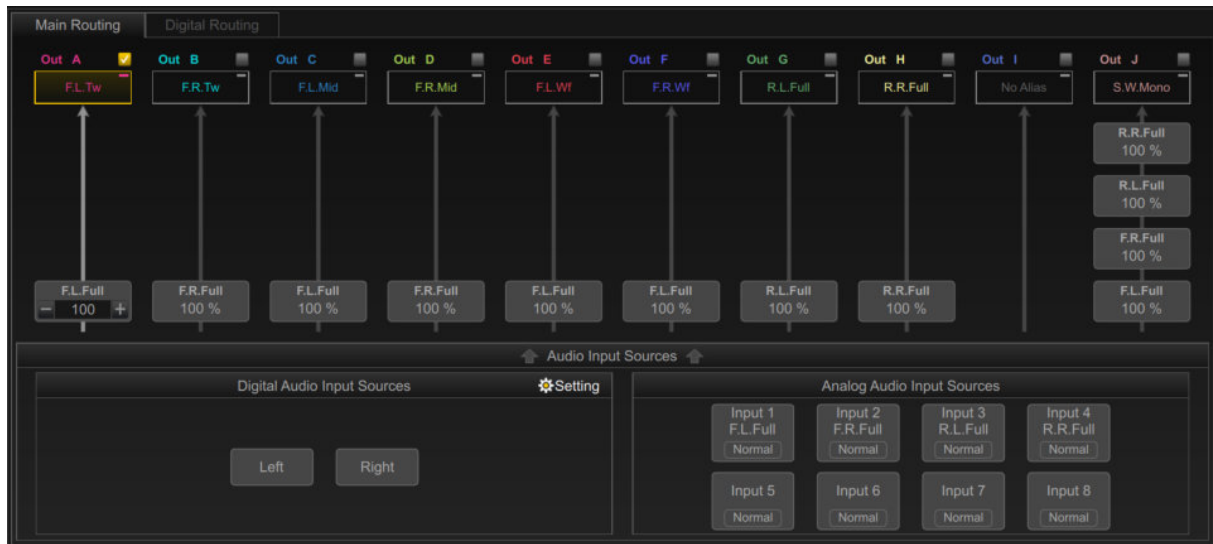
Routing

Routing refers to the assignment of input channels to output channels within your sound system. It allows you to define how different input signals are processed and distributed to the output channels. This is a key element in configuring your DSP, as it ensures that each speaker receives the intended audio signals.

Assigning Inputs to Outputs

You can freely assign input signals to output channels. To do this, simply drag and drop the desired input signal from the input overview at the bottom onto the desired output channel. Keep in mind that each active output channel must have at least one assigned input signal in order to function.

It's also possible to create an output signal by **mixing up to four different input signals**. This feature is ideal, for example, for generating a summed signal for a subwoofer by combining the signals from the front and rear channels.



If you want to remove an input signal from an output channel, you have two options: click the X icon next to the input signal, or right-click on the signal to delete it.

When you assign multiple input signals to one output, they are **weighted equally by default**. The mix ratio is displayed as a percentage. If you want to adjust it, you can use the **plus (+)** and **minus (-)** buttons to change the value, or simply overwrite the percentage by typing in a new value directly.

These flexible routing and mixing options allow you to configure your sound system individually and tailor it to your personal sound preferences.

Expert Mode (Overview)

The Expert Mode in the DSP software provides advanced users with extended functions for precise fine-tuning of their sound system. Please note that not all ESX DSP models offer this feature – **availability depends on the specific model**.

To switch to **Expert Mode**, click the Mode toggle at the bottom center of the screen and change from Standard to Expert. (For more details, see the Expert Mode section on page 18.)

In Expert Mode, the number of analog inputs is reduced from eight to four. Instead, a new channel layer with four so-called **staging channels** is introduced, which sit between the input and output channels.



In Expert Mode, assigning input channels to output channels takes place in two steps. First, the analog and digital input signals are assigned to the four staging channels. Then, the staging channels are assigned to the respective output channels. Even in Expert Mode, it's especially helpful to give all channels alias names to keep track of everything and simplify routing.

A special feature of Expert Mode is that you have access to three equalizers, allowing you to finely tune your sound system.

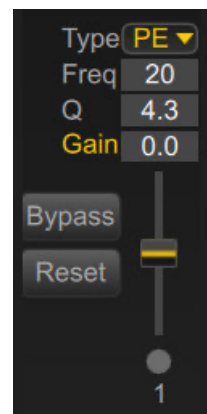
- **Input EQ** The input signal can be processed in the Input EQ to optimize sound quality before further processing. You can find this on the I/O page at the bottom right on the **Set Input EQ** button. This opens a full-featured 31-band equalizer under the **Base EQ** tab.
- **Staging EQ** The sound tuning is done using the Staging EQ for the entire left and right channels. You'll find the corresponding tab directly to the right of the Base EQ. To simplify settings, channels can be linked, allowing levels and EQs to be adjusted either absolutely or relatively for the group. The Staging EQ features a special function called Balance Link. This works like a balance control and, when linking left and right channels, subtracts from one side what it adds to the other. This applies to each individual EQ band, enabling precise fine-tuning of the system's soundstage.
- **Main EQ (Output EQ)** In the main menu Main, you'll find the output equalizer for each individual output channel. Here, the EQ bands are adjusted individually to the specific requirements of the speaker. Once set to linear, these parameters remain unchanged thereafter.

In the equalizer, each of the 31 bands offers various filter types to help you shape the frequency response according to your preferences. Choosing the right filter type depends on the specific requirements of your audio system and your personal sound preferences.

- **Parametric EQ (PE)** This filter type allows you to boost or cut specific frequency bands.
- **High-Shelf-Filter (HS)** A high-shelf filter boosts or attenuates all frequencies above a certain cutoff frequency by the same amount. This type of filter is ideal for increasing or reducing the overall treble of an audio signal.
- **Low-Shelf-Filter (LS)** Similar to the high-shelf filter, the low-shelf filter affects all frequencies below a defined cutoff frequency equally. It is ideal for boosting or attenuating the low end of an audio signal.
- **First-Order All-Pass Filter (AP1)** A first-order all-pass filter alters the phase of a signal without affecting its amplitude. It is often used to correct phase shifts or to modify certain sound characteristics without changing the volume.
- **Second-Order All-Pass Filter (AP2)** This filter type provides a steeper phase shift than the first-order all-pass filter and allows for more complex phase adjustments.

To the left of the **31-band parametric equalizer**, you'll find the **Bypass** and **Reset** buttons. The Bypass button allows you to bypass the equalizer, letting the audio signal pass through unaffected. The Reset button restores the selected EQ setting to its default value..

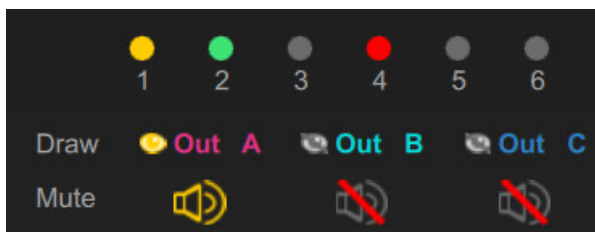
Depending on whether you have selected Type, Frequency, Q factor, or Gain, the corresponding parameters will be reset, allowing you to quickly return to a neutral starting point.



Each of the 31 numbered EQ bands features a color indicator that signals the status of the respective band:

- **Gray:** No adjustment; the band is in its original state.
- **Yellow:** The band has been modified; an adjustment is present.
- **Green:** The band is currently being edited; it is selected for adjustment.
- **Red:** The band is in bypass mode; it is bypassed and does not affect the signal.

These "indicators" allow you to see the status of each EQ band at a glance, making it easier to operate and adjust the parametric equalizer.



Below the graphic equalizer, you'll find yellow icons for each output channel that let you show, hide, or mute the respective channel within the equalizer.

Output Volume (Channel Volume)

The volume of each output channel can be adjusted independently by entering the value directly, using the left and right arrow keys on the keyboard, or dragging the slider with the mouse. The reference point for overall volume is 0.0 dB. The adjustment range spans from -12 dB to +12 dB in 0.1 dB increments.

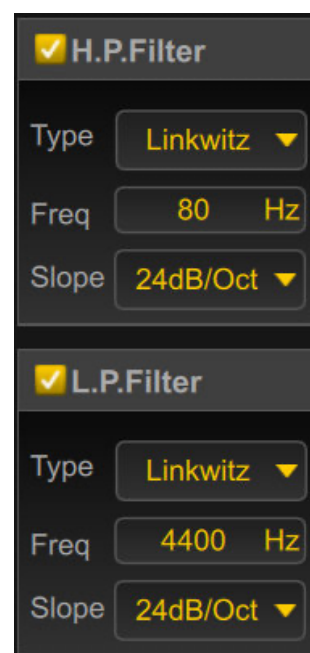
High-Pass and Low-Pass Filters

On the right side of the main menu, you'll find the settings for the high-pass and low-pass filters. This section allows you to set almost any **high-pass filter (above)** and **low-pass filter (below)** for the selected output channel, assigning the appropriate frequency range to each speaker. Both filters affect the selected channel only if the corresponding checkboxes are ticked, activating the filters.

Filter Type and Slope

Three different filter types are available to divide a signal's frequency range into distinct bands. A key characteristic of these filters is the **slope**, which describes how sharply the transition between the passband and stopband occurs. It is measured in **decibels per octave (dB/oct)** - a higher slope means a steeper transition and therefore stronger attenuation of unwanted frequencies. Another important concept is the crossover frequency, also known as the cutoff frequency. This defines the point where the filter starts to reduce the signal level and is crucial for separating frequency ranges in multi-way speaker systems.

The choice of the appropriate **filter type**, crossover frequency, and suitable slope depends on the specific requirements of your sound system, including the desired sound quality, phase coherence, and the transition characteristics between frequency bands.



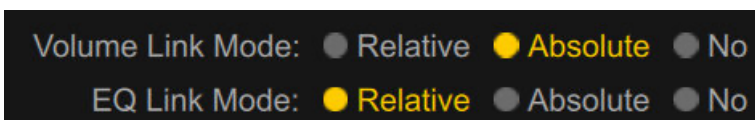
- **Butterworth Filter** is characterized by a very smooth frequency response. In the passband, there are no ripples or level fluctuations, which keeps the sound particularly balanced. The transition between passed and attenuated frequencies occurs smoothly at -3 dB, so adjacent frequency bands are not cut off abruptly. This makes the filter ideal for applications where a natural and harmonious sound reproduction is desired.

- **Linkwitz-Riley Filter** is based on the Butterworth filter but takes it a step further by combining two Butterworth filters of the same order. This results in a -6 dB attenuation at the crossover frequency, meaning the high-pass and low-pass filters perfectly complement each other. This ensures a linear summation of signals without level peaks or dips. This filter type is especially common in speaker crossovers, as it provides a clean transition between frequency bands and avoids sound coloration.
- **Bessel Filter** is especially known for its constant group delay. This means all frequencies within the passband are delayed evenly. As a result, the impulse response remains very precise and unaltered. Compared to other filter types, the Bessel filter causes the least phase distortion, which is why it is often used where a natural and time-accurate sound reproduction is desired. A downside is that the Bessel filter does not have a very linear frequency response, which can lead to tonal deviations depending on the application.

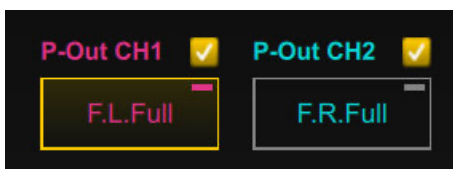
Link Mode

Our DSP software allows you to adjust each output channel individually. However, it can be useful to link multiple channels to apply identical settings simultaneously. This is especially recommended for high-pass and low-pass filters to avoid different parameters between the left and right output channels.

Two different linking modes are available, allowing **output levels and equalizer** values to be transferred from one channel to another either **absolutely or relatively**. Absolute means that all adjustments made are applied identically to all linked channels. Relative means that the difference between the linked channels remains consistent when adjustments are made.



The desired **output channels can be linked** by checking the corresponding checkboxes.



When you link two output channels together, any settings previously applied will not automatically be transferred to the other output channel. Only adjustments made to the level and/or equalizer after linking will apply to both channels.

Phase Switch

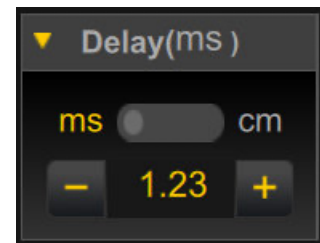
The phase switch allows the polarity of the output channel to be inverted by 180°. By clicking the Phase button, you can **switch between normal phase 0° and inverted phase 180°**.



Time Alignment

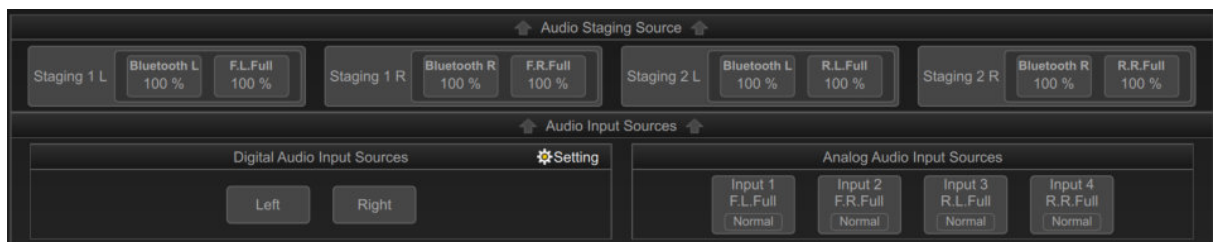
Below the phase switch on the main page, you will find the section for time alignment of each individual output channel. Our DSPs compensate for different distances between the speakers and the listening position by delaying the output signals accordingly (delay).

Without accurate time alignment, precise sound staging inside the vehicle cannot be achieved. For this reason, our DSP software allows fine adjustment in either millimeters or milliseconds to ensure an optimal sound stage. This brings us to the next tab: on the *Time* page, you will find a complete overview of all speakers, their assigned output channels, and a clear interface for configuring the time alignment settings.



Expert Mode (Continued)

Let's return to the I/O tab and enable **Expert Mode** again at the bottom of the screen. The most noticeable change is that only analog inputs 1 through 4 remain available, while inputs 5 through 8 are no longer used. In exchange, an additional layer with staging channels is introduced. These **staging channels** create a new processing layer between the input and output channels, significantly expanding the possibilities for sound optimization.



Die Staging Layer

The staging layer consists of four staging channels divided into two groups: Staging 1 and Staging 2, each with a left and right channel. Accordingly, each of these four channels also provides four EQ bands specifically intended for fine-tuning the front staging — in other words, the sound stage. This layer complements the input and output EQs and allows precise fine adjustment of your system.

A practical way to divide the staging channels could look like this:

- **Staging 1 left/right** for the front speakers and **Staging 2 left/right** for the rear speakers in the vehicle.
- Alternatively, **Staging 1 left/right** for the tweeters and **Staging 2 left/right** for the midrange speakers in a 3-way system, or for the mid-woofers in a 2-way system.

Once you have completed the assignment, you can select the **Staging EQ** via the **Input EQ** button on the right side of the I/O window. In the new window, you will find the frequency bands for each staging channel. Here, you can individually adjust the parameters to shape and position the sound stage of your audio system according to your preferences.

Staging Link Mode

As a special feature, the Staging EQ offers the so-called **Balance Link**. This functions like a balance control and, when the **left and right staging channels are linked**, automatically subtracts from one side whatever is added to the other side. This adjustment is performed precisely for each individual one of the 31 EQ bands, allowing the sound stage of your audio system to be adjusted with exceptional precision and detail.

The **Gain Link** works similarly to the previously described **Relative Link Mode**.

When adjustments are made, the linked

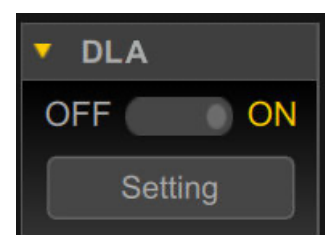
channels are changed identically while maintaining their relative relationship to one another.



Staging EQ Link Mode: Gain Balance No

Dynamic Loudness Adjustment (DLA)

To activate **Dynamic Loudness Adjustment (DLA)**, you must be in the **I/O** tab and enable **Expert Mode**. Then click on **Input EQ**. In the lower-right corner, you will see a switch used to activate DLA. Once enabled, the view of the parametric equalizer changes: additional status indicators for the DLA settings will appear below the first five EQ bands.



By clicking the status indicators, they will turn green, reserving these EQ bands for the DLC function. Please note that these bands **can no longer be used as regular EQ bands** afterward. The DLC function can be configured individually for each input channel. If you want to use DLC on multiple channels, it is recommended to link them so that the settings can be adjusted across all linked channels simultaneously.

Important: If you have already configured filter settings within the first five EQ bands, these settings will be overwritten when the DLA function is activated. In this case, you should move the corresponding filters to other bands of the 31-band EQ beforehand. To avoid this, it is recommended to leave the first five bands unused from the beginning if you intend to use DLA.

For additional DLA settings, click the **Settings** button located below the DLA switch. A new window will open containing **five EQ bands** that can be configured using the familiar filter types (parametric filter, high-shelf filter, and low-shelf filter). A new feature here is the ability to adjust the gain of these filters **depending on the volume level** between -12 dB and +12 dB. This is controlled via the Gain Factor located below the filter parameters.

The **volume-dependent filter adjustment** performed by the DLA function is based on the master volume setting configured within the DSP (main volume 0 to 60) — meaning the volume level you set via PC, app, or DSP remote control. It does **not take the input signal level** into account, for example the output level of your head unit.



Input, Output, Staging EQ

The use of three fully featured equalizers — Input EQ, Output EQ, and Staging EQ — provides enormous flexibility and precision in audio processing. Each equalizer fulfills a specific purpose, allowing the audio signal to be adjusted precisely and optimally for the respective application.

- **Input Equalizer (Input EQ / Base EQ)**

The Input EQ is used to correct the signal before it enters the signal processing chain. This is especially useful for eliminating frequency anomalies or sound coloration caused by the signal source (for example, the head unit). This ensures that the signal provides the most neutral possible foundation for further processing.

- **Output Equalizer (Main EQ)**

The Output EQ is applied directly before the signal is sent to the amplifier and speakers on the respective output channels. Its purpose is to optimize the processed signal for the specific speakers and their installation positions within the vehicle (for example, to correct resonances or peaks caused by the speakers themselves or by their mounting location).

- **Staging Equalizer (Staging EQ)**

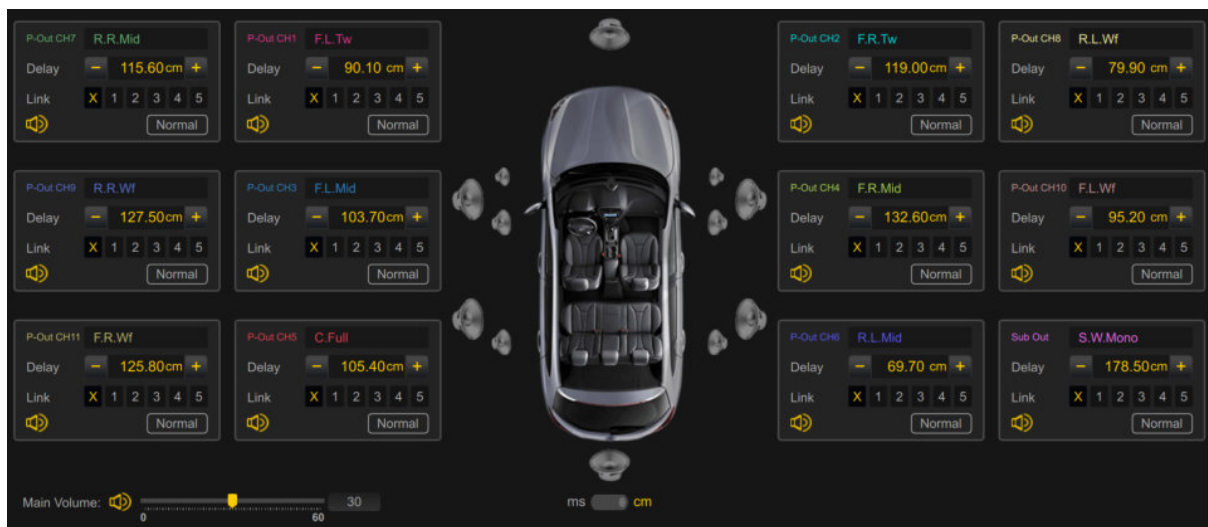
The Staging EQ is used to shape the sound stage within the DSP. The focus here is on tuning the speaker groups in order to create a balanced and harmonious sound stage (front staging). It enables precise adjustments such as balancing frequencies between speaker pairs, optimizing stage width and depth, and fine-tuning the overall sound according to the vehicle's acoustic characteristics or a reference curve.

The combination of these three equalizers not only provides exceptional sound quality, but also enables a professional workflow capable of meeting even the highest demands. With three equalizers available for processing, the audio signal can be adjusted precisely at every stage of the signal chain without the need for compromises.

Time Alignment (Delay)

In a vehicle, the speakers are never positioned at the same distance from the listener (usually the driver). This difference causes the sound waves from the various speakers to reach your ears at different times. The result is clearly audible: vocals sound diffuse, the sound stage (also referred to as front staging) appears shifted, and the bass from the subwoofer loses precision.

The **time alignment function** (also referred to as delay) within the Time menu is an essential tool for compensating for these differences. With its help, you can ensure that the sound from each speaker reaches the listening position at the same time. The result is a clear and balanced sound image in which all speakers work together perfectly.

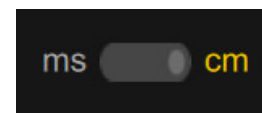


The DSP software provides two (maybe three) different approaches to time alignment:

- **Entering the delay in milliseconds (Delay)**
- **Entering the distance difference in centimeters (Distance)**
- **Adjusting the delay using group correction (Link Mode)**

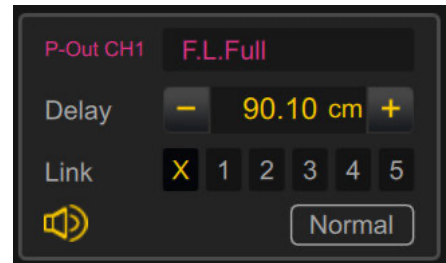
Here as well, it is beneficial to have assigned alias names on the main page, since on the Time page the speakers are now correctly mapped to their positions within the vehicle and can therefore be selected more conveniently.

First, select whether you want to enter the delay time for the individual speakers in **milliseconds (ms)** or in **centimeters (cm)**, and adjust the corresponding slider below the vehicle illustration. You can change this selection at any time, and the software will automatically convert your previous entries accordingly.



Regardless of whether you enter the values in milliseconds, centimeters, or use channel linking, the time alignment function is intended exclusively for optimizing the sound stage at a single listening position. As a result, the spatial imaging at other seating positions, such as the passenger seat, may be less precise compared to a setup without time alignment focused on the driver's position.

Each output channel has its own field where you can set the delay for the corresponding speaker. In addition, channels can be linked into up to five groups whose distances to the listening position are related. The familiar Mute and Phase buttons are also available here.



Using Group Correction (Link Mode)

Using the Link function, you can create **up to five speaker groups** by linking the respective output channels together. To do this, click on the fields labeled **1 through 5**, assigning the same number to speakers within the same group. This linking creates a **relative grouping** and ensures that the distance differences between the delay values of the linked channels remain consistent, even when the delay is adjusted. Clicking the **X** removes the link again.

Further adjustment is performed as usual either by entering the delay in milliseconds or by measuring the distance to the listening position.

It may be useful to link output channels, for example if you want to further increase the delay of the rear speakers to achieve a better front stage. This makes it possible to create greater delay values than the actual distance to the speakers would normally require.

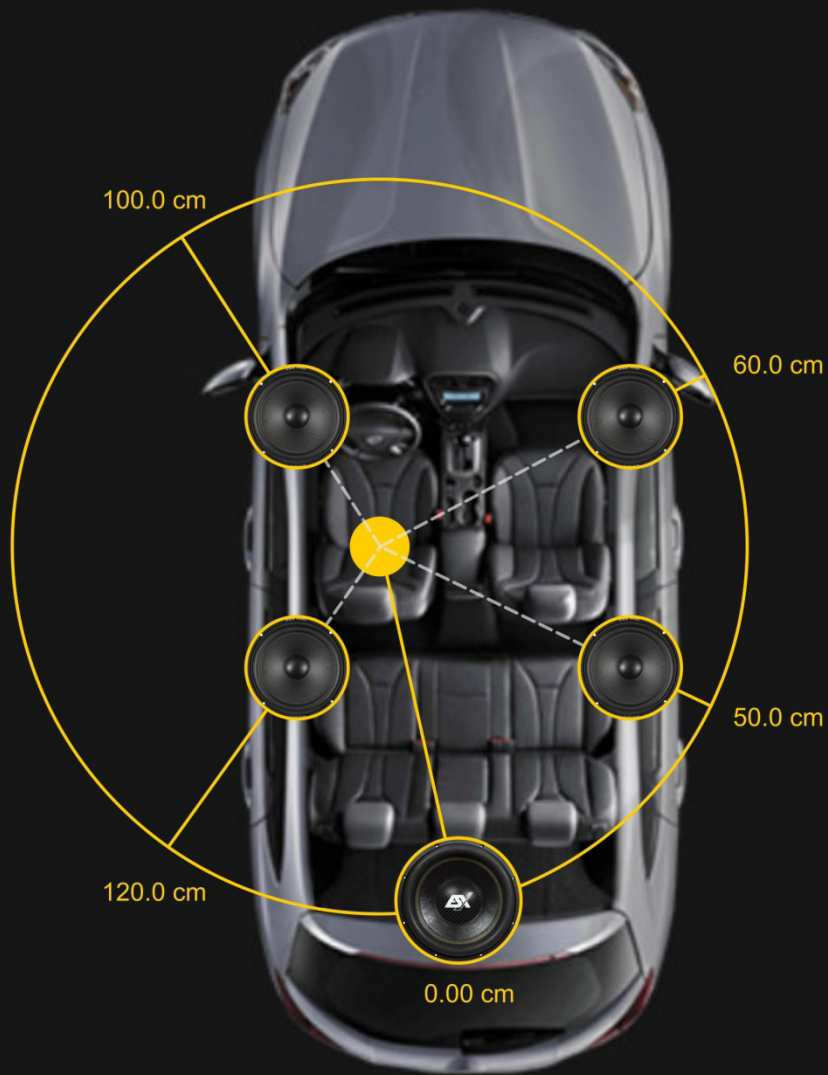
Entering the Delay (Time)

In this setting, you can manually enter the delay times for each output channel in milliseconds. This gives you full control over the values and allows you to adjust the delay precisely.

Entering the Distance Difference (Distance)

To configure the time alignment correctly, measure the exact distance between each speaker and your listening position — ideally at head level. A simple measuring tape is sufficient for this task. Start with the **speaker that is farthest away from the listening position**. This speaker is defined as the reference speaker and receives **no delay (0 cm or 0.00 ms)**.

Next, measure the distances to all other speakers and subtract them from the distance of the reference speaker. Enter the **calculated difference** into the corresponding output channels. The DSP software will then automatically convert the distance values into delay times in milliseconds so that all audio signals arrive simultaneously at your listening position.



Entfernung zum Hörplatz

Differenz

Verzögerung

Subwoofer	$180.0 \text{ cm} > \text{Referenzlautsprecher} = 0.0 \text{ cm}$	(Verzögerung 0.00 ms)
Vorne links	$80.0 \text{ cm} > 180.0 \text{ cm} - 80.0 \text{ cm} = 100.0 \text{ cm}$	(Verzögerung 2.94 ms)
Vorne rechts	$120.0 \text{ cm} > 180.0 \text{ cm} - 120.0 \text{ cm} = 60.0 \text{ cm}$	(Verzögerung 1.77 ms)
Hinten links	$60.0 \text{ cm} > 180.0 \text{ cm} - 60.0 \text{ cm} = 120.0 \text{ cm}$	(Verzögerung 3.53 ms)
Hinten rechts	$130.0 \text{ cm} > 180.0 \text{ cm} - 130.0 \text{ cm} = 50.0 \text{ cm}$	(Verzögerung 1.47 ms)

Meter

The Meter tab provides access to the **level meters**. Here, you can check whether the gain structure of your audio system is set correctly. The level indicators provide a clear visualization of the current **signal strength** and are available for both **input and output signals**.



The **green area** indicates that the level is optimally adjusted and no further adjustments are required. Using the yellow slider located on the side, you can individually increase the level of each channel by up to +12 dB or reduce it by up to -20 dB.

Yellow indicates that the level is approaching the critical threshold and that you should proceed carefully to avoid clipping. **Red**, on the other hand, indicates that the signal level is too high and must be reduced immediately to prevent distortion and potential damage.

It is recommended to keep the input level as low as possible so that the signal source — for example the head unit of your vehicle or, when streaming, your smartphone — must provide a higher output volume. At the same time, the output level of the DSP should be increased. This approach minimizes background noise and ensures the best possible audio quality.

Always make sure to leave sufficient **headroom**. This means maintaining a safety margin between the maximum possible level and the actual signal level. This ensures that even sudden level peaks will not cause clipping, allowing your audio system to remain clean and distortion-free.



CAR AUDIO SYSTEMS



ESX ist eine Marke der Audio Design GmbH
Am Breilingsweg 3 • D-76709 Kronau
www.esxaudio.de • dsp@esxaudio.de
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